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DEFINITY® Communications System Generic 2
Feature Descriptions, Vol. 1
Features A-I

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Operation of this equipment in a residential area is likely to cause interference, in which case the user at his/her own expense will be required to take whatever measures may be required to correct the interference.

Network Registration Number. This equipment is registered with the FCC under FCC network registration number AS593M-13283-MFE.

Answer-Supervision Signalling. Allowing this equipment to be operated in such a manner as to not provide proper answer-supervision signaling is in violation of Part 68 rules. This equipment returns answer-supervision signals to the public switched network when:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user.

This equipment returns answer-supervision on all DID calls forwarded back to the public switched telephone network. Permissible exceptions are:

- A call is unanswered
- A busy tone is received
- A reorder tone is received

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Acknowledgment

This document was prepared by the AT&T Technical Publications Department, Denver CO.

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Introduction

This manual describes the features of the AT&T System 85, Release 2 and the DEFINITY™ Communications System, Generic 2 (hereafter abbreviated as either DEFINITY Generic 2 or Generic 2). Where a distinction exists between versions, the versions are specifically identified. For System 85, Release 2 switches, versions are numbered from 1 through 4 and identified as Release 2, Versions 4 or R2V4. For DEFINITY Generic 2 switches, versions are identified as either Generic 2.1 or Generic 2.2.

Use of This Manual

This document is designed to help switch administrators and managers select communications features to be included in their initial system or to be added after the system is in service. It provides feature descriptions, administration and hardware information, and other considerations applicable to each available feature.

This manual can also be used as a tool for answering questions about the interactions between specific features. Each feature description lists the significant interactions with other features. Both ***supporting interactions*** (those interactions where two features work together or can be used to enhance each other's performance), and ***inhibiting interactions*** (those interactions where one feature may interfere in some way with the operation of another) are included. Specific feature capacities and limitations are described in each separate feature chapter and are summarized in tabular form in Appendix A.

Organization of This Manual

This manual is divided into two volumes. Each volume has a table of contents (for that volume) and a complete index. The manual as a whole consists of the following sections:

- System 85 and DEFINITY Generic 2 FEATURES — These sections list and describe the features currently available on the System 85 and DEFINITY Generic 2 switch. Included are:
  - — Traditional voice telephone services
  - — Telemarketing services
  - — Data communication features including integrated voice/data services
  - — Call-processing capabilities provided by the switch
  - — Call management and support features
  - — Networking features.

The features are presented in alphabetical order by feature name. The feature chapters use a format that is divided into major headings as follows.

Description

Defines the feature and tells what service it performs for the user or what function it serves within the switch. This section may be further subdivided depending on the complexity of the feature.

User Operations

Lists the common step-by-step operations needed to use the feature. Comments about an individual step are enclosed in parentheses. System responses to an individual step are enclosed in brackets.

Considerations

If applicable, indicates characteristics that need to be considered when the feature is used. If no significant considerations apply, this section is omitted.

Interactions With Other Features

Lists and discusses the interactions between this feature and other features.

Restricting Feature Use

If applicable, describes how the use of this feature can be restricted or controlled (by the customer and / or vendor). If no restrictions apply, this section is omitted.

Hardware Requirements

Identifies hardware items that each feature requires (beyond the normal station and switch hardware cofiguration). Where applicable, differences in hardware requirements between traditional and universal modules are identified. Also, when a suffix issue (for example, ANN11C) is indicated, this is the minimum version required. Earlier versions of the same item will not provide satisfactory service.

Feature Administration

Describes feature assignment using various means of administration: DEFINITY Manager II, MAAP (Maintenance and Administration Panel), SMT (System Management Terminal), TCM (Terminal Change Management), and FM (Facilities Management).

The TCM and FM features on the AP 16 apply only to System 85, Release 2, Version 3 and earlier switches.

The DEFINITY Manager II is used with the DEFINITY Generic 2 switch. The MAAP and the DEFINITY Manager II and Manager IV are the only administration vehicles that can be used with the Generic 2 switch.

- Appendix A: Configuration Limits — Lists the capacities and limitations applicable to the System 85, Release 2 and the DEFINITY Generic 2 switches. This information is presented in tabular fashion to allow a quick comparison of the differences between versions.
- Appendix B: Call Distributors — Compares the call-distribution features: DDC (Direct Department Calling), UCD (Uniform Call Distribution), EUCD (Enhanced Uniform Call Distribution), and ACD (Automatic Call Distribution). It lists the various features and attributes that are available from each of these features.
- Appendix C: Administration Facilities — Lists and briefly describes the various administration tools available for the System 85, Release 2 and the DEFINITY Generic 2 switch.
- Appendix D: Data Communications — Lists and describes some of the data communications signaling formats used with the System 85 and the DEFINITY Generic 2 switch.
- Appendix E: Images, Appearances, and Extensions — Describes and discusses the three basic ways of terminating a telecommunications line at a terminal device.
- Appendix F: Enhanced Trunking — Lists and describes some of the new and enhanced trunking arrangements and services available with the System 85 and DEFINITY Generic 2 switch.
- Appendix G: Integrated Services Digital Network — Provides an Overview of the ISDN (Integrated Services Digital Network) general concept and basic principles. This appendix also provides general information on network features and services and the message-oriented signaling used in an ISDN.
- Appendix H: Data Communications Interface Unit — Describes the DCIU and discusses its various applications. This appendix also provides some recommended administration values for different feature and application configurations
- Appendix I: Compatibility Matrix — Lists various adjuncts, interfaces, and feature elements and shows their compatibility with the different versions of the System 85, Release 2 switch and the DEFINITY Generic 2 switch.
- Appendix J: Acronyms and Abbreviations — Lists the acronyms and abbreviations used in this reference manual and their respective terms.
- Glossary — Lists commonly used terms and their definitions.
- Index.

Enhancements for DEFINITY Generic 2.2

Several new features and enhancements are available for the DEFINITY Generic 2.2 switch.

New Features

- Expert Agent Selection

  The EAS (Expert Agent Selection) feature enhances Call Vectoring and ACD (Automatic Call Distribution) capabilities by distributing selected ACD calls to subsets of ACD agents who are members of larger splits. These agent subsets are based on the agents' call-handling skills, which could be based on agent training, type of product or service, foreign-language skills, or other expertise.

- Call Work Codes

  A CWC (Call Work Code) is a customer-defined code such as an account number, a call activity code, a credit card number, or a social security number. An ACD agent using a DCP voice terminal can enter a CWC during or after an ACD call. The agent must be measured by and logged into CMS (Call Management System).

- WCR (World Class Routing)

  The World Class Routing feature is introduced in DEFINITY Generic 2.2. This is an advanced networking feature that provides more flexibility in network design than was previously available, and combines the functions of the previous networking features AAR (Automatic Alternate Routing) and ARS (Automatic Route Selection). The WCR feature replaces the AAR and ARS features.

Enhanced Features

The following are the features that are enhanced in DEFINITY Generic 2.2:

- ACD (Automatic Call Distribution)

  - 2048 Agents Per System

    The maximum number of ACD agents per system increases from 1024 to 2048.

  - Split Size Restrictions Removed

    Split size restrictions have been removed. That is, the size of an ACD split does not have to be specified in multiples of 16. An ACD split may contain any number of agents from 0 to 1024 (1023 if the split is measured by CMS).

  - Splits Administered as Measured by CMS

    ACD splits are administered as either measured by CMS or not. As many as 1023 extension number can be measured by CMS. Prior to DEFINITY Generic 2.2, individual extension numbers or a range of extension numbers is administered as measured by CMS.

- 106B Display Unit Assignments Based on Extension Number

  The assignment of agents to 106B display units (used to monitor calling activity) is based on extension number. Prior to DEFINITY Generic 2.2, 106B assignments are based on agents' split and member numbers. If an agent assigned to a 106B moves from one split to another, the 106B assignment has to be changed. Beginning with DEFINITY Generic 2.2, 106B assignments do not have to be changed when agents move from one split to another.

- Lamp Indication for Stroke-Count Buttons

  If the stroke count function is assigned to a button with a status lamp, the status lamp lights for 2 seconds if a message can be sent to CMS when the button is pressed. If a message cannot be sent, the flash rate of the status lamp is set to broken flutter for 2 seconds.

- Call Vectoring

  - 511 Vectors Per System

    The number of vectors that can be administered per system increases from 128 to 511.

  - Go To Vector Command

    This new command enables vector processing to branch to a different vector, similar to the way the "go to step" command enables vector processing to branch to a different step in the same vector.

  - 255 Recorded Announcement Trunks

    The number of recorded announcement trunks increases from 84 to 255.

  - 475 Abbreviated Dialing Group-List Items

    The number of Abbreviated Dialing group-list items (used by the "route-to" command) increases from 95 to 475 (from 1 group list to 5 group lists).

  - Administration Changes

    Beginning with DEFINITY Generic 2.2, Procedure 030, Word 3 is used to program the steps of a vector and Procedure 030, Word 2 is used to transfer a vector from the scratch pad to permanent memory. Prior to DEFINITY Generic 2.2, Procedure 030, Word 3 was used to program the steps of a vector and to transfer a vector from the scratch pad to permanent memory.

    The allowed values for the "wait" command (previously called the "delay" command) have been changed from even values between 2 and 998 seconds to even values between 0 and 998 seconds.

- IXC (Interexchange Carrier) Access

  In previous versions, the application of the IXC Access feature was controlled completely within switch administration. There was no way that a caller could influence the choice of IXC (if more than one choice was available). Beginning with DEFINITY Generic 2.2, an administrable option is available (through the WCR feature) that can allow the caller to enter an IXC code as part of the dialed number.

Feature Name Changes

AUDIX™ Voice Messaging System

This feature has been available as the AUDIX feature since System 85, Release 2, Version 2. In 1991 the term AUDIX became a trade mark necessitating a feature name change.

CallVisor™ ASAI Gateway Interface

This feature was introduced in DEFINITY Generic 2.1 (issue 3.0) as ITGI (Integrated Telemarketing Gateway Interface). For Generic 2.2, the feature name changes to CallVisor ASAI Gateway Interface. This feature is commonly referred to as the ASAI Gateway Interface feature.

Feature Independent Enhancements

Some enhancements available with DEFINITY Generic 2.2, are not specifically related to a particular feature. These include:

- Increased Number of ORs (Originating Registers)

  The total number of ORs available on System 85, Release 2 and DEFINITY Generic 2.1 switches is 300. With the DEFINITY Generic 2.2 switch, the total number of ORs available increases to 512.

- Increased Capacity of ORs

  The size (maximum number of digits) of an OR on System 85 and DEFINITY Generic 2.1 switches is 36 digits. On Generic 2.2 switches, the size of an OR is increased to 68 digits.

Abbreviated Dialing

Description

The Abbreviated Dialing feature allows terminal users to dial frequently called or emergency numbers with significantly fewer button presses than if the number were dialed in the usual way (that is, one digit at a time). Abbreviated Dialing can be used to dial on- or off-premises numbers. A single Abbreviated Dialing call can dial as many as 36 characters.

Repertory Dialing:

The name Repertory Dialing is sometimes used as a synonym for Abbreviated Dialing. In this manual, the name repertory dialing is used for a subset or part of Abbreviated Dialing rather than as a synonym. In this manual, repertory dialing refers to a function of some voice terminals that provides specific buttons for storage and one button dialing of numbers. The storage and dialing functions are provided by the voice terminal rather than the switch. Specific operations of repertory dialing are dependent on the voice terminal used. In the following discussion, the name repertory dialing is used in a generic sense unless a specific terminal (such as the AT&T 7103A Voice Terminal) is identified.

AD (Automatic Dialing)

The term automatic dialing is also sometimes used as a synonym for Abbreviated Dialing. In this manual, however, this term refers to a subset of the Abbreviated Dialing feature. Automatic dialing refers to the use of an administrable button on a voice terminal that is used for one button dialing. The differences between automatic dialing and repertory dialing are as follows:

- Automatic dialing buttons are administerable (their function is not fixed by hardware)

  Repertory dialing buttons have a fixed function (they cannot be used for any other purpose)

- Automatic dialing buttons access numbers stored in switch memory

  Repertory dialing numbers are stored on the terminal

Feature History and Development

This feature was first available with System 85 in Release 1. Subsequent enhancements have included the following

- The system list was expanded from a maximum of 99 different items (numbers) in Release 2, Version 2 to a maximum of 9999 different items in Release 2, Version 3. The system list can have 9, 99, 999, or 9999 items in version 3 and later switches.
- Release 2, Version 2 switches have a capacity of 999 group lists. Each of these group lists can hold as many as 30 items.

Release 2, Version 3, and later switches can have as many as 9999 group lists. Each group list can have up to 95 items, in increments of 5 (that is, 5, 10, 15, etc.).

- Personal lists on Release 2, Version 2 switches can have up to 30 items, while on Release 2, Version 3 or later switches, they can have up to 95 items.
- The Manual Digit Entry function was not new with Release 2, Version 3. However, the special encode that allows manual digits to be entered in the middle of a stored number was new.
- In Release 2, Version 4, the total number of personal and group lists combined was increased to 52,223, and the total number of records (list and button entries) was increased to 262,143. The basis for assigning Abbreviated Dialing was changed from extension number to equipment location.
- The *End of Dialing* special function character is added in DEFINITY Generic 2.1, Issue 2.0 and System 85, Release 2, Version 4, Issue 3.0.

Available Characters

The characters available for Abbreviated Dialing are the digits 0 through 9 and the special dialing characters "*" and "#." Special function characters are also available. These special function characters can be stored with the number to be dialed and will cause the switch to perform actions such as:

- Allowing digits to be entered manually from the terminal
- Causing a pause (approximately 1.5 seconds) in the stream of digits being outpulsed
- Waiting (from 4 to 10 seconds) during the dialing sequence before finishing dialing
- Suppressing selected digits on displays

Number Size and Type

Each memory location used for Abbreviated Dialing can contain as many as 20 characters (digits and special function characters combined). Up to 15 digits can be added manually to the stored digits. If the need arises, multiple storage locations can be used together to dial as many as 36 characters on System 85 and Generic 2.1 switches, and as many as 60 characters for Generic 2.2 switches, automatically.

Any of the following types of numbers can be dialed automatically using Abbreviated Dialing

- Feature* or trunk-group access codes
- Extension numbers

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* While feature access codes generally are permitted there are some cases when specific feature access codes are blocked. See the "Considerations" section for details.

- Public and private network numbers
- International numbers
- Portions of complete numbers (where Manual Digit Entry is used)
- Account Codes
- Authorization Codes.

Methods of Storage

Both the System 85 and the DEFINITY Generic 2 switches provide two forms of number storage: button-stored numbers and list-stored numbers.

Button-Stored Numbers

Button-stored numbers are associated with an AD (Automatic Dialing) button on a multiappearance voice terminal or a *Repertory Dialing* button on a 7103A Programmable* voice terminal. Numbers accessed by an AD or repertory dialing button are associated with the specific voice terminal and are not shared. On terminals other than a 7103A these numbers are store in switch memory.

Numbers stored on a 7103A Programmable voice terminal do not use memory space on the switch. Rather, these numbers are stored within the voice terminal. This arrangement is sometimes referred to as *Repertory Dialing* and Repertory Dialing buttons. Numbers stored within the terminal itself are functionally equivalent to those accessed by AD buttons (numbers stored on the switch) and list-stored numbers with one exception. That exception is that terminal stored numbers cannot be programmed with *special function* characters unless provided for in the terminal design (the pause function is available on the 7103A). Several special function characters are available to Abbreviated Dialing users accessing switch stored numbers. Special function characters are discussed in detail later in this feature description.

List-Stored Numbers

List-stored numbers exhibit the same general characteristics as button-stored numbers except that list-stored numbers can either be reserved for a single terminal (personal list) or shared by several users (group or system list). These lists are kept in switch memory, and access to them (either for the purpose entering list items or to use list items to place a call) is controlled through switch administration.

* The 7103A Programmable voice terminal is no longer manufactured and cannot be ordered. However, its functionality is supported and it can be used where available.

Abbreviated Dialing List Types

Abbreviated Dialing lists are categorized in either (or both) of two ways:

- Functionally
- Relationally

Functions/List Types

There are three functional types of Abbreviated Dialing lists:

- System List
  - Up to 9999 items (numbers) can be provided.
  - Access is provided either globally (any user on the switch can access system list items), or access can be specifically assigned to selected users.
  - The switch administrator programs this list (adds, changes, or deletes entries).
- Group Lists
  - Up to 9999 group lists can be assigned. The maximum number depends on the size of the lists.
  - Each group can have from 5 to 95 items in increments of 5. For DEFINITY Generic 2.2, up to 5 group lists can be designated as Vector-group lists.
  - Numbers are accessible by a selected group of users who frequently dial the same numbers.
  - One of the group voice terminals is assigned to program the group list numbers (or the list can be programmed by the switch administrator).

    In Version 3 and earlier switches, any extension that homes to that terminal may be used to perform the programming.

    In Version 4, any extension assigned to that terminal can be used to program the group list.
- Personal Lists
  - Any number of personal lists (up to two per terminal) can be provided while switch memory space is available.
  - Each list can have from 5 to 95 items in increments of 5.
  - Numbers are accessible only from the assigned terminal. Personal lists provide abbreviated dialing of numbers for users without AD buttons that would otherwise be used for button-stored numbers.
  - The owner (or switch administrator) programs this list.

Relational List Types

There are two relational list types. These categories are not in addition to the functional types, but rather categories of the same lists in relation to a specific user terminal or terminal group.

- List A

  Each terminal can have access to two abbreviated dialing lists. In relation to a particular terminal, the first of these lists is usually designated as List A. List A can be either a personal list or a group list. Traditionally, List A has been used for the personal list of a terminal; however, this is not a required relationship.

- List B

  List B is the second of the two possible lists that can be assigned to a terminal. This can be a personal list or a group list. Traditionally, List B has been used for the group list of a terminal; however, this is not a required relationship.

Not only can List A and List B be either a personal or group list, but they can both be personal lists or they can both be group lists. The System List is the only functional list category that cannot apply to either of the relational list types.

List-Stored Number Addresses

Each list-stored number has an address that consists of a list-access code and a list-item (or index) number. Users can access specific list items by dialing the appropriate list access code and index item number, or they can be provided with feature function buttons That perform this function. Three list-access codes are assigned in each System 85 or DEFINITY Generic 2 switch.

- One list-access code or button type *(sitm)* is assigned to the system list. Restricted users are unable to dial this code.
- One list-access code or button type *(alst)* is assigned to provide users with access to their terminal's A-List. This list may be a group list or personal list.
- The third list-access code or button type *(blst)* is assigned to provide access to a terminal's B-list. This list may also be a group or personal list.

The list-item number corresponds to a stored number's position in the list. Lists of five numbers have index numbers 1 through 5. Lists of ten numbers have index numbers 1, 2, 3, etc., through 0, with 0 as the tenth item. All other sizes of lists use 2-, 3-, or 4-digit item numbers (for example, list size 100 — list index number 01 through 00, with 00 as the 100th item). Leading zeros must be dialed when appropriate (index item of less than 10 or less than 100) to access stored numbers in lists of 100 or more.

Special Functions

The following special functions can be used with Abbreviated Dialing to meet special dialing needs:

Pause

Used when a second dial tone is required before sending additional digits. This is typically used when a pause of 1.5 seconds or less is expected before second dial is tone provided on a trunk call.

Wait

Used when a longer wait is needed before sending additional digits. This function is used when a pause of from 4 to 10 seconds is expected before second dial tone is provided. A 4 second wait occurs when the terminal used does not have a wait button assigned, and a 10 second wait occurs when the terminal used has a wait button assigned. The longer wait allows for trunk dial tone or a dialing prompt to be provided from other switches (long distance and credit card calls).

Mark

Used when an outgoing call requires the characters asterisk (*) or pound (#) to be entered from the dialing pad and then outpulsed. Without the mark, the * or # characters are interpreted as the first digit of a dial access code and are not outpulsed. When the mark function is programmed into a stored number, the function is activated after a talk path is established with another switch via a trunk. If the user has a MARK button, the lamp lights when the function is active for dialing purposes. That is, once the MARK button lamp is lit, a * or # can be dialed from the keypad and will be outpulsed. This function remains active for the duration of the call.

An * or # character entered as part of the stored number does not require the use of the mark character.

Stop or ***Wait for Dial Tone***

Used in conjunction with Data Terminal (Keyboard) Dialing. The Stop function conditions the data terminal for return of dial tone from the switch. See the Data Call Setup feature. The Stop function character can be entered in an Abbreviated Dialing list item through switch administration or from a voice terminal using a FUNCTION button. There is no prevision for a Stop Function Programming button.

Manual Digit Entry

Used when it is desired (or necessary) for the caller to enter some of the digits to be dialed from the terminal dialing pad. This function is useful for entering a security, authorization, or access code in a stored number where it is not desired to make the code a part of the stored number string.

Suppress

Used when a security code, such as a password or authorization code, is used in an Abbreviated Dialing list entry and a terminal with the Display—Voice

Terminal feature is used. This function is available on Version 3, and later System 85 switches, and on the DEFINITY Generic 2 switch. By enclosing the password or authorization code in the suppress function, the characters of the security code are converted to "s" on the display. This reduces the possibility of inadvertently compromising the security code.

End of Dialing

The end of dialing character (#) is used to indicate to the switch that all digits have been dialed. Since no more digits are expected the originating register is released immediately when this character is encountered. This function is intended for use on outgoing (trunk access) calls and is not needed on station-to-station calls. In a DCS (Distributed Communications System) arrangement, especially where Extension Number Portability is used, it may not always be obvious to the user when a call will use an outgoing trunk. In such an arrangement, use of the End of Dialing function on what appears to be a station-to-station calling number should not cause any problems.

Special Function Code Entry

Special function codes are entered in a list item along with the character string to be dialed. These codes are entered in the same way as the dialed digits.

- The switch administrator can enter any of the special function codes using the Manager II, MAAP or SMT.
- Authorized terminal users can also enter special function codes if their terminals are equipped with either:
  - A FUNCTION button

    or

  - Individual Special Function programming buttons such as WAIT *(wait),* PAUSE *(paus)* etc., except that there is no special function programming button for the *Stop* or *Wait for Dial Tone* character

Special function characters cannot be entered into list-items or AD (Automatic Dialing) buttons from a voice terminal that does not have either a FUNCTION button or Special Function programming buttons.

The Function Button

The FUNCTION button is useful when there is a need to conserve buttons on the terminal being used. The special function characters can be programmed by entering a function number (see User Operations) after pressing the FUNCTION button. When there is no need to conserve buttons, individual special function programming buttons can be assigned for the separate special functions.

Special Function Programming Buttons

A voice terminal can be equipped with special function programming buttons which will allow the following special function characters to be entered into an Abbreviated Dialing list-item with a single button press.

Special Function	Procedure 059, Word 3 Encode	Manager II, Task Mode Encode
Pause	14	*paus*
Wait	15	*wait*
Mark	16	*mark*
Manual Digit Entry	18	*mdgt*
Suppress (display)	19	*supp*
End of Dialing	20	*eoad*

NOTE: No programming button is available for the *Stop* or *Wait for Dial Tone* Special function. This function can be entered using the FUNCTION through switch administration.

Manual Digit Entry

Abbreviated Dialing can accept one or more digits dialed from the dialing pad to create multiple uses for the same stored number. There are two ways of providing for manually entered digits.

- Digits Entered Manually After the Stored Digit String

  Abbreviated Dialing automatically appends manually dialed digits to the end of the stored number digit string (unless the End of Dialing special function character is used). This allows flexible use of a single stored number. For example, the stored number can be used to dial a trunk-group access code and network location code. The user can then manually dial a specific extension number.

- Manual Digits Inserted in the Middle of the Stored Digits

  Through switch administration (Procedure 059, Word 3), a special function character (encode 18) can be entered in a stored number that causes the switch to wait for a designated number of digits (from 01 to 15). The switch waits at the point where the encode is encountered. For example, the switch can outpulse the first four or five digits of a stored number such as a data trunk-group access code, wait for an account code to be dialed by the user, and then dial a distant data port. The Manual Digit Entry function code can also be entered from a voice terminal using either the FUNCTION button (encode 5) or a Special Function Programming button *(mdgt).* However, when entered from a voice terminal, the number of digits that can be dialed manually is reduced (1 to 9). Three character spaces or the number of digits to be entered manually (whichever is greater) are used when this function code is entered in a stored number.

User Operations

The following are the user operating procedures for this feature.

Making a Call Using Abbreviated Dialing

With Automatic Dialing (AD xxxx) buttons:

1. Go off-hook. [Dial tone]
2. Press the desired **[AD]** button. [Call-progress tone] (The button label should identify the number or name associated with the button.)

Without AD buttons (2-Step Abbreviated Dialing):

1. Go off-hook. [Dial tone]
2. Press the list-selection button (**[SYSTEM]** , **[GROUP]** , or **[PERSONAL]**),

   or

   Dial the list access code. [Second dial tone]
3. Dial the list-item number associated with a stored number. Leading zeros must be dialed when appropriate. [Call-progress tone]
4. If two list items are to be dialed together to form a dialed number of more than 20 characters, press the list button or dial its access code again* and dial the second list-item number.

Programming a Stored Number

With AD buttons:

1. Press **[ABRVDIAL PROGRAM]** ,

   or

   Dial the Abbreviated Dialing program access code. [Confirmation tone]
2. Press an **[AD]** button. [Dial tone]
3. Dial the number to be stored. *(See also* "Programming Special Functions.")
4. Press the **[AD]** button again. [Confirmation tone]

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* When a list access code is used to access a second list item for 2-step Abbreviated Dialing the first digit of the access code must be either a "*" or a "#."

Without AD buttons (list-stored numbers):

To program an Abbreviated Dialing list, a terminal with the appropriate permissions must be used.

1. Press **[ABRVDIAL PROGRAM]** ,

   or

   Dial the program access code. [Confirmation tone] (Once in the programming mode, any number of list-items can be programed.)
2. Press list-selection button (that is, **[GROUP]** , **[PERSONAL]** , **[List A]** , **[List B]**),

   or

   Dial the appropriate list access code. Dial tone]
3. Dial an item number in the list. [Second dial tone]
4. Dial the number to be stored. (See "Programming Special Functions.")
5. Press the list-selection button again to enter the new number,

   or

   Dial **[#]** if no button is provided. [Confirmation tone]
6. Press **[ABRVDIAL PROGRAM]** to leave the program mode. This button can be pressed instead of the list button to leave the list item unchanged.

Programming Special Functions (Used While Entering the Stored Number)

Using ABRVDIAL FUNCTION (also known as "ESCAPE") button:

1. Press **[ABRVDIAL FUNCTION]** .
2. Dial the digit, **[1]** , **[2]** , **[3]** , **[4]** , **[5]** , **[6]** , or **[7]** to program:

   1 Pause

   2 Wait

   3 Mark

   4 Stop (Wait for Dial Tone)

   5 Manual Digit Entry

   6 Suppress

   7 End of Dialing.

   NOTE: Manual Digit Entry can be programmed by a voice terminal user, however, the number of digits entered is limited to a single key press (one of the numbers 1 to 9). If more digits need to be entered manually (up to 15) this function must be entered by the switch administrator or service personnel.

Using Special Function Programming buttons:

For 1-button programming, press one of the special function programming buttons (Pause, Wait, Mark, etc.) to enter the named function into an Abbreviated Dialing number location.

Reviewing the Contents of Stored Number (Display— Voice Terminal Feature Required)

To review the contents of an AD (Automatic Dialing) button:

1. Remain on-hook.
2. Press the **[AD]** button. [Display shows the stored number.]

NOTE: Beginning with Release 2, Version 4, if the AD button is used to access a list stored number (system, group, or personal list), the contents of the list item are displayed.

Considerations

Feature Capacities

Group list items, personal list items, data terminal default dialing entries, and Automatic Dialing buttons share the same storage area within switch memory. The number of list items and buttons assigned depends on the memory area allocated One Automatic Dialing button consumes the same memory space as one list item entry. The maximum area allows for 65,535 Automatic Dialing buttons and list entries in System 85, Release 2, Version 3 and 262,143 in Version 4 and DEFINITY Generic 2.

Additional Automatic Dialing functionality (using Repertory Dialing buttons), independent of switch memory, can be provided on 7103A* Programmable voice terminals. The 7103A can be assigned as many Repertory Dialing buttons as are available on the set. With the 7103A, numbers are stored on the terminal and do not require memory space on the switch.

The maximum number of list items and AD buttons that can be assigned, by switch version, is shown in Table 2-A.

As 10-item, 15-item, and 20-item lists are assigned, the maximum number of lists is reduced. The same is true as Automatic Dialing buttons are assigned.

* The 7103A Programmable voice terminal is no longer manufactured and cannot be ordered However, its functionality is supported and it can be used where available.

A stored number can contain up to 20 characters. Allowable character include the digits 0 through 9, the asterisk (*), the pound sign (#), and the Abbreviated Dialing special function codes. Special function codes, except Manual Digit Entry, require two characters. Manual digit entry requires three characters.

TABLE 2-A. Abbreviated Dialing List Capacities

Characteristic	System 85, Release 2				DEFINITY
	V1	V2	V3	V4	G2
Characters Per Button or List Item	20	20	20	20	20
Maximum System List Items	99	99	9,999	9,999	9,999
Maximum List Items in a Nonsystem List	30	30	95	95	95
Number of NonSystem Lists	2,047	5,118	13,107	52,223	52,224
Number of Group Lists	500	1,000	9,999	9,999	9,999
Maximum Entries (List Items and AD Buttons Combined)	24,000	65,535	65,535	262,143	262,144

Virtually any domestic telephone number can be stored in a single list item or Abbreviated Dialing button. For international calls and special dialing situations (such as credit card calls) where more characters may be required, two or more separate list-items can be used back-to-back. When multiple list-items are used back-to-back, up to 36 digits can be dialed on System 85 and Generic 2.1 switches, and up to 60 digits can be dialed on Generic 2.2. This allows any dialable destination to be reached. Table 2-B shows capacities per call (applicable during programming and use).

List Access Codes for 2-Step Abbreviated Dialing

When a list access code is used to access a *second* list item for 2-step Abbreviated Dialing the first digit of the access code *must be* either a "*" or a "# ." Otherwise, the switch might perceive the dialed access code as additional digits to outpulse in completing the call.

Calls Using Dialing Prompts

Special dialing requirements exist for selected calling situations, such as placing credit card calls. In these cases, a second dialing sequence is needed after the controlling switch returns a dialing prompt signal. Because the time internal between the first and second dialing sequence varies significantly, these dialing sequences can be difficult to program for the Abbreviated Dialing feature. The following two methods, both using two buttons, have been extensively tested and are recommended for these special dialing situations:

- *With Wait Button*

The AD buttons are programmed with two or more wait functions between the first and second digit string. Since the caller can cancel the wait function by pressing the wait button, the number of wait functions is not critical. The second AD button stores the second digit sequence (for example, a credit card number). To place a call, the user goes off-hook, presses the first and second AD buttons consecutively, and listens for the dialing prompt signal. After hearing the prompt, the caller presses the wait button to cancel the wait and the second dialing sequence is outpulsed immediately.

TABLE 2-B. List Item Memory Usage

Characters	Character Spaces Required		
	Stored Item	Storage Access Per Call	Dialing (Outpulsed) Per Call
Stored Digits (0-9, *, #)	1	1	1
Digits from Keypad*	N/A	1	1
Special Functions			
Pause	2	2	0
wait	2	2	0
Mark	2	2	0
stop	2	2	0
Suppress	2	2	0
End of Dialing	2	2	0
Manual Digit Entry	3	3**	n†
Total Allowed For System 85 and Generic 2.1	≤ 20	≤ 36	≤ 36‡
For Generic 2.2	≤ 20	≤ 60	≤ 60‡

* Dialed during activation, other than 2-step dialing list access, and for the Manual Digit Entry function-

** The Manual Digit Entry function requires 3 spaces or the number of digits to be entered manually, whichever is greater.

† The number of digits outpulsed is equal to the number of digits entered manually.

‡ It is possible to use Abbreviated Dialing to invoke the AAR, ARS, or WCR feature. These features can modify the digits, which could result in digit insertion. If the combination of digits generated by Abbreviated Dialing list items, dialed digits, and digit insertion exceeds the applicable limit, truncation will occur and the call will fail.

- *Without Wait Button*

This method is similar to the preceding method except that here the user must estimate the time delay needed between the first and second digit string. Without a wait button to cancel any remaining wait time programmed, the wait function must time out before the second group of digits is sent. The AD buttons are programmed with enough wait and pause functions between the two digit strings to approximate the desired time delay. The second button again stores the second dialing sequence. To place a call, the user goes off-hook, presses the first and second AD buttons consecutively, and waits for the call to complete. If the call fails, the buttons need to be reprogrammed to add more time delay. This process must be repeated until the call completes successfully.

Attendant Consoles

The users of attendant consoles do not have access to the Abbreviated Dialing feature. An attendant cannot access the system list, a group list, or a personal list. Also, automatic dialing buttons cannot be assigned to an attendant console. There are two features that serve to reduce attendant dialing. These features are Attendant Direct Extension Selection and Attendant Direct Trunk Group Selection.

Abbreviated Dialing and Restricted Numbers

During the programming of an automatic dialing button or an Abbreviated Dialing list item, the switch does not check the programmed number for validity. Therefore, numbers that are restricted to a user can be programmed in the usual manner. However, when a call is placed using the Abbreviated Dialing feature, the switch performs the necessary validity checks, and when necessary, denies the call when the programmed digits are outpulsed.

Blocked Feature Dial Access Codes

While feature dial access codes can generally be dialed using the Abbreviated Dialing feature, the Abbreviated Dialing and the LND (Last Number Dialed) feature dial access codes (Encodes 90 through 94, and 59) are specifically blocked. These codes can be programmed into the abbreviated dialing lists and buttons. However, when the list items or buttons are used, the calls are blocked by the switch (intercept tone is returned). This operation avoids the possibility of recursive feature activation.

Button-Stored and List-Stored Numbers

The following items should be considered before deciding on whether to assign a number to button- or list- storage:

- It is more economical (that is, saves memory space) to assign numbers to Repertory Dialing buttons than to an Abbreviated Dialing list.
- If the desired number already exists in the system list or in a group list, it would be wasteful to use button-stored Repertory Dialing. It would be better to use common list-stored Abbreviated Dialing to point to the list and item of the existing stored number. The user must have access to the list in order to use this method.

Shared Extensions

Multiappearance voice terminals and straight line sets can share extension numbers (see the Bridged Call feature). Specifically, images of an extension's appearance(s) can reside on more than one voice terminal.

In System 85, Release 2, Version 3 and earlier switches, one of these terminals (possessing an image of every appearance) is assigned as the primary or "home terminal" for the extension. The Abbreviated Dialing lists for such an extension can only be accessed from the home voice terminal.

In Version 4 and in DEFINITY Generic 2, Abbreviated Dialing is assigned on the basis of ELL (Equipment Line Location) rather than extension number. With ELL as the basis for assignment, each terminal can have its own Abbreviated Dialing list(s), and "home terminal" status is not a factor. All extensions assigned to the same terminal (ELL) have access to that terminal's list(s). For voice / data stations, both the voice and data terminals have access to the same list(s). However, other images and appearances of the same extension on different terminals, have different ELLs and cannot access these lists (except for group lists where both terminals belong to the same group).

Manual Digit Entry

Users must be aware of the dialing requirements in order to use the Manual Digit Entry function properly. In most cases, the stored number will have only one application where a fixed number of manual digits is used. This avoids problems that can arise when attempting to use the stored number in some other way.

- Dialing fewer than the designated number of digits

  This can be done; however, the "#" must be dialed to show end-of-dialing. Otherwise, the interdigit timer will time out, and the user receives reorder tone.

- Dialing more than the designated number of digits.

  If the user dials more than the specified number of manual digits, the extra digits will be dialed at the end of the stored number string, regardless of where in the string the Manual Digit Entry encode appears. In most cases these extra digits are ignored since the destination usually expects only a certain number of digits and terminates digit collection when this number is reached.

Hard and Soft Processor Swaps

The contents of the system list, group lists, and personal lists are stored in a translation portion of switch memory. Therefore, these lists endure a hard processor swap.

Stable calls placed using the Abbreviated Dialing feature endure a hard swap.

If a voice terminal user is changing a list item when a hard swap occurs, the list item remains unchanged after the hard swap.

The Abbreviated Dialing feature operates normally during a soft processor swap.

Interactions with Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Call Detail Recording

Outgoing calls made by this feature are recorded just as if the stored number had been manually dialed.

Call Vectoring

In addition to System List availability which is authorized separately, the Abbreviated Dialing feature permits voice terminal users to access a maximum of two Abbreviated Dialing lists: two group lists, two personal lists, or one group list and one personal list. The vector-group list counts as one of these abbreviated dialing lists. Therefore, if a voice terminal user is designated as the controller of the vector-group list, that user can have access to only one other personal or group list.

Call Vectoring group-list items *must not* contain special function characters (for example, pause, wait, and mark). The networking features (AAR, ARS, or WCR) automatically control the timing for routing calls to the destinations of "route to" commands.

Centralized Attendant Service

A backup voice terminal user cannot extend a call using the Abbreviated Dialing feature.

Data Call Setup

With keyboard dialing, Abbreviated Dialing can be used to originate data calls. On switches prior to Release 2, Version 4, Abbreviated Dialing lists are assigned based on extension numbers. Data terminals (that have their own extension numbers) can have their own Abbreviated Dialing lists. In Release 2, Version 4, and DEFINITY Generic 2, Abbreviated Dialing lists are assigned based on equipment locations. For these switches, data terminals share a common ELL with their associated voice terminals (in a Voice/Data Station arrangement) and consequently share Abbreviated Dialing lists.

Display—Voice Terminal

When AD (Automatic Dial) Buttons or list-based Abbreviated Dialing is used on a terminal with display capability, the stored number is displayed. If the Names Data Base is loaded for the number, the associated name immediately replaces the number on the display. For ISDN—BRI terminals with displays, the timing of displayed information is different. The outpulsed numbers and subsequent Names Data Base information is not displayed until outpulsing has been completed and the OR (Originating Register) is released.

If the stored number includes a security code, such as a password or authorization code, this code will also be displayed unless it is protected in the list by the ***suppress function*** code. If the suppress function code is used, the protected characters (except "*" or "#") will be replaced on the display by "s," indicating that a character has been suppressed.

The suppress function code is available on Version 3 and later switches.

Beginning with the R2 V3 (Issue 1.2 tape), a user can press an AD button while the voice terminal is on-hook, to display the stored number for that button. The display shows "AD=stored number."

Beginning with R2 V4, if the AD button points to an Abbreviated Dialing list item (system, group, or personal list), the display shows the contents of the list item.

Extension Number Portability

After an extension is ported from one node to another within a portability subnetwork, frequent callers to this extension need not change the contents of their Automatic Dialing buttons or Abbreviated Dialing list items. When Abbreviated Dialing is used to call the usual extension number, the switch automatically routes the call to the new node in the subnetwork.

Extension Number Steering

Extension Number Steering allows from one to four steering digits to be associated with a trunk-group or feature dial access code for call routing within a Main / Satellite complex. Abbreviated Dialing can use these steering digits (as stored numbers) to access a trunk group or a feature (from a terminal at a local or a distant switch within the complex). However, these steering digits cannot be used as controlling extension numbers for abbreviated dialing lists. This is because the steering digits, rather than being actual extension numbers, represent trunk or feature dial access codes.

Hot Line

Abbreviated Dialing must be provided in order to provide Hot Line, since Hot Line is implemented using a personal list.

ISDN—BRI (Basic Rate Interface)

Generally, Abbreviated Dialing works the same for ISDN—BRI terminals as it does for DCP terminals on the DEFINITY Generic 2 switch. In both cases, more than one terminal can be assigned to a single ELL (Equipment Line Location). For example, in a voice / data station configuration, a voice terminal and a data terminal share the same ELL. When both terminals are assigned to the same ELL, they both appear to the switch (and to the Abbreviated Dialing feature software) as the same terminal. As a result, both terminals (voice and data) share the same abbreviated dialing lists.

Redial Button: ISDN—BRI voice terminals have Redial Buttons. This button functions in a manner similar to the LND (Last Number Dialed) feature, except that Abbreviated Dialing Station-to-Station calls that are recorded and redialed by the LND feature are not recorded and redialed when the Redial Button is used. (See the LND feature interaction.)

IPA (Interpartition Access)

Abbreviated Dialing list items are not checked for legality at the time they are entered. Illegal list items are accepted into the list, but calls to these illegal numbers are denied when the digits are outpulsed by the switch. When a user attempts to use Abbreviated Dialing to place a call over another partition's trunk group or to an extension in another partition group, the switch returns intercept treatment to the calling party.

LND (Last Number Dialed)

The Last Number Dialed feature will record and redial number called using the Abbreviated Dialing feature as follows:

- Station-to-Station Calls: When a local extension number is called using Abbreviated Dialing the LND feature will redial that call exactly.
- Station-to-Trunk Calls:
  - Complete AD button and list-stored Abbreviated Dialing numbers are *not* recorded or redialed by the LND feature. Instead, the previous manually dialed number is retained and redialed when LND is used
  - When an access code (AAR, ARS, WCR or trunk group dial access code) is manually dialed, and then an Abbreviated Dialing number (either button- or list-stored) is used, the LND feature records (and can be used to redial) the feature or trunk group access code that was manually dialed. However, the contents of the Abbreviated Dialing button or list item are not recorded or redialed. These calls can be dialed by pressing the LND feature button followed by the appropriate AD button or dialing the Abbreviated Dialing list access and index number. *(See also* Redial Button under the ISDN—BRI feature interaction.)

Look-Ahead Interflow

Look-Ahead Interflow "route to" steps share the Abbreviated Dialing group list that is designated as the vector-group list for Call Vectoring (in Procedure 030, Word 1).

To use one of these list items (from 1 to 95) for Look-Ahead Interflow, the list item is programmed to contain a VDN at the receiving switch. In turn, the vector assigned to the receiving VDN performs the inflow processing for the receiving switch.

The following are the acceptable formats for programmed digit strings for Look-Ahead Interflow.

1. 1-Digit AAR Access Code + RN(X) (Location Code) + XXX(X) (VDN at Receiving switch)
2. 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + NXX (Office Code) + XXXX (VDN at Receiving Switch)
3. 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + NPA (Area Code) + NXX (Office Code) + XXXX (VDN at Receiving Switch)

4. 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + International Telephone Number (Including a VDN at Receiving Switch)
5. If the sending and receiving switches belong to the same DCS and/or ENP subnetwork, XXXXX (VDN at receiving switch)
6. 1- to 4-Digit WCR Access Code + ("1")(Prefix Digit if required) + the appropriate World Class Routing digit strings.

Whenever a vector-group list item for the Look-Ahead Interflow feature contains either a ***null*** or ***invalid*** digit string, vector processing at the sending switch either treats the "route to" step as a "stop" step (if the final effective step) or continues with the next sequential step in the vector.

Personal CO (Central Office) Lines

The Abbreviated Dialing feature can be used to place calls over Personal CO lines that are administered as "rotary out."

The Abbreviated Dialing feature cannot be used to place calls over Personal CO lines that are administered as "touch-tone out." In this case, the serving CO scans Personal CO lines to detect an off-hook signal. After the CO recognizes an off-hook, the CO returns dial tone to the calling party. The local switch software is not involved in this call-origination process. So, during the process, the local switch does not invoke an originating register. Without an originating register, the switch cannot outplse the stored digits.

Precedence Calling

The Abbreviated Dialing feature is fully compatible with the Precedence Calling feature. That is, Precedence Calling dialing sequences, including the dial access code and precedence level digit, can be stored in an Abbreviated Dialing storage location and then used to place precedence calls.

Remote Access

When "Global" access is granted to the System List, Remote Access users as well as local terminal users can use the System List. This is a change that applies to System 85, Release 2, Version 4 and DEFINITY Generic 2 switches and can be made available to Version 3 switches.

Tenant Services

Abbreviated Dialing list items are not checked for legality at the time they are entered. Illegal list items are accepted into the list, but calls to these illegal numbers are denied when the digits are outpulsed by the switch. When a user attempts to use Abbreviated Dialing to place a call over another partition's trunk group or to an extension in another partition, the switch returns intercept treatment to the calling party.

A voice terminal user from any partition can access the system list, but the switch will only allow completion of calls that are consistent with the dialing capability of that user's partition.

Each partition can have a group list that every voice terminal user in the partition can access. This would provide the equivalent of the system list on a per-partition basis. Again, the switch will deny calls using list items that are not consistent with a partition's dialing capabilities.

Touch-Tone Calling Senderized Operation

The Touch-Tone Calling Senderized Operation feature is a prerequisite for implementing Abbreviated Dialing.

WCR (World Class Routing)

The Abbreviated Dialing feature works with the World Class Routing feature in the same way as with the earlier networking features, AAR and ARS. A network dialing digit sequence can be stored in an abbreviated dialing storage location and then used to place a WCR network call.

Hardware Requirements

None.

Feature Administration

Assignment of the Abbreviated Dialing feature is on a per-system basis. Abbreviated dialing access and permissions are assigned on a per-terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

The Abbreviated Dialing feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

Administration Procedures Abbreviated Dialing Feature			
Procedure	**Word**	**Purpose**	**SMT**
059	1	Assigns access to the system list, access to a group list, and assigns personal lists to voice terminals. Assigns the list size and list controller for group / personal lists.	Yes
059	2	Assigns list-stored numbers.	Yes
059	3	Assigns the list-access and special function buttons to voice terminals.	Yes
059	4	Assigns button-stored numbers to voice terminals.	Yes
059	5	Displays the amount of space in the switch for additional lists.	Yes
275	3	Enables system-wide access to the system list. Also assigns the system-list size.	Yes
350	1	Assigns the first digit of the Abbreviated Dialing feature access codes (if required).	No
350	2	Assigns the dial access codes for the Abbreviated Dialing feature. The applicable encodes are as follows: 90 Dial the system (global) list-touch-tone terminal 91 Dial list A—touch-tone terminal 92 Dial list B-touch-tone terminal 93 Program auto dial number or list 94 Dial the system list—rotary terminal 95 Dial list A—rotary terminal 96 Dial list B—rotary terminal.	No

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Abbreviated Dialing	
Path Name	**Purpose**
terminal-change extensions attributes	Assigns access to group list(s), personal list(s), and the system list to an extension number.
terminal-change group abbreviated dialing	Assigns the size of a list and list-stored numbers.
terminal-change system parameters (select the List-Administration option)	Enables system-wide access to the system list.
terminal-change terminal buttons	Assigns the Abbreviated Dialing feature buttons to voice terminals.

Notes:

ACCUNET® Service Interface

Description

This feature provides an interface between the System 85 or DEFINITY Generic 2 switch and the AT&T Communications ACCUNET® Switched Digital Service. ACCUNET Service Interface is similar to the DS1 Interface feature. Modem Pooling is not needed for data communications over ACCUNET service. Preindication *(see* the Data Call Setup feature) is not necessarily needed (but is recommended) on System 85 switches, however, Data Preindication is required on DEFINITY Generic 2 switches.

For many users, with a part-time requirement for high-speed data or other wide band carrier services, the AT&T ACCUNET Switched Digital Service offers an attractive alternative to a dedicated, full-time facility. ACCUNET Switched Digital Service offers billing on a use and distance basis rather than requiring the customer to pay for a full-time facility that is only used part time.

ACCUNET Switched 56 Service

ACCUNET Switched 56 Service is a switched 56 Kbps, digital end-to-end, public (subscriber) networking service available from AT&T Communications. A special in-band signaling mode is used to provide 56 Kbps access to ACCUNET Switched 56 Service. ACCUNET Switched 56 Service may or may not require local Central Office (CO) switching functions. The specific requirements vary from one region to another. If ***AT&T Direct Access*** is available, there may be some economic advantage to connecting directly into the AT&T Communications network. Evaluation of this option must be made on a case-by-case basis.

Feature History and Development

The ACCUNET Service Interface feature was first available on System 85 in Release 2, Version 3.

Effective January 1990, the network requires a synchronization system that is more stable than the previous standard clocking synchronization system used by PBXs such as System 85, Release 2. This new requirement is met by an optional adjunct, the Synchronization Clock, which provides a Stratum 3 clocking synchronization capability. The Synchronization Clock can be used with System 85, switches (Release 2, Version 3 and later) and with DEFINITY Generic 2 switches. This adjunct is optional in that if the switch does not use the ACCUNET Service Interface feature, it is not required at this time. Also, with Generic 2, the introduction of the Bearer Capability feature provides an alternative that may be more efficient and more economical.

User Operations

ACCUNET Switched 56 Service

ACCUNET Switched 56 Service is accessed using Voice Terminal Dialing procedures from the Data Call Setup feature. Some special settings are required on the data module (MPDM / ACCUNET). Keyboard dialing procedures from the Data Call Setup feature cannot be used for ACCUNET Switched 56 Service. Applicable user operations are as follows:

Setting the Data Module:

1. Set the ANET option switch to ON.
2. Set the Data Rate selection to 56K

Placing an ACCUNET Service Call From a Voice Terminal:

1. At the originating voice terminal, go off-hook. [Local dial tone]
2. Dial the ACCUNET Service Interface access code and network number.
3. If One Button Transfer is available:

   a. Press the appropriate **[DATA]** button.

   or

   If One Button Transfer is not available:

   a. Press **[TRANSFER]** .

   b. Dial the extension number for the data module to be used.

   c. Press **[TRANSFER]** again.

   d. Go on-hook.

Considerations

Billing Provisions

For billing purposes, one listed directory number must be assigned to the ACCUNET Service Interface feature.

DNHR (Dynamic Nonhierarchical Routing)

ACCUNET Service Interface calls are routed through the AT&T Switched Network using the DNHR routing algorithm. DNHR routing, in contrast to the older "hierarchical" method of public-network routing, quickly routes data calls through as few as two (and never more than three) 4 ESS™ switches acting as peers in the AT&T Communications Network. Since the DNHR algorithm can have up to 13 alternate routes selected on a look-ahead basis, this method of routing also minimizes blocked calls. Even during high traffic periods, blocked calls should not exceed 1 percent of the traffic volume.

No Satellites for Domestic Calls

Some private and public networks commonly route long-distance calls through orbiting satellite(s). Using satellites to route calls can cause noticeable and annoying delays in a data transmission. By default, the AT&T Switched Network routes domestic ACCUNET Switched Digital Service calls over terrestrial facilities.

Synchronization Clock System

Effective January, 1990, ACCUNET Switched Digital Service requires a Stratum 4, Type I clocking and synchoronization system. The standard synchronization system provided by the system clock synchronizer (SCS) does not meet this requirement. A Stratum 3 clocking and synchronization system, the Synchronization Clock, is available as an optional adjunct to meet this network requirement. When the Synchronization Clock system is used, its interface circuit occupies the same slot that would otherwise be occupied by the SCS. The SCS must not be administered (Procedure 250, Field 11 is set to "0") when the Synchronization Clock system is used The Synchronization Clock adjunct and the network synchronization system are described in detail in the DS1 Interface feature.

Interactions with Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

ARS (Automatic Route Selection)

The ARS feature is compatible with the ACCUNET Interface feature and on System 85 or Generic 2.1 switches, if ACCUNET Interface calls are to be routed over ISDN—PRI facilities, the ARS feature must be used to perform call routing.

Bearer Capability

The Bearer Capability feature is compatible with the ACCUNET Service Interface feature. However, the default for the trunk types used with the ACCUNET Service Interface feature (trunk types 108 and 109) is Bearer Capability Class of Service (BCCOS) 1. BCCOS 1 is for Mode 2 data, and for ACCUNET Service (which uses Mode 1), this is not the best choice. Among the predefined BCCOSs, BCCOS 7 (Mode 1) more closely matches the needs of ACCUNET Service Interface, and for specific sites, it may be most desirable to locally define a BCCOS for this application. On DEFINITY Generic 2 switches, ACCUNET Service Interface trunks should be specifically administered to assign either BCCOS 7 or an appropriate, locally defined BCCOS.

Data Call Setup

The Voice Terminal Dialing option of the Data Call Setup feature is used to access the ACCUNET Switched 56 Service network.

Keyboard dialing, as described in the Data Call Setup feature, does not work for ACCUNET Service calls.

DMI (Digital Multiplexed Interface)

The DMI feature BOS version is not compatible with the ACCUNET Service Interface feature.

The DMI feature MOS version is compatible with the ACCUNET Service Interface feature, at data rates of 56 Kbps or less. However, the DMI feature is capable of operating at data rates up to 64 Kbps. Above 56 Kbps, ACCUNET Service Interface and DMI MOS are not compatible.

DS1 (Digital Service 1) Interface

The ACCUNET Service Interface feature uses the DS1 Interface feature as its trunking service. ACCUNET Service Interface is a specific application of the DS1 Interface feature.

Host Computer Access

The Host Computer Dialing function of the Host Computer Access feature cannot be used to place calls through the ACCUNET Service Interface feature. This is because host computer dialing uses the keyboard dialing function of the Data Call Setup feature, and keyboard dialing data call setup is not compatible with ACCUNET Service Interface.

ISDN—BRI (Basic Rate Interface)

The ISDN—BRI feature (DEFINITY Generic 2) is not compatible with the ACCUNET Service Interface feature. The ISDN—BRI asynchronous data module (ADM) does not support either 56 Kbps operations or mode 1 data operations in the DEFINITY Generic 2 time frame.

ISDN—PRI (Primary Rate Interface)

The ACCUNET Service Interface feature works with the ISDN—PRI feature either through the *interworking* function or via the ISDN Call-by-Call Service Selection function. Both interworking and call-by-call service selection are described in some detail in the ISDN—PRI feature. Call-by-Call Service Selections is also known as Network Specific Facilities (NSF).

There is one case where a problem can be encountered ACCUNET Service calls are limited to a maximum data rate of 56 Kbps. There is no problem processing such calls through the System 85 or DEFINITY Generic 2 switch to PRI facilities. ISDN calls, however, can run at data rates as high as 64 Kbps. If an ISDN call is received that is using a data rate higher than 56 Kbps, it cannot be passed to ACCUNET Service facilities and is blocked by the switch.

WCR (World Class Routing)

The WCR feature is compatible with the ACCUNET Interface feature and on DEFINITY Generic 2.2 switches, if ACCUNET Interface calls are to be routed over ISDN—PRl facilities, the WCR feature must be used to perform call routing.

Hardware Requirements

The ACCUNET Service Interface feature requires the following additional or special hardware.

For Traditional Modules:

- DS1 Interface Unit, consisting of:
  - DS1/73 Series Port Carrier

    Each DS1/73 series port carrier can accommodate two DS1 circuit packs.
  - ANN11C (or later) DS1 Interface Circuit Pack

    Each ANN11 circuit pack supports one 24-channel DS1 interface. For ACCUNET Service Interface applications, the ANN11 is set up as trunk type 108 or 109 using bit rubbed signaling and is located in slot 5 or slot 18 of the DS1 port carrier.
  - ANN35 ISDN Primary Rate Port

    If ACCUNET Service Interface is provided over ISDN—PRI facilities, the ANN35 is needed to provide ISDN—PRI trunking service. The ANN35 is a single circuit pack that supports the 23 bearer channels and the D-channel called for in the ISDN—PRI standard. For the NFAS confirmation, the ANN35 can also be configured as 24 B. It also provides termination facilities for the ISDN levels 1 and 2 protocol. Note that while physically similar, this circuit pack is functionally different from the standard DS1 interface circuit packs (ANN11C, 11D, and 11E).
  - SN261C ADFTC (Analog/Digital Test Circuit)

    The ADFTC supports both automatic (self tests) and time available trunk testing.
  - TN380D, Module Processor

    If ISDN—PRI trunks are used, the TN380D (Module Processor) must be used in place of the TN380B or TN380C for ISDN—PRI applications. The TN380D provides for the message signaling format and larger processor instruction strings required by the ISDN feature.

For Universal Modules:

- Common Port Carrier

- TN767, DS1 Interface circuit pack.

  Each TN767 circuit pack provides 24 channels for DS1 (or ISDN—PRI) interface. This circuit pack is the equivalent of the ANN11C (used on the Traditional Module) when used for DS1 interface. When used in conjunction with the TN555, the pair of circuit packs is functionally equivalent to the ANN35 used on a Traditional Module.

- TN555 DS1 Packet Adjunct Circuit Pack

  If ISDN—PRI trunks are used for ACCUNET Service Interface, the TN555 DS1 packet adjunct circuit pack provides packet handling service for the TN767 DS1 interface circuit pack to support ISDN—PRI messaging requirements. The TN555 DS1 packet adjunct adjunct pack, in combination with the TN767 DS1 interface circuit pack provides the functional equivalent of the ANN35 used with the traditional module.

- TN771B MTCP (Maintenance Test Circuit Pack)

  The MTCP supports both automatic (self test) and demand testing. The MTCP is equivalent to the ADFTC on the traditional module and is first available with Generic 2.1 Issue 2.0.

- UN154B Universal Bus Interface

  If the TN771B is used, the port carrier must be equipped with the UN154B Universal Bus Interface. This is an updated version from the UN154 used with initial releases of the universal module. Earlier Generic 2 switches can use the TN771B only if they are also upgraded with the UN154B.

Regardless of Module Type:

- MPDM/ACCUNET (Modular processor Data Module / ACCUNET)

  The MPDM/ACCUNET is also known as the MPDM/M1*. In this case, the asterisk (*) is part of the name and not a foot note call.

- 551 T1 Channel Service Unit (CSU)

  The 551 T1 CSU may be provided by the local exchange or switching office, in which case it is not part of the System *85* or DEFINITY Generic 2 hardware configuration.*

  Also, the CSU could be replaced by a 740/741 multiplexer with a built-in CSU, or a DR23 microwave can be used to directly connect to the AT&T Service Node.*

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* The alternatives to the basic 551 T1 CSU, may present some physical connection problems when the Stratum 3 Synchronization Clock is used. The available cabling is desired to connect a local CSU to the Yellow Cross-connect field Any other arrangement may require customized cabling.

- Synchronization Clock (Stratum *3)*.

  This unit is available for System 85, Release 2, Version 3 and later switches and for DEFINITY Generic 2 switches. It provides a local, external ***Stratum 3*** clock interface. This adjunct consists of the following

  — TN2131, External Clock Interface Circuit Pack

    Replaces the TN463 System Clock Synchronizer (SCS) Circuit Pack and mounts in the switch cabinet and slot that would normally house the SCS (either the TMS or Module Control cabinet). This varies depending on the switch vintage and configuration *(see* the DS1 feature for more details).

  — Synchronization Clock, J58909A.

    The Stratum 3 Clock adjunct is available only in a Stand alone, duplex (duplicated) configuration, mounted in an AUDIX small cabinet.

    This configuration is available in either an AC-powered or a DC-powered version.

  — Connecting Cables.

    The Synchronization Clock system requires the use of three special connecting cables. The specific cables required depend on the switch configuration.

    Multi-Module TMS Control or Single Module Traditional Module Control

      H-600-260

      H-600-274

      H-600-293.

    Single Module Universal Module Control

      H-600-271

      H-600-274

      H-600-293.

Feature Administration

Assignment of the ACCUNET Service Interface feature is on a per-system basis and by trunk group within the switch.

On System 85 switches, this feature is administered using the Maintenance and Administration Panel (MAAP). The customer can partially administer this feature using the System Management Terminal (SMT).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using Manager IV.

The following are the applicable administration procedures.

Administration Procedures **ACCUNET Service Interface Feature**			
Procedure	**Word**	**Purpose**	**SMT**
100	1	Assigns trunk-group dial access code, trunk type, and dial access restriction. The applicable trunk types are: Field 6 = 108 or 109.	No
100	2	Assigns the BCCOS to trunk groups, Field 2 (DEFINITY Generic 2 only). BCCOS 7, or an appropriate locally defined BCCOS, is recommended for ACCUNET Service trunks.	No
100	3	Assigns signaling type (other than default signaling) to trunk groups. For ISDN—PRI trunks use signaling type 20.	N/A
101	1	Assigns trunk-group characteristics for System 85 switches. For ACCUNET Service Interface, rotary dial signaling was at one time required, however, this requirement now depends on the capabilities of far end equipment. Touch-tone signaling can now be used in most cases.	No
101	2	Administers the trunk redial delay timer.	No
103	1	Assign trunk group translations for network trunks (for example, Data Protection, FRL, Bridge-On).	Yes*
115	1	Assigns trunk group termination (ACD, CAS, etc).	No
116	1	Assigns trunks to a DS1 Channel. For DEFINITY Generic 2 (only), also assigns D-channel group number if appropriate (ISDN—PRI NFAS arrangements).	No
250	1	Assigns (or unassigns) the SCS circuit to either the Module Control Carrier or the Time Multiplexed Switch Carrier. When the Synchronization Clock is used, Field 11 must be set to "0."	No
* Display only procedure for SMT.			

(Continued)

Administration Procedures ACCUNET Service Interface Feature *(Continued)*			
Procedure	**Word**	**Purpose**	**SMT**
260	1	Assigns DS1 circuits to equipment locations and assigns signaling requirements and other circuit characteristics. For ACCUNET Service Interface, the following encodes apply: Field 8 = 1 (Robbed Bit Signaling) Non-ISDN trunks or Field 8 = 0 (24th Channel Signaling) for ISDN trunks Field 12 & 13 = 0 (No SCS) when the Synchronization Clock is used Field 14 = 0 (Trunks / Mixed Application - System 85; DS1 trunks, OPS - DEFINITY Generic 2) Field 14 = 5 (ISDN—PRI, DMI-MOS).	No
275	1	Assigns system class of service features such as trunk-to-trunk calling and CDR.	Yes
279	1	For DEFINITY Generic 2: Where ISDN—PRI trunks are used, assigns the ISDN Network Specific Facilities (NSF) codes. Applicable entries are: Field 1 = 357 (Standard for R2V4) Field 2 = 1 (Binary) Field 3 = 1 (Service) Field 4 = 6 (ACCUNET switched digital).	N / A
290	1 & 2	Displays circuit assignments (equipment locations) and identification information.	Yes
309	5	For System 85 and Generic 2.1 switches, assigns ISDN trunk type, BCCOS, and NSF to ARS routes when used for ACCUNET Service Interface. Applicable encodes are: Field 4 = 108 or 109 Field 5 = 357 (or 999) Field 6 = 7 (BCCOS 7) or a local BCCOS.	N / A
318	1 & 2	For Generic 2.2 switches, defines network routing patterns and preferences including trunk group, ISDN Sending Index, and BCCOS.	N / A

(Continued)

Administration Procedures ACCUNET Service Interface Feature *(Continued)*			
Procedure	**Word**	**Purpose**	**SMT**
321	5	For System 85 and Generic 2.1 switches, assigns ISDN trunk type, BCCOS, and NSF to routing patterns when used for ACCUNET Service Interface. Applicable encodes are: Field 3 = 108 or 109 Field 4 = 357 (or 999) Field 5 = 7 (BCCOS 7) or a local BCCOS.	N/A
322	1	For Generic 2.2 switches, defines an ISDN Sending Index. The ISDN Sending Index specifies selected ISDN—PRI messaging requirements including the NSF IE to be sent.	N/A

The following is the applicable TCM path name used with the AP 16.

TCM Screen - ACCUNET Service Interface	
Path Name	**Purpose**
terminal-change terminal equipment	Assigns an extension number to the DS1 Interface. The DS1 Interface must first be assigned to an equipment location by the installer.

Advanced Private Line Termination

Description

The APLT (Advanced Private Line Termination) feature provides access to and termination from the following private-line networks:

- CCSA (Common Control Switching Arrangement)
- EPSCS (Enhanced Private Switched Communications Service).

The APLT feature allows network inward dialing and direct outward dialing to distant network locations. Incoming network calls are processed the same as a DID (Direct Inward Dialing) call. Active features at the called terminal such as Call Forwarding and Call Waiting operate as usual *(see* Interactions with Other Features for exceptions).

These private networks provide call routing over dedicated facilities. Based on the network numbering plan, on-network calls are routed to terminals or attendants within the network. Private networks can optionally provide off-network calling to a desired destination based on the 7- or 10-digit public-network number.

EPSCS Networks

If the private network is an EPSCS network, services may include tandeming through the distant switch to available outgoing trunks such as local CO (Central Office), FX (Foreign Exchange), or WATS (Wide Area Telecommunications Service) Access trunks as well as to on-net terminals. When an EPSCS network is used, an authorization code is sometimes needed to complete the network call. This requires dialing additional digits after the outgoing trunk is seized. Based on the authorization code, the network either denies or completes the call.

CCSA Networks

If the private network is a CCSA network, tandeming through the distant switch is not possible. With a CCSA network, attendant assistance is required to pass a call through a distant switch to a second distant switch. In this type of network, authorization codes are not generally used because of the intervention of attendants when extended calling services are required.

Attendant positions are assigned network LDNs (Listed Directory Numbers) that are different from the public-network LDNs. Network LDNs consist of the private-network office code and a 2-, 3-, 4-, or 5-digit number in the dialing plan including 01, 011, 0111, and 01111. Incoming attendant-seeking calls are identified as network calls by the Attendant Display feature.

Feature History and Development

This feature was first available with System 85 in Release 1. At that time, it was part of the Private Network Access feature. It was reintroduced under the name Advanced Private Line Termination in Release 2, Version 3. No other changes have been made.

User Operations

The following are the user operating procedures for this feature.

Direct Outward Dialed Network Calls:

1. Dial the network access code. [Network dial tone]
2. Dial the destination code as defined by the network dialing plan. For example:
   a. Network service access code (usually 3-digits)
   b. A 7- or 8-digit destination number for an on-network call
   c. A 10-digit destination number for an off-network call
3. If an authorization code is required, dial the network authorization code. [Call-progress tone, typically, ringback tone]

Outward Dialed Network Calls Via Tie Trunk to Main:

1. Dial the access code of the tie trunk group. [Dial tone]
2. Dial the network access code if required. [Network dial tone]
3. Dial the destination code as defined by the network dialing plan.
4. If an authorization code is required, dial the network authorization code. [Call-progress tones, such as, ringback tone, busy tone, etc.]

Incoming Attendant Seeking Network Calls:

1. Obtain destination information from calling party (ICI identifies the call as a network call).
2. Extend the call by using DXS or Attendant Direct Trunk Group selection features,

   or

   Press **[START]** (calling party put on hold). [Dial tone]
3. Dial the desired extension number or trunk-group access code.
4. Press **[RELEASE]** (calling party hears call-progress tones). [Attendant dropped from connection.]

   or

   If further attendant assistance is needed, press **[HOLD]** . [Call is held on the console.]

Considerations

Access Codes

In System 85 and DEFINITY Generic 2.1 switches, the APLT feature can be implemented through the AAR (Automatic Alternate Routing) or ARS (Alternate Route Selections) features. When AAR/ARS is used, a separate network or trunk-group access code is not required for use of the APLT feature. On DEFINITY Generic 2.2 switches, APLT can be implemented through the World Class Routing feature

Restricting Feature Use

Individual terminal access to this feature and access to individual terminals via this feature can be restricted by the Voice Terminal Restrictions feature and the Attendant Control of Voice Terminals feature.

Access to private-line trunk facilities can also be restricted by using the following:

- Attendant Control of Trunk Group Access
- Authoriation Codes
- Facilities Restriction Levels
- Miscellaneous Trunk Restrictions
- Trunk-to-Trunk Restrictions.

Interactions with Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

AAR/ARS (Automatic Alternate Routing/Automatic Route Selection)

The APLT feature is compatible with the AAR and ARS features. Incoming APLT calls can route to the AAR or ARS feature for subsequent routing. Outgoing calls can use APLT trunks as a routing preference. If a dial pulse station is assisted by the attendant in placing a call that routes to an APLT trunk, and if the APLT network requires that an authorization code be entered, the attendant must enter the authorization code for the caller. The (dial pulse) station user cannot enter authorization code digits after the attendant releases from the call.

Bridged Call

The APLT off-net and DID restrictions are assigned to an extension class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. Then the the APLT and DID restrictions are assigned to a ***shared extension,*** the restrictions apply to every image of the extension.

CDR (Call Detail Recording)

- Incoming Tie Trunk Calls

  If the caller dials an account code and then calls across a tie trunk to an attendant on another network node, the account code is recorded at originating node only.

  If desired, the attendant at the terminating node can enter the account code to record the account code at the terminating node.

- Outgoing Tie Trunk Calls

  If the caller dials a CDR account code and then calls the attendant, the account code is tipped. The attendant can reenter the account code before extending the call across the tie trunk.

ISDN—PRI (Primary Rate Interface)

Although ISDN—PRI can be used for private network service, APLT trunk types (12 through 15) cannot be assigned signaling type 20 *(see* Tables F-A and F-B in Appendix F). APLT and ISDN—PRI are not compatible features.

Look-Ahead Interflow

ISDN—PRI trunk groups are required for Look-Ahead Interflow. Since trunk types (12-15) cannot be assigned to ISDN—PRI trunk groups, APLT trunk groups cannot be used to route Look-Ahead Interflow calls.

Precedence Calling

The Precedence Calling feature uses APLT trunk types for connection to the AUTOVON, however, Precedence Calling does not use the APLT feature for routing.

Route Advance

APLT trunks (trunk types 12 through 15) can be included in Route Advance patterns. When the APLT feature is accessed through one of the other network routing features (AAR, ARS, or WCR), Route Advance patterns are ignored if encountered. That is, if a trunk group is accessed through AAR, ARS or WCR, the AAR, ARS, or WCR routing pattern is followed. If the trunk group is also part of a Route Advance pattern, the Route Advance pattern is not followed. However, if a trunk group is accessed using a trunk group dial access code and is part of a Route Advance pattern, the Route Advance pattern is followed if no available trunk is found in the first trunk group accessed. This is true even if the trunk group accessed is part of an AAR, ARS or WCR routing pattern.

WCR (World Class Routing)

The APLT feature works with World Class Routing as it did with the earlier networking features, AAR and ARS. Incoming APLT calls can route to the WCR feature for subsequent routing. Outgoing calls can use APLT trunks as a routing preference. If a dial pulse station is assisted by the attendant in placing a call that routes to an APLT trunk, and if the APLT network requires that an authorization code be entered, the attendant must enter the authorization code for the caller. The (dial pulse) station user cannot enter authorization code digits after the attendant releases from the call.

Hardware Requirements

The following specific hardware items are required to implement this feature.

For Traditional Modules:

- SN233 Tie Trunk Circuit Pack (four circuits per circuit pack)

For Universal Modules:

- TN760 Tie Trunk Circuit Pack (four circuits per circuit pack).

Applicable Trunk Types

Trunk types 12 and 13 are used for wink start with dial tone, delay dial with dial tone, or dial tone operation. Trunk types 14 and 15 are used only for through dial operation.

Feature Administration

Assignment of the APLT feature is on a per-system basis.

On System 85 switched, this feature is administered using the Maintenance and Administration Panel (MAAP). The customer can partially administer this feature using the System Management Terminal (SMT), the Terminal Change Management (TCM) feature, or the Facilities Management (FM) feature.

On DEFINITY Generic 2 switch, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager lV.

The following are the applicable administration procedures.

Administration Procedures Advanced Private Line Termination			
Procedure	**Word**	**Purpose**	**SMT**
010	3	Assigns miscellaneous trunk restriction group, APLT off-net permissions, DID and FRL to an extension class of service.	Yes
100	1	Assigns the dial access code, dial access restriction, and public network access and egress to trunk groups. For System 85 R2 V3 and earlier switches, also assigns route advance. The applicable APLT trunk types include the following: 12 through 15 = CCSA / APLT 32 through 47 = ETN Tie trunks.	No
100	3	Where the default signaling type is inappropriate, assigns alternate signaling types to trunk groups.	No
100	4	For System 85, R2 V4 and DEFINITY Generic 2 switches, assigns Route Advance patterns.	N / A
101	1	Assigns trunk characteristics such as APLT features allowed and CDR to private-network trunk groups.	No
102	1	Assigns trunk-group access codes to a trunk restriction group.	Yes
103	1	Assigns network features associated with a trunk group including FRL.	Yes*
110	1	Assigns a restricted dial code entry number to a dial access code.	No
111	1	Assigns a restricted dial entry code to a trunk group.	No
150	1	Assigns private-network trunks to equipment locations.	No
275	1	Assigns system class of service and features such as automatic identification of outward dialing tandem tie-trunk switching, and trunk-to-trunk connections.	Yes
350	1 & 2	Assign feature dial access codes as required.	No
* Display only procedure for the SMT.			

The following are the applicable TCM path names used with the AP 16.

TCM — Advanced Private Line Termination	
Path Name	**Purpose**
terminal-change class-of-service attributes	Assigns miscellaneous trunk restrictions to an extension class of service. These restrictions include Inward, Outward, and Full. This screen also assigns an FRL to an extension class of service and APLT on or off.

The following are the applicable FM path names used with the AP 16. A printed report of the displayed information can also be generated.

FM Screens — Advanced Private Line Termination	
Path Name	**Purpose**
facilities-mgmt routing conversion	Displays and changes the correspondence between a private-network location code and the public-network telephone number that routes to that location code (refer to the ARS feature description for an explanation of 10-Digit Conversion). Either the location code or the public-network destination code (telephone number) can be entered, and the corresponding values are displayed.
facilities-mgmt facility-restriction	Displays and assigns FRLs to extensions, incoming tie trunks from subtending switchs, and authorization codes. Also assigns network access to authorization codes.

Notes:

Attendant Auto-Manual Splitting

Description

This feature allows an attendant to privately identify the calling party to the called voice terminal user. The caller is automatically "split away" from the attendant and the called party when the attendant presses either the START or a DXS (direct extension selection) button. The feature is activated manually by pressing the SPLIT button provided on the attendant console. Pressing the SPLIT button changes the condition from split to unsplit or from unsplit to split.

A split condition remains in effect until the attendant manually removes the split condition or one of the parties disconnects.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Automatically Split From a Call:

1. Be sure there is a 2-way connection.
2. Press **[START]** ,

   or

   Press a DXS (direct extension selection) button.

To Manually Split From a Call:

1. Be sure there is a 2-way connection.
2. Press **[SPLIT]** .

To Cancel a Split Condition

If the called voice terminal is busy or the called party refuses the call:

Press the **[CANC]** button.

If the called party accepts the call:

Press the **[RELEASE]** button.

If the called party accepts the call and the attendant wishes to handle a subsequent call:

Press another loop button.

If the calling party has requested use of the Serial Calls feature:

Press the **[HOLD]** button.

If the attendant is needed for further assistance during the call:

Press the **[SPLIT]** button to allow the attendant and both parties to converse.

Considerations

Called Party Splitting

Splitting of the called party from the attendant and the calling party is not allowed.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Centralized Attendant Service

Manual splitting is inactive. The SPLIT lamp and button do not function on release link trunk calls.

Malicious Call Trace

If an attendant activates Malicious Call Trace while a calling party is split from the console, the split condition is disabled. On the attendant console, the split lamp goes out For the previously split party, the switch returns dial tone to an internal caller or disconnects an outside caller from the switch.

Privacy—Attendant Lockout

The Attendant Lockout feature overrides the Attendant Auto-Manual Splitting feature, and the SPLIT button becomes nonfunctional. Privacy is essentially a form of splitting that prevents the attendant from bridging onto a talking connection. The Attendant Lockout feature does not override splitting; however, it denies the attendant the ability to reenter an established connection held on the console position, unless recalled by a voice terminal. Attendant Recall Privacy works with the Splitting feature to cause automatic splitting. Manual splitting / unsplitting is disabled.

Serial Calls

If the Serial Calls feature is provided, the call is not split when an attendant presses the HOLD button to hold the call on the console.

Hardware Requirements

None.

Feature Administration

Assignment of the Attendant Auto-Manual Splitting feature is on a per-system basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance Administration Panel).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE ATTENDANT AUTO-MANUAL SPLITTING		
PROCEDURE	**WORD**	**PURPOSE**
203	1	Assigns the SPLIT button to the attendant console(s). The applicable encode is as follows: 4 SPLIT Button.

Notes:

Attendant Call Waiting

Description

This feature allows a call to wait when an attendant wants to extend that call to a busy single-appearance voice terminal. Allowing the call to wait limits the attendant work load by reducing the number of repeated attempts needed to reach a busy extension.

Two bursts of distinctive ringing tone signal the busy terminal user that a call is waiting. The busy terminal user can answer the call by momentarily pressing the switchhook and dialing the Call Waiting-answer hold access code or by going on-hook. This allows a single-appearance voice terminal to function like two terminals. A user, busy on a 2-party call can be alerted to the arrival of another call, and can answer the call if desired.

Feature History and Development

This feature was fit available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Process an Incoming Call

By the attendant to a busy voice terminal without DXS buttons:

1. Attendant receives a call. [ATND lamp lights, and ICI display lights.]
2. Press the appropriate loop button. [PA lamp goes out. The attendant and calling party are connected.]
3. Press **[SPLIT]** lamp lights, the calling party is split away from the attendant, and dial tone is heard.]
4. Dial the called voice terminal's extension. (The called voice terminal is busy.) [Confirmation tone is heard, the BUSY lamp lights, the SPLIT lamp goes out, and the attendant is reconnected to the calling party.]
5. Press **[RELEASE]** . (The calling voice terminal's call becomes a waiting call.) [The called voice terminal user bean the 2-burst tone, ATND lamp goes out, ICI display goes out, and timed reminder begins. If no calls are waiting , the PA lamp lights.]
6. The called party goes on-hook. [Busy lamp goes out, RING lamp lights, the waiting party hears ringback tone, and the called party beam 2-burst ringing.]
7. The called party goes off-hook. [RNG lamp goes out, and the calling and called parties are connected. The ANSWER lamp lights if the attendant is in the connection.]

By the attendant to a busy voice terminal with DXS buttons:

1. Attendant receives a call. [ATND lamp lights, and ICI display lights.]
2. Press the appropriate loop button. [PA lamp goes out. The attendant and calling party are connected.]
3. Press the associated DXS/BLF selection button. [SPLIT lamp lights, the calling party is split away from the attendant, and dial tone is heard.]
4. Press the appropriate DXS button. (The called voice terminal is busy.) [Confirmation tone is heard, the BUSY lamp lights, the SPLIT lamp goes out, and the attendant is reconnected to the calling party.]
5. Press **[RELEASE]** . (The calling voice terminal's call becomes a waiting call.) [The called voice terminal user hears the 2-burst tone, ATND lamp goes out, ICI display goes out, and timed reminder begins. If no calls are waiting the PA lamp lights.]
6. The called party goes on-hook. [Busy lamp goes out, RING lamp lights, the waiting party hears ringback tone, and the called party hears distinctive (2-burst) ringing.]
7. The called party goes off-hook. [RING lamp goes out, and the calling and called parties are connected. The ANSWER lamp lights if the attendant is on the connection.]

Considerations

Call Waiting Tone

The Call Waiting tone is two 100-millisecond, 400-hertz beeps.

Attendant Call Waiting Limitations

Attendant Call Waiting is limited to single-appearance voice terminals within the switch. Calls to multiappearance voice terminals route to idle appearances.

Attendant Call Waiting Denial

Attendant Call Waiting is denied when directed toward a busy single-appearance voice terminal that is not in a stable talking condition. See also the Feature Interactions with the Bridged Call, Call Forwarding—Busy and Don't Answer, Call Park, Conference, Data Protection, Hold, Recorded Telephone Dictation and Transfer features.

Hard and Soft Processor Swaps

If an attendant call or an attendant-extended call is waiting on a busy single-appearance terminal when a hard processor swap occurs, the waiting call does not endure the hard swap.

The Attendant Call Waiting feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

ACD (Automatic Call Distribution)

Attendant calls to a local ACD split are not queued. The switch attempts to complete the call to an idle agent either by scanning the agent queue or byscanning the split. if no idle agents are found in the split, the call waits on the split supervisor's single-appearance terminal. When an attendant places a call to an individual ACD agent's single-appearance terminal and that terminal is busy, the call waits on the busy terminal when Attendant Call Waiting is provided.

Attendant calls cannot wait on an ACD agent's single-appearance voice terminal while an observer (using agent override) is connected to the agent's call.

Bridged Call

Attendant Call Waiting is partially allowed toward shared extensions with only one appearance. When only the straight line set is active on this type of shared extension, the switch allows an attendant call to wait. However, when a multiappearance terminal is active on the shared extension, Attendant Call Waiting is denied. The switch returns busy tone.

While an attendant-extended call is waiting on a straight line set, a multiappearance voice terminal can be used to bridge onto the active call. However, the waiting call cannot be retrieved until the multiappearance voice terminal leaves the connection.

Attendant Call Waiting is denied toward shared extensions with more than one appearance. Instead, attendant calls are routed to an idle appearance (if available) of the shared extension. When every appearance is busy, the switch returns busy tone.

Busy Verification of Lines

An attempt to verify a terminal line via the Busy Verification of Lines feature takes precedence over the Attendant Call Waiting feature.

CDR (Call Detail Recording)

The CDR feature records the called extension number, as well as the account code, if dialed, for attendant-assisted incoming calls using Attendant Call Waiting.

Call Forwarding—Busy and Don't Answer

When Call Forwarding—Busy and Don't Answer is active at the called voice terminal, the forwading operation occurs before Attendant Call Waiting is allowed. There are four possible operations. If the originally called voice terminal is busy and the forwarded-to voice terminal is idle, attendant calls forward to and ring at the forwarded-to voice terminal. If the originally called voice terminal is busy and the forwarded-to voice terminal is busy, Attendant Call Waiting is denied and busy tone is returned. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is idle, attendant calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, attendant calls continue ringing at the originally called voice terminal.

Call Forwarding—Don't Answer

When Call Forwarding—Don't Answer is active at the called voice terminal, the forwarding operation occurs before Attendant Call Waiting is allowed. There are two possible operations. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is idle, attendant calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, attendant calls continue ringing at the originally called voice terminal.

Call Forwarding—Follow Me

When Call Forwarding—Follow Me is active at the called voice terminal, the forwarding operation occurs before Attendant Call Waiting is allowed. There are two possible operations. Attendant calls forward to and ring at the forwaded-to voice terminal (if this voice terminal is idle). Otherwise, attendant calls forward to, and then wait on, the forwaded-to voice terminal (if this voice terminal is busy).

Call Park

When the Call Park feature is active on the called extension, Attendant Call Waiting is denied.

Centralized Attendant Service

The Attendant Call Waiting feature is compatible with the CAS (Centralized Attendant Service) feature.When a CAS attendant extends a call to a branch location with Attendant Call Waiting assigned, then the extended call is allowed to wait (with 2-burst waiting tone) on an active single-appearance voice terminal at the branch.

Code Calling Access—Universal

A call is not allowed to wait (via Attendant Call Waiting) on a line that has accessed code calling.

Conference—Attendant Five Party

Attendant Call Waiting is denied when the attendant is attempting to add a busy called party to an attendant established conference.

Conference—Attendant Six Party

Attendant Call Waiting is denied when an attendant is attempting to add a busy called party to a conference.

Data Protection

Attendant Call Waiting is denied when the Data Protection feature is active on a call.

DDC (Direct Department Calling)

Attendant calls to a DDC group are not queued. The switch scans the group in an attempt to find an idle terminal to complete the call. If no idle line is found in the group, the call waits on the controlling terminal if Attendant Call Waiting is provided. When an attendant places a call to a DDC individual terminal and that terminal is busy, the call waits on the busy individual terminal if Attendant Call Waiting is provided.

DCS (Distributed Communications System)

In a DCS environment, direct attendant-calls and attendant-extended calls to an EUCD split in another node are queued. However, for attendant-extended calls, the attendant- does not receive confirmation tone to indicate that the queue has been entered.

EUCD (Enhanced Uniform Call Distribution)

Attendant calls to a local EUCD split are not queued. The switch attempts to complete the call to an idle agent by scanning the split. If no idle agent is found in the split, the call waits on the split supervisor's terminal. When an attendant places a call to an EUCD individual terminal and that terminal is busy, the call waits on the busy individual terminal when Attendant Call Waiting is provided.

Hold

Attendant Call Waiting is denied when the Hold feature is active on the called extension.

Hunting

When a called terminal line is associated with a hunting group, hunting is performed before the call waiting is implemented. If an idle terminal is not found in the hunt path, the call waits on the originally called terminal.

Line Lockout

An attendant call is not allowed to wait on a voice terminal that has been locked out The switch returns busy tone to the calling party.

Loudspeaker Paging

A call is not allowed to wait on a line that has accessed Loudspeaker Paging.

Music-on-Hold Access

When Attendant Call Waiting and Music-on-Hold are provided, a call released by the attendant to a busy single-appearance voice terminal is connected to music until the called party answers or the attendant reconnects to the waiting call.

Override

Override is not allowed toward a line that is waiting, but is allowed toward the 2-party call that has a call waiting.

Queuing

The callback sequence associated with ringback Queuing is delayed until there are no waiting calls.

Recorded Telephone Dictation Access

The Attendant Call Waiting feature is denied toward an extension with the Recorded Telephone Dictation feature activated.

Tenant Services

A partitioned System 85 or DEFINITY Generic 2 allows attendant calls to wait on local voice terminals, and always provides 2-burst tone for the called voice terminal. When allowed, 2-burst tone is provided for attendant calls directed to a voice terminal using the extension number. When allowed, 2-burst tone is also provided for attendant calls directed to a voice terminal using DXS (Direct Extension Selection).

Transfer

Attendant Call Waiting is denied toward an extension with the Transfer feature activated.

Trunk Verification—Attendant and Voice Terminal

If the trunk is being held or answered by a terminal using the Call Waiting-answer hold code of Attendant Call Waiting, Trunk Verification is denied.

UCD (Uniform Call Distribution)

Attendant calls to a UCD group are not queued. The switch scans the group in an attempt to find an idle terminal to complete the call. If no idle line is found in the group, the call waits on the controlling terminal if Attendant Call Waiting is provided. When an attendant places a call to a UCD individual terminal and that terminal is busy, the call waits on the busy individual terminal if Attendant Call Waiting is provided.

Hardware Requirements

None.

Feature Administration

Assignment of the Attendant Call Waiting feature is on a per-system basis in the system class of service.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel), or the customer can administer Attendant Call Waiting using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE ATTENDANT CALL WAITING FEATURE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
275	1	Assigns Call Waiting to the system class of service.	Yes
350	2	Assigns feature dial access codes. The applicable encode is: 6 = Call Waiting-answer hold	Yes

Notes:

Attendant Control of Trunk Group Access

Description

This feature prevents voice terminal users from directly accessing selected trunk groups. If a voice terminal user dials a controlled trunk group, the call routes to an attendant. The attendant then decides, based on instructions from the switch administrator, whether to allow the call.

During high demand periods, efficient use of trunk groups can be increased by limiting access. For example, an attendant can control trunk-group usage during peak outgoing call demand periods and can allow access during other periods. This has the effect of allowing outgoing calls on a priority basis. Also, an attendant can control outgoing access on 2-way trunk groups during peak incoming call demand periods. This would have the effect of reserving 2-way trunk groups for incoming calls.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Activate Control of a Trunk Group by the Attendant:

1. Press an idle loop button. [PA lamp goes out]
2. Press **[START]** . [Dial tone]
3. Dial the activation access code. [Second dial tone]
4. Dial the trunk-group access code,

   or

   Press the appropriate trunk-group selection button. [The corresponding "CONT" lamp lights, and the switch returns confirmation tone.]
5. Press **[RELEASE]** .

To Deactivate Control of a Trunk Group:

1. Press an idle loop button. PA lamp goes out]
2. Press **[START]** . [Dial tone]

3. Dial the deactivation access code. [Second dial tone]
4. Dial the trunk-group access code,

   or

   Press the appropriate trunk-group selection button. [The corresponding "CONT" lamp goes out, and the switch returns confirmation tone.]
5. Press **[RELEASE]** .

Considerations

Maximum Application

This feature can be activated for any trunk group assigned to a trunk-group selection button with an associated control lamp. There are 12 trunk-group selection buttons with associated lamps. Therefore, access can be controlled to a maximum of 12 trunk groups.

CONF Button

The CONF button also uses a trunk-group selection button. Therefore, when the Conference—Attendant Six Party feature is provided, access can be controlled to a maximum of 11 trunk groups

Hard and Soft Processor Swaps

ACTGA activations are stored in a status portion of switch memory. Therefore, if an attendant activates control of a trunk group before a hard processor swap, the trunk group will not be controlled after the hard swap is finished. At this time, the attendant can reestablish control.

The ACTGA feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

AAR (Automatic Alternate Routing)

For System 85 and DEFINITY Generic 2.1 switches, the ACTGA feature takes precedence over AAR. A call directed by AAR to a controlled trunk group is routed to an attendant.

ARS (Automatic Route Selection)

For System 85 and DEFINITY Generic 2.1 switches, the ACTGA feature takes precedence over ARS. A call directed by ARS to a controlled trunk group is routed to an attendant.

Call Vectoring

If vector processing encounters a "route to" step where the call would route over a trunk group that is currently controlled by the attendant, the "route to" step is not executed. Instead, if the "route to" step is the final effective step in the vector, the switch treats the "route to" step as a "stop" step. If vector steps follow the "route to" step, vector processing continues with the next sequential step.

Data Communications Access

Calls to DCA (Data Communications Access) ports can be restricted by applying Attendant Control of Trunk Group Access to the DCA trunk groups. Any attempt to access a DCA port is directed to an attendant for screening. The attendant can then transfer the call to a DCA port. However, attendant-extended data calls are not provided Data Protection.

Data Protection

Trunks under control of the attendant group cannot be directly accessed. An attendant must access these trunks for voice terminal users Data Protection—Temporary is not available for attendant-extended calls. These trunks, if used for data transmission, should be assigned Data Protection-Permanent.

DCS (Distributed Communications System)

Attendant Control of Trunk Group Access can be used for DCS trunks. However, when this is done, whatever "transparence" would normally be available is lost.

Host Computer Access

Calls to Host Computer Access ports can be restricted by applying Attendant Control of Trunk Group Access to the Host Computer Access trunk groups. Any attempt to access a Host Computer Access port is redirected to an attendant. If the attendant is to screen these calls, a voice terminal equipped with transfer capabilities must be provided near the attendant console to perform the transfer. Data call transfers cannot be performed from the attendant console because of the DCP interface at the Host Computer Access port. Modem Pooling is not provided for attendant-extended calls.

ISDN—PRI (Primary Rate Interface)

Generally, the Attendant Control of Trunk Group Access feature will work normally for ISDN trunks. However, to place an ISDN call, one of the network routing features (AAR, ARS, or WCR) must be used. Therefore, attendants can not use the Attendant Direct Trunk Group Selection feature to complete ISDN calls.

Look-Ahead Interflow

If the vector processing for Look-Ahead Interflow encounters a "route to" step where the call would route over a trunk group that is currently controlled by the attendant, the "route to" step is not executed. Instead, if the "route to" step is the final effective step in the sending switch's vector, the switch treats the "route to" step as a "stop" step. If vector steps follow the "route to" step, vector processing continues with the next sequential step.

Main/Satellite

When an attendant activates control of a Main / Satellite trunk group, calls over the trunk group route to an attendant. This occurs even though the calling party does not necessarily realize it is a trunk call and not an intraswitch call.

Precedence Calling

The ACTGA feature functions normally for AUTOVON trunks. When in effect, Precedence Calling calls are routed to the attendant priority queue.

Queuing

An outgoing call is routed to the attendant instead of an outgoing call queue when the ACTGA feature is activated. If the ACTGA feature is activated when there are already calls in an outgoing call queue, those calls are serviced from the queue without consideration to the ACTGA feature.

Recorded Telephone Dictation Access

If Attendant Control of Trunk Group Access is active on a recorded telephone dictation trunk group when the switch is in the preselected call routing mode of operation, any attempt to access a recorded telephone dictation trunk in the restricted trunk group results in intercept tone.

Route Advance

The Route Advance feature overrides ACTGA unless attendant control is active on the first trunk group in the Route Advance sequence. ACTGA works for the first trunk group of a Route Advance sequence. However, once the switch has activated a Route Advance sequence, the Attendant Control of Trunk Group Access feature provides no control over trunk groups that appear later in the sequence.

Tenant Services

An attendant (in a partition other than Attendant Partition 0) can control access to trunk groups that are assigned to that attendant's partition. When control is allowed the switch returns confirmation tone to the attendant (as would occur in an unpartitioned switch). However, when an attendant attempts to control a trunk group that is not assigned to the attendant's partition, the switch denies the attempt and returns intercept treatment to the

When Attendant Control of Trunk Group Access is activated toward a *shared* trunk group the switch applies control to every extension partition sharing the trunk group. Controlled outgoing calls from and an extension partition *with an* attendant partition assigned route to the assigned attendant partition. Controlled outgoing calls from an extension partition without an attendant partition assigned route to Attendant Partition 0 (if assigned). If

Attendant Partition 0 is not assigned, the switch denies these outgoing calls by retuning intercept treatment.

An attendant in Attendant Partition 0 can control access to any trunk group (except the Attendant Conference trunk group) in the lowest two rows of Attendant DTGS buttons. These two rows of buttons have "CONT" lamps.

Unattended Console Service-Alternate Console Position

When the Alternate Console Position and the ACTGA features are provided and activated concurrently, calls to a controlled trunk route to the alternate console position for subsequent processing.

Unattended Console Service-Preselected Call Routing and Call Answer From Any Voice Terminal

When the Preselected Call Routing or Call Answer From Any Voice Terminal features and the ACTGA feature are activated concurrently, calls to a controlled trunk receive intercept tone.

WCR (World Class Routing)

For DEFINITY Generic 2.2 switches, the ACTGA feature takes precedence over WCR That is, a call directed by WCR to a controlled trunk group is routed to an attendant.

Hardware Requirements

None.

Feature Administration

Assignment of the ACTGA feature is on a per-system basis.

On System 85 switches, this feature is only administered using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES ATTENDANT CONTROL OF TRUNK GROUP ACCESS		
PROCEDURE	**WORD**	**PURPOSE**
204	1	Designates the desired alphanumeric display for ACTGA to the attendant console(s). The applicable encode is as follows: R2 V1 to R2 V3: 291 Attendant Control of Trunk Group Access R2 V4 and later 2291 Attendant Control of Trunk Group Access.
350	1	Assigns the first digit of the ACTGA dial access codes (if required).
350	2	Assigns the dial access codes for the ACTGA feature. The applicable encodes are as follows: 20 Activate control of trunk group access 21 Cancel control of trunk group access.

Attendant Direct Extension Selection With Busy Lamp Field

Description

This feature allows an attendant to select an extension from as many as 1800 extension numbers by pressing two buttons instead of dialing an extension number. The BLF indicates the busy/idle status of the extension number. The BLF minimizes the time between an attendant's answer of an incoming call and the attendant's report that a voice terminal is busy. An attendant need only look at the field to determine whether the terminal is busy or idle.

Extended DXS is available for switches with more than 1800 extension numbers. For Extended DXS, the GROUP SELECT and GROUP DISPLAY buttons are assigned to the attendant console(s) as shown in Figure 8-1.

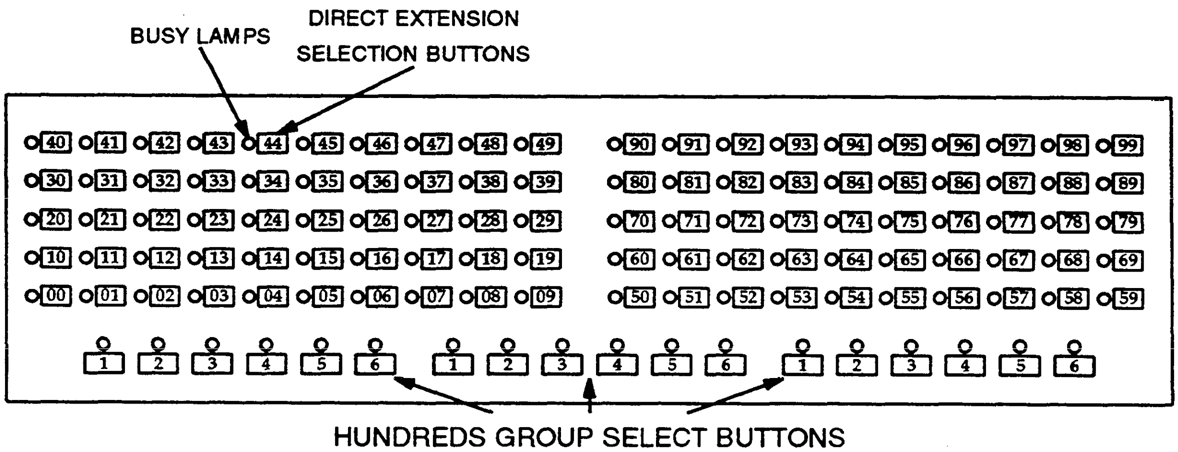


Figure 8-1. Attendant Console With DXS/BLF Option

Feature History and Development

This feature was first available on System 85 in Release 1. Because line capacities were increased Release 2 switches, the maximum number of hundreds groups that Extended DXS can access has also been increased. Extended DXS on Release 1 switches allows for as many as 32 hundreds groups, while 100 hundreds groups are allowed by Release 2 switches and DEFINITY Generic 2.

User Operations

The following are the user operating procedures for this feature.

To Call an Extension (such as, 7382) Using DXS:

1. Press an idle loop button. [PA lamp goes out.]
2. Press the appropriate hundreds group selection button (for example, 73).
3. Press the DXS button corresponding to the last two digits of the extension number (for example, 82).

To Call an Extension (such as, 6458) Using Extended DXS:

1. Press an idle loop button. [The GROUP SELECT lamp automatically lights.]
2. Press the **[START]** button. [Dial tone]
3. Press the DXS button corresponding to the first two digits of the extension number (for example, 64). [The alphanumeric display shows "64**".]
4. Press the DXS button corresponding to the last two digits of the extension number (for example, 58). [Adjacent BLF lamp lights.]

To Display the Currently Active Hundreds Group for Extended DXS:

Press **[GROUP DISPLAY]** .

Considerations

5-Digit Dialing

The Attendant DXS feature cannot be used on Release 2, Version 3 and later switches that use 5-digit dialing plans. Direct Extension Selection is inoperative on these switches.

DXS Versus Extended DXS

To facilitate operations for the attendant(s), either DXS or Extended DXS can be used, but not both. When Extended DXS is enabled, the hundreds group selection buttons for DXSS are disabled.

As many as 100 hundreds groups are accessible to Extended DXS. Therefore, using Extended DXS, attendants have convenient access to as many as 10,000 extension numbers.

Hard and Soft Processor Swaps

Attendant DXS button assignments are stored in a translation portion of switch memory. Therefore, these assignments will endure a hard processor swap.

The DXS buttons and lamps operate normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

ACD (Automatic Call Distribution)

In a switch with 3- or 4-digit extensions, an attendant can use the appropriate DXS buttons to place or extend calls to the associated extension number of an ACD split. However, since a split's queue is never really "busy," the BLF lamps adjacent to these DXS buttons are never lit.

Call Vectoring

In a switch with 3- or 4-digit extensions, an attendant can use the DXS buttons to place or extend calls to a VDN. However, since a VDN's associated vector is never really "busy," the BLF lamps for these DXS buttons are never lit.

Centralized Attendant Service

When a console at a Centralized Attendant Service main location is handling a call from a branch location, the BLF gives no indication of busy/idle status of branch location voice terminals. An attendant's DXS buttons cannot be used to call voice terminals at the branch.

The Attendant Direct Extension Selection With Busy Lamp Field feature is not available for backup voice terminals at branch locations.

Data Communications Access

When a terminal number is assigned to a Data Communications Access port, a Data Communications Access trunk group can be accessed using the appropriate DXS button. However, the associated busy lamp does not indicate the busy or idle status of the Data Communications Access port.

DDC (Direct Department Calling)

An attendant can use the appropriate DXS buttons to place or extend calls to the listed directory number of a DDC group. However, since a group's queue is never really "busy," the BLF lamps adjacent to these DXS buttons are never lit.

DCS (Distributed Communications System)

The DXS portion of this feature is compatible with the DCS feature as long as the dialing plan uses 4-digit extension numbers. Attendant DXS or Extended DXS cannot be used in a DCS where 5-digit extension numbers are used. Also, DXS does not work properly in a DCS where one or more of the nodes is an Enhanced DIMENSION® System switch.

The BLF portion, however, does not function properly in a DCS. The BLF field on attendant consoles does not operate transparently in a DCS environment. The BLF does not display the busy/idle status of extensions residing on different DCS nodes. The BLF only displays the busy/idle status of extensions residing on the same node as the attendant console.

EUCD (Enhanced Uniform Call Distribution)

An attendant can use the appropriate DXS buttons to place or extend calls to the associated extension number of an EUCD split. However, since a split's queue is never really "busy," the BLF lamps adjacent to these DXS buttons are never lit.

Main/Satellite/Tributary

The Attendant DXS portion of this feature can select a voice terminal on a satellite switch providing the dialing plan is limited to 4-digits. However, the BLF does not indicate busy/idle status for these voice terminals.

Off-Premises Terminals

When trunk port off-premises terminals have Multidigit Steering assigned, Attendant DXS With BLF can access an off-premises terminal via the appropriate DXS button. However, the associated busy lamp does not indicate its busy or idle state.

Tenant Services

Assignment of DXS or Extended DXS is on a system-wide basis. Every attendant in a partitioned System 85 or DEFINITY Generic 2 has the complete set of DXS buttons and the associated BLF lamps. The BLF lamps are updated on every attendant console, but attendant calling is limited by the Tenant Services feature.

An attendant (in a partition other than Attendant Partition 0) is allowed to use DXS or Extended DXS to call a voice terminal in any partition that the attendant can call using the touch-tone dialing pad. When an attendant attempts to call a voice terminal in a partition that is not allowed by partitioning, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to use DXS or Extended DXS to call any voice terminal in the switch.

UCD (Uniform Call Distribution)

An attendant can use the appropriate DXS buttons to place or extend calls to the listed directory number of a UDC group. However, since a group's queue is never really "busy," the BLF lamps adjacent to these DXS buttons are never lit.

Hardware Requirements

The following additional or special hardware is required for the DXS/BLF feature.

- Every attendant console must be the same type and provided with the same features, lamps, and button arrangements.

  NOTE: Two types of consoles are available for System 85 and DEFINITY Generic 2: type 30 and type 34. This feature (that is, DXS or Extended DXS) is only available using type 34 consoles which are equipped with DXS buttons and the busy lamp field.

Feature Administration

Assignment of the Attendant DXS With BLF feature is on a per-system basis.

On System 85 switches, this feature is administered using the MAAP.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
ATTENDANT DXS WITH BLF FEATURE

PROCEDURE	WORD	PURPOSE
200	1	Enables (or blocks) Extended DXS.
201	1	Assigns DXS group selection buttons to the attendant console(s).
201	2	Assigns hundreds groups accessible to Extended DXS.
203	1	Administers the GROUP SELECT and GROUP DISPLAY buttons for Extended DXS. The applicable encodes are as follows: 17 Group select 18 Group display.

Notes:

Attendant Direct Trunk Group Selection

Description

An attendant can access an idle outgoing trunk by pressing the button assigned to the desired trunk group. The single-button operation reduces the time and effort needed in handling outgoing calls. Each console has 24 DTGS (Direct Trunk Group Selection) buttons (see Figure 9-l).

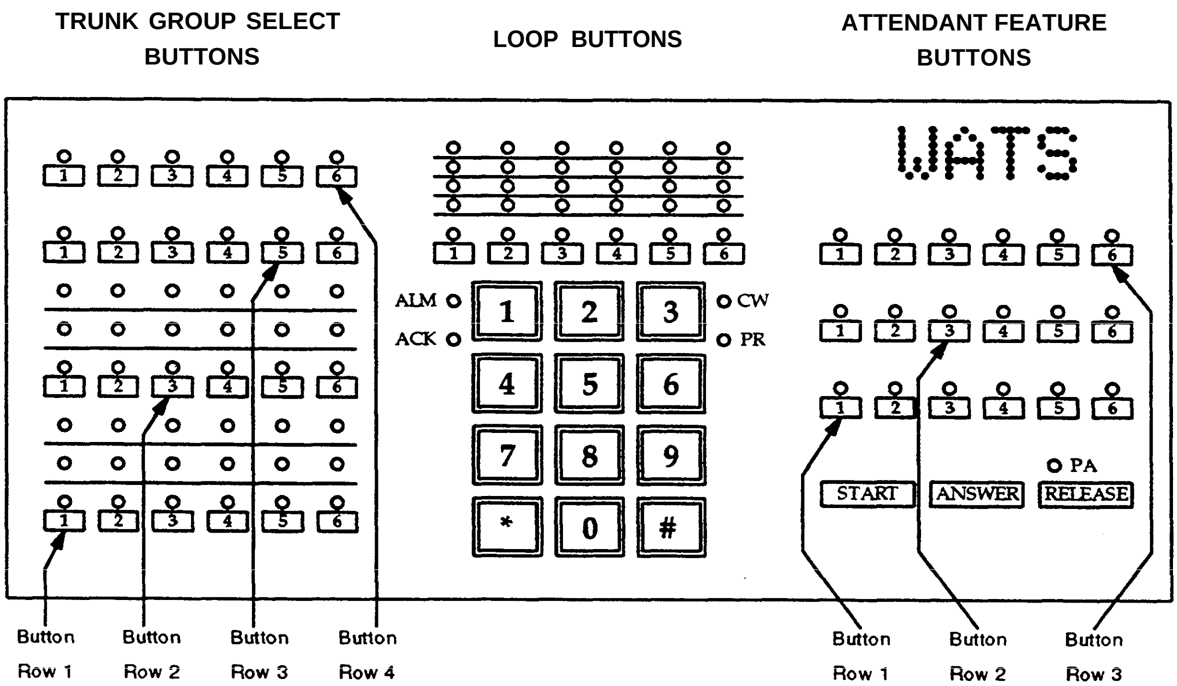


Figure 9-1. Attendant Console With DTGS Buttons

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following is the user operating procedure for this feature.

To Access an Idle Outgoing Trunk:

Press the desired trunk-group button.

Considerations

Dial Access Restriction

Dial Access Restriction (assigned to a trunk group in Procedure 100, Word 1) prevents attendants, voice terminal users, and data terminal users from directly accessing a trunk group by dialing the trunk-group access code. Dial access restriction also prevents an attendant from accessing a restricted trunk group with a DTGS (Direct Trunk Group Selection) button.

Hard and Soft Processor Swaps

Attendant DTGS button assignments are stored in a translation portion of switch memory. Therefore, these assignments will endure a hard processor swap.

The DTGS buttons and lamps operate normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

ISDN—PRI (Integrated Service Digital Network/Primary Rate Interface)

Because of messaging and information requirments for ISDN calls, all such calls should be placed using the AAR (Automatic Alternate Routing) or ARS (Automatic Route Selection) feature. Attendant Direct Trunk Group Selection should not be used to place an ISDN call.

Tenant Services

Assignment of Attendant Direct Trunk Group Selection is on a system-wide basis. Every attendant console in a partitioned System 85 or DEFINITY Generic 2 is provided with the complete set of 24 DTGS buttons and the associated lamps. However, the Tenant Services feature limits the use of these buttons. Also, the switch only updates the lamps associated with a specific button for consoles that can access that button's trunk group.

An attendant (in a partition other than Attendant Partition 0) is allowed to use the DTGS feature to access any trunk group that the attendant can access by dialing the trunk-group access code. When an attendant tries to select a trunk group that is not allowed by partitioning, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to use every DTGS button on the attendant console. Also, the switch updates the lamps associated with every DTGS button on these consoles.

Hardware Requirements

The only hardware required for this feature is an attendant console. Any standard attendant console available with the System 85 or DEFINITY Generic 2 can be used.

Feature Administration

Assignment of the Attendant Direct Trunk Group Selection feature is on a per-trunk group basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This features can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES ATTENDANT DIRECT TRUNK GROUP SELECTION		
PROCEDURE	**WORD**	**PURPOSE**
200	1	Enables the Attendant Direct Trunk Group Selection feature.
202	1	Assigns Direct Trunk Group Selection buttons to the attendant console(s).

Notes:

Attendant Display

Description

The Attendant Display feature is provided through an alphanumeric display on the attendant console (see Figure 10-1). This display provides information needed for call identification and rapid call completion. The attendant can display the extension or trunk number, nature, status, and class of service of the calling line or trunk. Information available includes the following.

- Calling Number
- Class of Service
- Incoming Call Identification
- Trunk Group Identification
- Specific Trunk Identification.

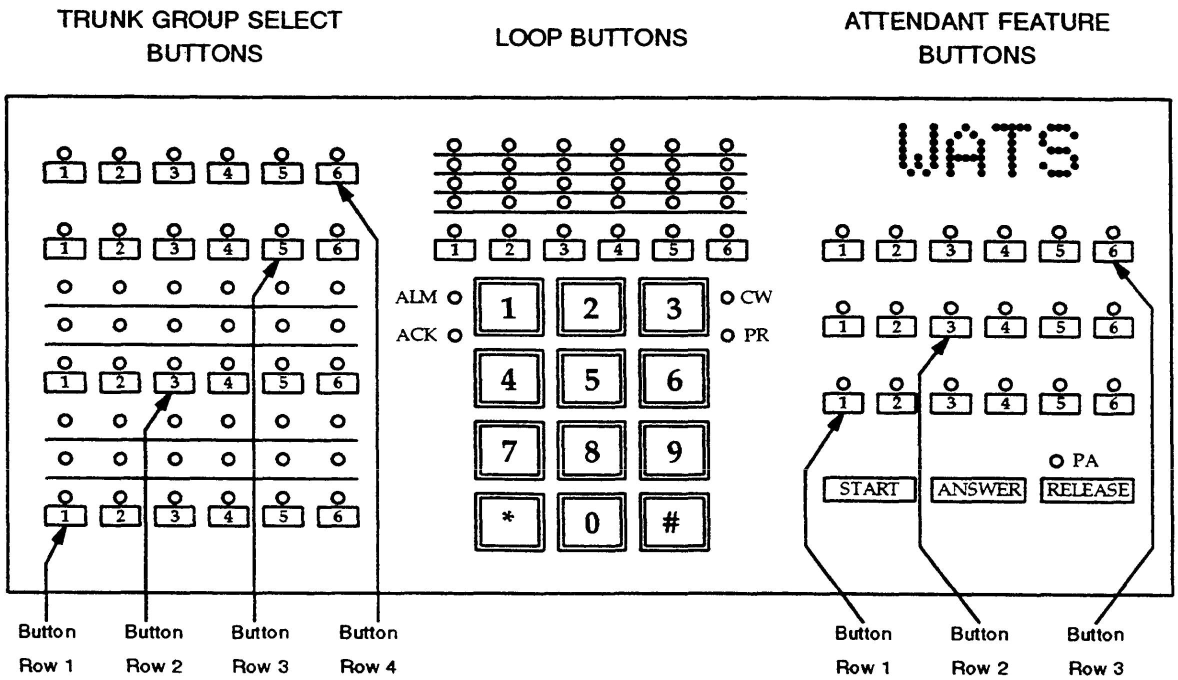


Figure 10-1. Attendant Console With Alphanumeric Display

Calling Number Display

The Calling Number display provides an immediate display of the internal extension number of a calling party.

With a 5-digit dialing plan, only the first (leading) four digits of the 5-digit extension number are shown on 4-character alphanumeric displays (for example, if the calling number is "52180," the attendant display shows "5218"). The eight-character alphanumeric displays show all five digits.

NOTE: Eight-character displays are strongly recommended for the attendant consoles when a switch (or DCS network) uses a 5-digit plan.

Class of Service Display

The Class of Service display specifies calling privileges (features and Facilities Restriction Level) allowed to an internal terminal. The switch can have 63 classes of service assigned, numbered 1 to 63. If specific information on each class of service is needed, a chart could provide the attendant with the pertinent information about each class of service. When general information is sufficient, displays something like the following can be used:

- **NON** — The calling terminal is unrestricted.
- **TOLL** — The calling terminal is toll restricted.
- **REST** — The calling terminal is outward restricted.
- **FULL** — The calling terminal is fully (inward and outward) restricted.

ICI (Incoming Call Identification)

The ICI display identifies the type of call being handled. The display can provide 30 different messages of up to 4 characters. The first three messages are standard in all switches. They are:

- **INC** — Incoming call
- **ATND** — Calls to the Attendant
- **RCL** — Attendant recall.

The customer selects the other messages when ordering the switch. The following are some examples of messages that could be used:

- **NY, PHIL, WASH** — Used to identify the city of origin for an FX (Foreign Exchange) trunk.
- **CONF** — Indicates that a conference is recalling the attendant.
- **PC** — Identifies an interposition call from another attendant.
- **CTGA** — Indicates an intercepted call routed to the attendant because of the Attendant Control of Trunk Group Access feature.
- **ACAS** — Indicates that the ACA (Automatic Circuit Assurance) feature has detected a possible trunk malfunction. This message also identifies the type of referral ("L": long, or "S": short) that occurred.
- **CFWD** — Indicates a call forwarded to the attendant, etc.

Trunk Identification Display

The Trunk Identification display identifies the specific trunk being used on an incoming or outgoing call by displaying the Trunk Group Dial Access code and the individual trunk number. This display is particularly useful when the attendant wishes to place test calls for trunk verification.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Access the Class of Service Display While Connected to an Internal Party:

Press the **[CLASS]** button. [ICI display changes to class of service display.]

To Access the Specific Trunk Display While Connected to a Trunk Appearance:

Press the **[TK-ID]** button. [Trunk Group display changes to identify the specific trunk number.]

Considerations

Legal Consideration

Laws governing the use of calling number displays differ in different locations and are subject to change. It is the responsibility of the customer's switch administrator to understand and comply with the applicable regulations.

Assigned Control Buttons

The class of service (CLASS) button and the trunk identification (TRK ID) button are assigned to control buttons on the console.

Trunk-to-Trunk Call Identification

On a trunk-to-trunk call, incoming trunk identification can only be made before completing the trunk-to-trunk connection. Subsequent trunk codes displayed are for the outgoing trunk only.

Hard and Soft Processor Swaps

The contents of the Attendant Display assignments are stored in a translation portion of switch memory. Therefore, these display assignments will endure a hard processor swap.

An attendant display does not operate during a hard processor swap. However, the display operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

ACD (Automatic Call Distribution)

When standard ACD intraflow diverts a coverage call to the attendant queue by activating Call Forwarding (with the Attendant DAC as the destination) at the split supervisor's voice terminal], the attendant who answers the redirected call receives the usual display information on the alphanumeric display (i.e., the same display information that would have been provided for a direct call to the attendant). The called principal's identification is not provided.

Call Vectoring

When Call Vectoring diverts a coverage call to the attendant queue (using a "route to" step in the covering vector), the attendant who answers the redirected call receives the usual display information on the alphanumeric display (i.e., the same display information that would have been provided for a direct call to the attendant). The called principal's identification is not provided.

Conference—Attendant Five Party

The Attendant Display feature is denied when the attendant is connected to a conference.

Conference—Attendant Six Party

The Attendant Display feature is denied to an attendant when the attendant is connected to a conference circuit.

Look-Ahead Interflow

At a receiving switch, using a "route to" step with the Attendant Dial Access code (Encode 8) as the destination, an incoming Look-Ahead Interflow call can reach the attendant queue. When this is done, the receiving switch will deliver the normal ICI (Incoming Call Identification) display associated with the incoming trunk group to the answering attendant.

Hardware Requirements

None.

Feature Administration

Assignment of the Attendant Display feature is on a per-system basis.

On System 85 switches, this feature can be administered with the MAAP.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES ATTENDANT DISPLAY FEATURE		
PROCEDURE	**WORD**	**PURPOSE**
200	1	Specifies the type of display given (Field 2) for calls from local extensions.
203	1	Assigns the display buttons to the attendant console(s). The applicable encodes are: 1 CLASS Button 28 TRK-ID Button 42 STA-ID Button.
204	1	Designates the desired alphanumeric displays.

Notes:

Attendant Interposition Calling and Transfer

Description

An attendant at one position, in a multiposition switch, can call another attendant by dialing an access code and the position's assigned number. The called attendant receives a special alphanumeric display (such as, IPC) to designate the interposition call. This process is used for consultation or when questions about call processing arise. The attendant can also transfer a call to another position on the same switch for special handling.

When the called attendant is busy on another call, the interposition call is held waiting in a priority queue, and the PR (priority call waiting) lamp on the called attendant console lights. When the called console becomes available, the interposition call switches to the first idle loop and takes precedence over the other calls in queue.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following arem the user operating procedures for this feature.

To Call Another Attendant

Without a source party:

1. Press an idle switched loop button. [If lit, the PA lamp goes out. ATND lamp lights.]
2. Press the **[START]** button. {Dial tone is heard, and SPLIT lamp lights.]
3. Dial the interposition access code. [Dial tone]
4. Dial the attendant position's assigned number. [At calling console, RING lamp lights, and ringback tone is heard. At the called console, ATND lamp flashes if unbusy, PR lamp flashes once quickly, alerting tone sounds, and the ICI display lights.]

With a source party:

1. Press the **[START]** button. [Dial tone is heard, and SPLIT lamp lights.]
2. Dial the interposition access code. [Second dial tone]
3. Dial the attendant position's assigned number. [At calling console, RING lamp lights, and ringback tone is heard. At called console, ATND lamp flashes if unbusy, PR lamp flashes if busy, alerting tone sounds, and the ICI display lights.]

To Answer an interposition Call at the Called Console:

Press the **[ANSWER]** button. [At calling console, RING lamp goes out, ANS lamp lights, and ringback tone is removed. At called console, alerting tone is removed, and the ATND lamp lights. If there's a source party at the called console, the SPLIT lamp lights.]

To Transfer a Source Party to Another Attendant

Before a 2-way connection is made with the other attendant:

1. Call the other attendant from the console with the source party.
2. Press the **[RELEASE]** button. [At the calling console, the attendant is released from the source party and the RING, SPLIT, and ATND lamps go out If no calls are waiting the PA lamp lights. At called console, the ATND lamp flashes, and the ICI display changes from indicating an interposition call to indicating the source-party number.]

After a 2-way connection is made with the other attendant:

Press the **[RELEASE]** button. [At the calling console, the attendant is released from the source party, and the SPLIT lamp goes out. If no calls are waiting, the PA lamp lights.]

[At called console, the attendant is connected to the source party, and the SPLIT lamp goes out]

To Cancel a Call to Another Attendant:

Press the **[CANC]** button.

Considerations

Intercept Tone

Intercept tone is heard by the calling attendant when the called attendant position is unattended (handset removed).

Extending and Transferring Calls

Calls from a voice terminal user to one attendant can be transferred to another attendant, but an attendant cannot extend or transfer an attendant to another position.

2-Party Connections

An attendant cannot transfer a 2-party connection to another attendant position.

Queuing Calls

Interposition calling takes precedence over all attendant-queued calls. Interposition calling is automatically placed in the priority queue and processed on a first-in, first-out basis.

Hard and Soft Processor Swaps

Stable interposition calls will endure a hard processor swap. However, an interposition call cannot be placed during a hard processor swap.

Attendant Interposition Calling and Transfer operatesnormally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Centralized Attendant Service

Calls originating to a centralized attendant via a release link trunk cannot be transferred to another attendant using the Attendant Interposition Calling and Transfer feature.

Malicious Call Trace

If an attendant selectively calls the activating or the controlling attendant during an active Malicious Call Trace, the switch returns ringback to the calling attendant. However, the called attendant will only receive the alerting tone if the attendant has unbusied the

For the controlling attendant, priority lamp activity is provided for interposition calls during a trace. However, the controlling attendant does not receive an ICI display for these calls. Instead, the alphanumeric display is reserved for displaying trace information.

For the activating attendant to receive an interposition call, the malicious caller must be put on hold, and the console must be unbusied. Then, an interposition call will cause the loop lamp to flash and the PR lamp to go out.

Tenant Services

An attendant (in a partition other than Attendant Partition 0) is only allowed to place interposition calls to attendants in the same partition or to attendants in Attendant Partition 0. If an attendant tries to place an interposition call to an attendant in any other partition, the switch denies the call by retuning intercept treatment.

An attendant in Attendant Partition 0 is allowed to place an interposition call to an attendant in any partition.

Unattended Console Service—Preselected Call Routing

If an attendant position calls another attendant with Preselected Call Routing active, the call is placed in the called attendant's priority queue. When Preselected Call Routing is deactivated, the call is removed from the priority queue and alerting is provided. The call does not route to the preselected voice terminal.

Hardware Requirements

None.

Feature Administration

Assignment of the Attendant Interposition Calling and Transfer feature is on a per-system basis.

On System 85 switches, this feature is administered using the MAAP.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES ATTENDANT INTERPOSITION CALLING AND TRANSFER		
PROCEDURE	**WORD**	**PURPOSE**
200	1	Enables the Interposition Calling and Transfer feature.
204	1	Designates the desired Attendant Display for interposition calling. The applicable encode is as follows: R2 V1 to R2 V3: 297 Interposition calling R2 V4 and later: 2297 Interposition calling.
210	1	Assigns the equipment location of an attendant console position.
350	1	Assigns the first digit of the dial access codes for the Interposition Calling and Transfer feature (if required).
350	2	Assigns the feature dial access codes. The applicable encode is: 29 Interposition call.

Attendant Recall

Description

This feature allows a voice terminal user on a 2-party call or on a conference call to call the attendant for assistance. The ICI (Incoming Call Identification) displays "RCL," or the extension number of one party in the 2-party connection to indicate an attendant recall.

Feature History and Development

This feature was first available on System 85 in Release 1. The enhancements to this feature include:

- The ability to recall an attendant using an LDN (listed directory number) was first provided in Release 2, Version 3.
- An administrable recall button is provided for R2 V4 and was also retrofitted to the R2 V2 and R2 V3 software packages.

User Operations

The following are the user operating procedures for this feature.

To Recall the Attendant From a Connection Held on the Attendant Console

From a single-appearance voice terminal without RECALL button:

1. Momentarily press the switchhook.
2. Held loop alerts on the attendant console. [ANS lamp flashes, and ICI shows the extension number of the terminal NOT performing the recall.]

From a single-appearance voice terminal with RECALL button or a multiappearance voice terminal:

1. Press **[RECALL]** .
2. Held loop rings on the attendant console. [ANS lamp flashes, and ICI display shows extension number of the terminal NOT performing the recall.]

From a single-appearance voice terminal without a RECALL button:

1. Momentarily press the switchhook. [Recall dial tone] (Other party is put on soft hold.)
2. Dial the attendant (not a selected attendant) access code,

   or

   Dial an LDN. [Ringback tone]
3. An idle attendant loop rings. [Loop lamp flashes, and ICI display shows RCL.]

From a single-appearance voice terminal with a RECALL button:

1. Press **[RECALL]** . Recall dial tone] (Other party is put on soft hold.)
2. Dial the attendant (not a selected attendant) access code,

   or

   Dial an LDN. [Ringback tone]
3. An idle attendant loop rings. [Loop lamp flashes, and ICI display shows RCL.]

From a multiappearance voice terminal:

1. Press **[CONFERENCE]** or **[TRANSFER]** . [Dial tone] (Other party is put on hard hold.)
2. Dial the attendant access code, a selected attendant access code,

   or

   Dial an LDN. [Ringback tone]
3. An idle attendant loop rings. Loop lamp flashes, and ICI display shows extension number of the terminal performing the recall.]

Considerations

Administrable Recall Buttons

Some multiappearance voice terminals do not have a fixed RECALL button. If RECALL buttons are needed for these voice terminals, RECALL buttons can be assigned to the terminals using Procedure 054, Word 1.

Hard and Soft Processor Swaps

Stable attendant recalls will endure a hard processor swap. However, a voice terminal user cannot recall the attendant during a hard processor swap.

The Attendant Recall feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Conference—Three Party and Transfer

For the Attendant Recall feature to function properly, the Conference—Three Party and Transfer features must be enabled in Procedure 010, Word 1.

Tenant Services

For calls held on an attendant console, the Attendant Recall feature is naturally partitioned on System 85 and DEFINITY Generic 2. In a properly partitioned switch, an attendant cannot receive a call that breaks the rules of partitioning. The attendant must extend the call according to partitioning rules. So, for a call held on a console, both parties in the held call can access the attendant holding the call. And, the Attendant Recall feature always alerts the attendant who is holding the call.

For calls ***not held*** on an attendant console, the Attendant Recall feature is also partitioned. For these recalls, one of the talking parties puts the other on hold and then dials the attendant group. In a properly partitioned switch, both of these parties can access the same attendant partition. So, when either party dials the attendant queue (using either the attendant access code or an LDN) the same partitioning checks used by the Dial Access to Attendant feature are made.

Hardware Requirements

None.

Feature Administration

The Attendant Recall feature, as such, does not require administration. However, if administrable recall buttons are needed, they are administered on System 85 switches using the MAAP (Maintenance and Administration Panel) or the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, administrable Recall buttons can be administered using the DEFINITY Manager II.

Administerable Recall Buttons can also be administered using the Manager IV.

ADMINISTRATION PROCEDURES ATTENDANT RECALL FEATURE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
		Assigns the RECALL button to a multiappearance voice terminal without a fixed RECALL button. The applicable encode is: 27 Recall.	Yes

Attendant Release Loop Operation

Description

An attendant can hold an incoming trunk call off the console if completion of the call has to be delayed (such as when the extended call waits on a busy extension or when an idle extension does not answer). This frees the switch loop to handle other incoming calls. Therefore, by relieving congestion at the attendant console, this feature is useful to customers with a high volume of incoming traffic. The result is that more calls can be handled by fewer attendants with fewer attendant consoles. Without this feature, an attendant console would only be able to handle six calls at a time.

Timed Reminder

An Attendant Release Loop timed reminder starts timing on release of the call. If the called extension does not answer before the timed reminder interval expires, the call is returned to the attendant incoming call queue for further processing.

Timed Reminder Interval

If not otherwise administered, the timed reminder interval is set by default at 30 seconds. Otherwise, the timed reminder interval can be set by the attendant at from 02 to 98 seconds in 2-second increments.

Call Identification

A timed reminder tone is heard at the attendant console. When the attendant answers the call, the trunk identification number is shown on the alphanumeric display. The called extension can be identified by the attendant pressing the STA ID button.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Release the Attendant Loop and Hold a Call Off Console

After completing a call to a local station:

Press **[RELEASE]** .

During ringing:

Press a different loop button.

To Change the Timed Reminder Interval:

1. Press an idle loop button. [PA lamp goes out]
2. Press **[START]** . [Dial tone]
3. Dial the ARL time change access code. [Second dial tone]
4. Dial a 2-digit even number between 02 and 98. [Confirmation tone]
5. Press **[RELEASE]** . [PA lamp lights.]

Considerations

Hard and Soft Processor Swaps

The value of the ARL Timed Reminder Interval is stored in a translation portion of switch memory. Therefore, if an attendant sets the interval and then a hard processor swap occurs, this interval will endure the hard swap.

If an attendant-extended call is ringing a voice terminal when a hard swap occurs, this unstable call is dropped and does not return to the attendant console.

The Attendant Release Loop Operation feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

ACD (Automatic Call Distribution)

The Attendant Release Loop Operation feature does not apply to calls that an attendant extends to an ACD split. Once an attendant-extended call enters the ACD queue, this call is not timed and no reminder will be given to the attendant.

Call Coverage

When an attendant extends a call to a voice terminal with coverage assigned (with coverage criteria that apply to the call) and releases the call, the call will not return to the attendant queue. Instead, the call will redirect to coverage according to the assigned coverage path.

For a call from an internal voice terminal extended by an attendant to another internal voice terminal, the call can redirect to coverage as an attendant call (like an external call). This occurs when the attendant does not release the call before the call type is checked for redirection to coverage.

Call Vectoring

The Attendant Release Loop Operation feature does not apply to calls that an attendant extends to a VDN (Vector Directory Number). Once an attendant-extended call enters vector processing, this call is not timed and no reminder will be given to the attendant.

EUCD (Enhanced Uniform Call Distribution)

The Attendant Release Loop Operation feature does not apply to calls that an attendant extends to an EUCD split. Once an attendant-extended call enters the EUCD queue, this call is not timed and no reminder will be given to the attendant.

Tenant Services

The attendant-calling operations of this feature are naturally partitioned on System 85 and DEFINITY Generic 2. In a properly partitioned switch, an attendant cannot receive a call that breaks the rules of partitioning. The attendant must extend the call according to partitioning rules. And, the Attendant Release Loop Operation feature always returns the call to the same attendant partition of the console that extended the call.

The ARL Time Change operation is not partitioned. The ARL timed reminder interval can be changed by an attendant in any attendant partition. When this is done, the new ARL timed reminder interval is applied to every attendant partition in the switch.

Timed Reminder

The Attendant Release Loop Operation feature is similar to the Timed Reminder feature, and both of these features can reside on the same switch. The Timed Reminder feature is always provided, while the Attendant Release Loop Operation feature is optional.

When both features reside on the same switch, these two features operate simultaneously and control two separate sets of calls. Incoming trunk calls are controlled by the Attendant Release Loop Operation feature, and the timed reminder interval for these calls is assigned in Procedure 275, Word 4. Other affected calls are controlled by the Timed Reminder feature, and the timed reminder interval for these calls is fixed at 30 seconds.

When only the Timed Reminder feature resides on the switch, ***all*** affected calls are controlled by the Timed Reminder feature, and the timed reminder interval is fixed at 30 seconds.

Hardware Requirements

None.

Feature Administration

Assignment of the Attendant Release Loop feature is on a per-system basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel) or SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
ATTENDANT RELEASE LOOP OPERATION

PROCEDURE	WORD	PURPOSE	SMT
203	1	Administers the Extension Number Display Time-Out button for the attendant console(s). The applicable encode is: 42 STA-ID Button.	No
275	4	Assigns the Attendant Release Loop Operation feature to the system class of service and set the timed reminder interval.	Yes
350	1	Assigns the first digit of the dial access code to change the timed reminder interval (if required).	No
350	2	Assigns the dial access code to allow the attendant to change the timed reminder interval. The applicable encode is: 31 ARL time change.	No

AUDIX™ Voice Messaging System

Description

The AUDIX Voice Messaging System is a message-handling or "Voice Mail" system. It allows AUDIX users or "subscribers" to send and receive voice messages. AUDIX subscribers can also receive messages from callers through the Call Answer or Leave Word Calling features. The AUDIX system uses recorded prompts and announcements to guide callers through the messaging operations.

The AUDIX Reference manual, 585-300-201, details all the AUDIX features, their applications, system administration, and switch feature interactions. This chapter summarizes the main AUDIX features and provides references to other documents for additional information. AUDIX highlights include:

- Easy 1-way information transfer through a voice terminal (telephone)
- Communication with people who are busy or unavailable during regular hours, away on business, or based in different time zones
- Interaction where an immediate response is unnecessary
- Reduction of incomplete calls
- Notification of new messages through audible or visual message-waiting indication (stutter dial tone or message-waiting lamps) or the AUDIX Outcalling feature
- Integration with the AT&T Unified Messaging and Applications Processor message facilities for more complete and efficient message handling
- Ability to take faulty voice ports out of service on the AUDIX side and to busy-out the corresponding switch port
- Flexible configuration and potential for future growth. AUDIX one- or two-cabinet systems can be ordered depending on the customer needs. AUDIX-Large (AUDIX-L) systems can no longer be ordered.

Feature History and Development

Early releases (prior to R2 V2) of System 85 switch support AUDIX in the Stand-alone mode only (no data link).

Beginning with System 85 Release 2, Version 2, AUDIX is available in a fully integrated (Enhanced) form.

System 85 R2 V4 and Generic 2 switches support the following AUDIX Enhanced features:

- Transfer Into AUDIX
- Enhanced Transfer Out of AUDIX

In 1991, the term AUDIX became a trade mark, necessitating a feature name change from AUDIX to AUDIX™ Voice Messaging System.

Main AUDIX Features

The following main features available with all AUDIX software releases, and each release enhances their basic functions. AUDIX Enhanced (R1 V2), AUDIX Enhanced II (R1 V3), and AUDIX Enhanced III (R1 V4) software support the following features:

Voice Mailbox

The Voice Mailbox feature gives all AUDIX subscribers access to disk storage where they create, receive, and store messages. AUDIX subscribers are users who have personal Voice Mailboxes which allow them to create, edit, send, and receive spoken messages. Subscribers access their Voice Mailboxes by logging on to AUDIX with a touch-tone telephone and supplying the correct password. Users create and alter messages in their mailboxes by using a set of activities.

Highlights of the Voice Mailbox features include:

- Addressing Messages by Name or Number

  Both methods of addressing may be used in a single message. AUDIX plays back subscriber names, so subscribers can confirm they have entered the correct extension numbers for their intended recipients.

- Broadcasting Messages

  Mailing lists allow subscribers to send a single message to up to 250 other subscribers.

- Delivery Scheduling

  Messages may be sent up to one year after creation. This feature may also be used for subscribers to send messages to themselves as reminders.

Call Answer

The Call Answer feature is administrator-assigned and allows AUDIX to answer calls for subscribers who are busy or otherwise unavailable. The Call Answer feature places AUDIX in a subscriber's call-coverage path, allowing AUDIX to automatically answer calls when the subscriber does not pick up the call. AUDIX may answer calls from subscribers or nonsubscribers located inside or outside the building (callers do not need a touch-tone telephone). Highlights of the Call Answer feature include:

- Depending on the switch setup, dual or multiple call-coverage paths or call forwarding points may be set up. This could allow external callers, for example, to go to a live answering service (such as a Message Center Service on an Applications Processor) while internal callers are redirected to AUDIX.

- Subscribers can record personal greetings for their callers, which may include information on their schedules or options for transferring the call.
- Subscribers may direct their calls to AUDIX as needed using the Call Forwarding or Call Coverage features on the switch.
- Callers who are redirected to AUDIX may optionally transfer to another extension or to a live covering agent, if set up by the system administrator.

Information Service (Bulletin Board)

The Information Service feature, (also known as Bulletin Board), is optional and allows any internal or external caller to dial a special extension to hear a prerecorded message of general interest. The Information Service is similar to Call Answer except callers may not leave a message; touch-tone capability is not required. Several extensions may be used for Information Service if needed. Messages may be up to 20 minutes long, depending on locally administered system limits.

LWC (Leave Word Calling)

LWC (Leave Word Calling) is a switch feature that allows an internal calling party (one administered on the local switch) to leave a standard-format message requesting the called party to return the call. LWC messages may be created by the caller pressing a feature button or using a dial access code. AUDIX LWC messages are placed in the subscriber's incoming mailbox and accessed along with other messages.

AUDIX Enhanced (R1 V2) Features

AUDIX Enhanced (R1 V2) software runs only on the one-cabinet AUDIX formerly called AUDIX-Small (AUDIX-S), and on the older AUDIX-L models. Features include:

ADAP (AUDIX Data Acquisition Package)

The ADAP (AUDIX Data Acquisition Package) allows an AUDIX system administrator to better manage the system's traffic and storage by transferring AUDIX traffic and subscriber data to a PC (Personal Computer) or WGS (Work Group System). Application programs written in dBASE III PLUS software can convert this data into easily read reports. ADAP is fully covered in the *AUDIX Data Acquisition Package* guide (585-302-502). The *Stella Business Graphics Package I* (585-302-504) covers an optional graphics package for drawing charts and graphs.

Dial by Name

Dial by Name allows callers who may not know an extension number to select name addressing with the *A (Alternate Addressing) command, and then transfer to any AUDIX subscriber by dialing the name instead of a number.

Directory of Subscribers

The Directory of Subscribers feature keeps a directory of subscriber names and extension numbers in AUDIX. Callers may use the **N (Names and Numbers Directory) command at any time to find out the name or extension number of an AUDIX subscriber, or to verify whether the person they are trying to reach is an AUDIX subscriber.

Escape to an Attendant

The Escape to an Attendant feature allows an AUDIX subscriber with the Call Answer feature to have a personal attendant or an operator administered to potentially pick up a call. Callers who reach AUDIX for that subscriber through Call Answer may immediately redirect the call to reach the live attendant by pressing "0," or first leave a message and then press "0" to reach the live agent.

> **NOTE:** The transfer destination should be a staffed position. Once the call is transferred, it leaves the originally called subscriber's coverage path. If no one picks up the call, or if call coverage for the covering agent is not available, callers may hear ringing with no answer. On some switches without Enhanced Call Transfer, callers may experience a delay before the covering party can hear them.

Guest Password

The Guest Password feature allows people who are ***not*** AUDIX subscribers to access AUDIX by dialing a subscriber's extension and entering a system-wide guest password. These callers can leave messages for that subscriber but cannot listen to other messages in the mailbox The guest password may also be used to leave messages for subscribers who don't have call-coverage to AUDIX or to bypass an agent in a coverage path in order to record an AUDIX message for another subscriber.

Multiple Sessions

The Multiple Sessions feature allows two or more AUDIX subscribers to logon to AUDIX sequqntially without needing to place a new call. For example, if subscribers are calling long distance, the first caller can dial **R (Relogin) to allow the next subscriber to log on.

On-Line Help

On-Line Help: Provides three levels of on-screen information for system administrators or service technicians working on an AUDIX terminal. PATH line command, form, and field help are available.

Restart AUDIX

The Restart feature allows subscribers who have reached AUDIX through the Call Answer feature to access their own mailboxes by typing the *R (Restart) command. This is especially useful for long-distance calls or for AUDIX Stand-alone users who wish to access AUDIX when all the Voice Mail ports are busy.

Return Call

The Return Call to Sender feature allows an AUDIX subscriber to immediately place a call to the originator of an incoming message if that person is in the switch's dial plan.

Transfer Into AUDIX

The transfer Into AUDIX feature allows an attendant (or other party) to transfer a caller who has been sent to coverage (or otherwise redirected) back to AUDIX to record a message. The subscriber's personal greeting will play normally. The Transfer Into AUDIX feature is a switch-dependent feature.

Transfer Out of AUDIX

The Transfer Out of AUDIX feature allows any caller with touch-tone capability who has reached AUDIX (either through Call Answer or by direct dialing) to leave AUDIX and transfer to another extension in the switch's dial plan by using the *T (Transfer) command. Call Answer callers may leave a message first, then transfer to another extension. Callers may also optionally select name addressing instead of extension numbers to transfer a call.

Either basic (switchhook flash) or enhanced call transfers using ES (Enhanced Services) signalling are allowed. Enhanced call transfers provide better error recovery, accurate display information, and faster transfers than basic call transfers (3 to 5 seconds instead of 8 or 10).

> **NOTE:** The Transfer Into AUDIX and Enhanced Transfer Out of AUDIX features require System 85 R2 V4 or Generic 2 software.

AUDIX Enhanced II (R1 V3) Features

All AUDIX models support AUDIX Enhanced II (R1 V3) software. This software includes the AUDIX R1 V2 features and introduces the following additional features.

Automated Attendant

The Automated Attendant feature can route callers to the correct department or extension by offering them a voiced menu of options. Callers can press a touch-tone key to be routed automatically. Callers could also wait for a live attendant to answer or for AUDIX to record a message, depending on the options set at the site. Fewer live attendants may be needed because AUDIX can answer incoming calls and route them to the appropriate department.

Exit Command

AUDIX callers can use the Exit Command (**X) to have AUDIX disconnect ***without*** hanging up. This feature is especially useful for toll calls or remote outcalls.

Networking

The AUDIX Networking feature allows each subscriber in the AUDIX network to schedule automatic delivery of voice mail messages to subscribers on up to 100 other AUDIX systems. Up to 32,000 subscribers may be administered on one AUDIX system, although additional non-administered subscribers may be addressed as well. AUDIX digital messages have better quality than analog recordings and can be transmitted more quickly.

Outcalling

The Outcalling feature allows AUDIX to call subscribers when they receive new messages. This is especially useful for systems that do not have other message-waiting indicators. Subscribers can select the time period during which AUDIX may call them and the number where they can be reached.

Real-Time Clock

An accurate internal Real-Time Clock allows AUDIX to keep time without relying on the switch.

Stand-Alone AUDIX

The AUDIX Stand-alone feature allows AUDIX to work ***without*** a data link, so it can be connected to a wide variety of switches that could not be supported before (such as a System 85 R2 V1 PBX. Some changes to the user interface exist because the switch is not fully integrated (messages that require a data link can no longer be sent, including calling- and called-party IDs and LWC messages). In AUDIX R1 V4, AUDIX Stand-alone subscribers can automatically receive ***Stand-alone Message Notification*** to either turn on a message-waiting lamp or activate a stutter dial tone.

AUDIX Enhanced III (R1 V4) Features

AUDIX Enhanced III (R1 V4) software is available on all AUDIX models. AUDIX R1 V4 software includes all the features available in previous AUDIX software releases, plus the following additional features:

Executive Features

The following set of AUDIX Executive Features are available on AUDIX R1 V4.

- Private Messaging

  Private Messaging provides subscriber commands to prevent an AUDIX voice mail recipient from forwarding the message.

- Untouched Message

  Untouched Message allows a subscriber or secretary to review message headers in the incoming mailbox without changing their status.

- Variable password Length

  A security password length of up to 15 characters. The AUDIX system administrator can optionally increase required subscriber password lengths to provide extra system security.

File Redundancy

The AUDIX system administrator can set up ***File Redundancy*** so that information is copied to a backup filesystem automatically while the system runs. In the event of a disk drive failure, the duplicate filesystem is automatically activated so service can continue without interruption.

Text Service Interface

The AUDIX Text Service Interface feature sends AUDIX message headers to an electronic mail service that resides on another vendors host computer, such as the IBM PROFS (Professional Office System) electronic text mail service. Information such as the sender's name and extension, message delivery time, and message length can be displayed on the text service user's terminal or PC screen. Other third-party interfaces can be supported.

However, customers must write their own application programs for such services. The AUDIX Text Service Interface guide (585-304-503) describes the Text Service Interface feature.

User Operations

AUDIX commands are 1- to 3-key commands that are entered using a digit or a character string beginning with an asterisk (*). Commands are ended with the pound (#) sign. AUDIX user operations are covered in the following AUDIX documentation:

- ***AUDIX User's Guide*** (585-302-701) — R1 V1 through R1 V3
- ***A Quick Guide to AUDIX*** (585-302-702) — R1 V1 through R1 V3
- ***AUDIX Wallet Card*** (585-302-703) — R1 V1 through R1 V3
- ***AUDIX Enhanced III User's Guide*** (585-304-701) — R1 V4
- ***A Portable Guide to AUDIX Enhanced III(*** (585-304-702) — R1 V4
- ***AUDIX Enhanced III Wallet Card*** (585-304-703) — R1 V4
- ***AUDIX Enhanced III User's Reference*** (585-304-704) — R1 V4.

Subscribers access the Voice Mailbox feature by logging on to AUDIX. They may use a touch-tone voice terminal (recommended) or a rotary telephone if they use a tone generator. The main AUDIX extension number routes to an ACD (Automatic Call Distribution) or EUCD (Enhanced Uniform Call Distribution) (on System 85, Release 2, Version 2 switches) split. ACD (or EUDC) software routes calls to an available AUDIX voice port If all the ports are busy, the call is queued and users hear ringback (optionally, followed by a recorded announcement) until a port becomes available.

When AUDIX answers, subscribers log on by entering their extension number and personal AUDIX password. The individual passwords control access to each subscriber's messages, allowing only authorized users to access the messages. In AUDIX R1 V4 software, a minimum password length may be assigned [see the ***AUDIX Enhanced III (R1 V4) Features*** section].

AUDIX next reports if any new messages were received and indicates what type they are (for example, Call Answer or Voice Mail). If the subscriber is low on space, a warning Message is given. AUDIX then voices three frequently used selections from the Activity Menu. Subscribers may type several commands in order without waiting for a prompt ***(dial-ahead)*** or may interrupt announcements ***(dial-through)*** to immediately begin AUDIX activities. Subscribers may request more detailed instructions at any time by using the *H (Help) command.

Subscribers can also reach the Voice Mailbox through the Call Answer feature. Before or after leaving a message, subscribers can press *R (Restart) to play the AUDIX welcome message. They may then access their own mailbox (as described above) or another subscriber's mailbox using a guest password. Subscribers can also log on to the same call one after another using the **R (Relogin) command (this can save the second subscriber from making a long-distance call).

Considerations

The following considerations apply to AUDIX and System 85 R2 V4 and Generic 2 PBXS.

Attendant Console

The attendant cannot operate the Transfer Into AUDIX feature because the access code cannot be dialed from the console. Attendant console calls cannot be forwarded to AUDIX on System 85 or Generic 2.

AUDIX Adjuncts

For R2 V2 and V3 switches, up to four AUDIX adjuncts can be attached to a single switch. Beginning with R2 V4, up to eight AUDIX adjuncts can be attached to a single switch due to flexible DCIU port assignments. The AUDIX Networking feature (AUDIX R1 V3 and later software) allows multiple AUDIX adjuncts to exchange voice mail messages.

AUDIX Parameters

TABLE 14A. Comparison of AUDIX Parameters

AUDIX Model	Number of Subscribers (Light Usage)	Number of Subscribers (Heavy Usage)	Number of Voice Ports	Hours of Voice Storage
One-Cabinet AUDIX	2000	750 to 1000	16	14 to 240
Two-Cabinet AUDIX	4000	1000 to 2000	32	14 to 480
AUDIX-L	4000	1500	32	200

Networking Differences Between Module Types

AUDIX Adjuncts supplied with early DEFINITY Communications Systems, Generic 2 were equippd with TN366 ACC (AUDIX Communications Controller) boards. The ACC provides switched connections between AUDIX adjuncts in AUDIX Networking arrangements. The TN366 supports communications on four I-Channels when connected to traditional modules, however, they support communications on only two I-Channels when connected to a universal module. Later models are equipped with TN366B ACC boards. The TN366B ACC board is interchangeable with the TN366 and supports communications on four I-Channels for both the traditional and universal modules.

Redirection to AUDIX

If the call is transferred to an alternate destination that has Call Coverage active, the call could be redirected back to AUDIX through Enhanced Call Transfer. If the call is transferred out of AUDIX to an unattended alternate destination that does not have Call Coverage active, the terminal can ring indefinitely. The transferring party must end the call. The call will ***not*** automatically return to AUDIX.

Reorder Tone

If the DCIU link or every voice port is in an out-of-service condition, reorder tone is heard by subscribers dialing the AUDIX extension number, or by parties attempting a Transfer Into AUDIX.

Trunk Availability

If no outgoing trunks are available from the host switch, a transferring party hears an "all circuits busy" message from AUDIX. Users are then able to leave a message or request another transfer. AUDIX subscribers are returned to the previous activity.

If no outgoing trunks are available from an intermediate remote switch (tandem switch) to the alternate destination remote switch, the transferring party hears a reorder tone from the tandem switch. In this case, a transferring party must end the call.

Interactions With other Features

The following System 85 and Generic 2 features affect or are affected by AUDIX operation.

Abbreviated Dialing

An abbreviated dialing button can be programmed to contain the AUDIX dial access code for the Transfer Into AUDIX feature.

ACD (Automatic Call Distribution)

The ACD feature in combination with the Call Coverage feature is used to direct calls to AUDIX. The AUDIX Adjunct is assigned as an ACD split and this split is then assigned as the final point in a coverage path.

Call Coverage

An AUDIX system is used as the final point in a coverage path. The ACD split number, rather than an associated extension number, is used to assign the coverage point. Coverage calls that are redirected to Call Answer by the AUDIX system are placed in the nonpriority queue.

Call Coverage and Call Vectoring

A VDN (Vector Directory Number) associated with a vector that queues calls to an AUDIX split can be assigned as the final point in a coverage path. When this is done, the full flexibility of the Call Vectoring feature can be applied to the redirected call. An AUDIX vector's processing could vary by time of day (to provide night service), by the status of the AUDIX split's queue (to provide intraflow or interflow), or by priority (to queue at any of four levels of priority).

If the AUDIX system has answered a call and Transfer Out of AUDIX is used to transfer the AUDIX call to a VDN, the Call Vectoring feature screens these transfers to limit undesirable vector treatment of the transferred call. A Call Vectoring subroutine quickly scans the VDN's vector on a per-call basis to be sum that one of a set of vector steps will operate on the call before the transfer is allowed. These steps include:

- Queue to main split (with staffed agents) step
- Route to step
- Forward disconnect with recorded announcement step (beginning with R2 V4, Issue 1.2).

Call Forwarding and Call Coverage

Call Forwarding and Call Coverage can be assigned to an AUDIX subscriber to redirect calls when the subscriber is unavailable. However, when both features are assigned, Call Forwarding takes precedence over Call Coverage (calls will be redirected according to Call Forwarding). If a coverage point has Call Forwarding—Follow Me active, that point is temporarily removed from the coverage path.

Call Forwarding—Busy and Don't Answer

The Call Forwarding—Busy and Don't Answer feature and the Call Forwarding—Don't Answer feature cannot be used to forward calls to AUDIX. If attempted, the switch returns intercept tone.

Call Forwarding—Follow Me

The Call Forwarding—Follow Me feature can be used to forward all calls to the main AUDIX extension number. Forwaded calls enter the AUDIX queue. When AUDIX answers, it plays the subscriber's personal greeting.

Call Vectoring

A vector can have a "queue to main split" step or "route to" (AUDIX extension number) step that routes calls to an AUDIX split. However, if the DCIU link is down or if every voice port to that AUDIX is unstaffed, these calls are not directed to the AUDIX queue. Instead, the switch returns reorder tone to the calling party.

Display—Voice Terminal

During message retrieval using a display unit, the display will notify the user whenever there are also AUDIX messages to be retrieved. If this notification contains the AUDIX-associated extension number [or VDN (Vector Directory Number)], the user can press the RETURN CALL button to easily access the AUDIX system.

DCS (Distributed Communications System)

DCS transparency is not provided for the Transfer Out of AUDIX feature in the following cases:

- If the transfer destination is on a remote DCS switch, and if the extension is busy and call coverage has not been activated, callers hear a busy tone and are not returned to AUDIX.

- If an outgoing trunk is not available from an intermediate remote switch to the destination remote switch, callers hear reorder tone and must end the call.

For complete information on AUDIX feature operation in a DCS network, see the ***AUDIX Reference*** manual (585-300-201).

Extension Number Portability

When an extension is ported from one node in a portability subnetwork to another and a centralized AUDIX serves both switches, the centralized AUDIX system must be informed of the move.

IPA (Interpartition Access)

See the "Tenant Services" interaction in this section.

Last Extension Dialed

The Last Extension Dialed feature can redial calls to an AUDIX-associated extension number.

Last Number Dialed

The LND (Last Number Dialed) feature can redial calls to the associated extension number for AUDIX. However, the LND feature does not redial any digits used after the caller accesses AUDIX (including the subscriber's extension and password.

LWC (Leave Word Calling)

If the principal's LWC messages are stored on the AUDIX system, the originator of an LWC message cannot cancel the message. If the originator tries to cancel the message, the switch will accept the request with confirmation tone. However, the LWC message is not removed from the principal's AUDIX mailbox.

Look-Ahead Interflow

Using the Look-Ahead Interflow feature, a "route to" step within a sending vector can divert calls from a local AUDIX system to the VDN of an alternate destination outside the switch (usually within the private network).

Using the Look-Ahead Interflow feature, a "route to" step within a sending vector can divert direct or redirected calls from a local destination to the VDN of a centralized AUDIX system within the DCS subnetwork. [When a diverted call undergoes DCS routing, the distant switch receives the reason for redirection, the identity of the originally called principal, and IMN (Integrated Message Notification) information in the DCS message.]

Message Center Service

The AUDIX system can be used as a form of night service for a Message Center split. The Message Center split supervisor can use Call Forwarding—Follow Me to forward all

Message Center calls to the main AUDIX extension number. Refer to the ACD feature description for details.

Modem Pooling

Early Release 1, Version 4 AUDIX adjuncts are equipped with TN366 ACC (AUDIX Communications Controller) boards. The TN366 is used to set up and operate the ACC link between AUDIX adjuncts in an AUDIX Networking arrangement. The ACC link is used for binary file transfer between AUDIX adjuncts and appears to the switch as a trunk data call. Based on the trunk facilities available for these calls (analog, digital or voice grade digital) a Modem Pooling conversion resource may be required on these links. If the switch uses 7400A Data Modules in its Modem Pooling conversion resources, there will be a problem if Modem Pooling is used on an ACC link. The TN366 ACC board does not function properly with the 7400A Data Module. In these cases, the TN366 ACC can be replaced with the TN366B ACC board which will work properly with the 7400A Data Module. The TN366B replaced the TN366 on later production models of the Release 1, Version 4 AUDIX adjunct.

Tenant Services

A partitioned System 85 or Generic 2 is not aware of AUDIX user permissions. When a voice terminal user dials the AUDIX extension number, the switch follows the usual rules for terminal-to-terminal calling to reach the AUDIX system. Therefore, the AUDIX extension number must either belong to the users partition or to Extension Partition 0. After reaching the AUDIX system, message can be left for, created by, or retrieved by any subscriber (regardless of the extension partition the subscriber belongs to).

The Transfer Into and Out of AUDIX functions are also partitioned. A voice terminal user in an extension partition other than Extension Partition 0 can transfer AUDIX calls to extension numbers in the same extension partition or to extensions in Extension Partition 0. When the user tries to transfer these calls to any other extension partition using an extension number, the switch returns intercept tone to the transferring party. A voice terminal user in Extension Partition 0 is allowed to transfer AUDIX calls using an extension number to any voice terminal in the switch.

Transfer Into AUDIX

The Transfer Into AUDIX feature can be used for calls that have been redirected by the following features: Call Coverage, Call Forwarding, Call Pickup, Hunting, and Unattended Console Service.

Unified Messaging

After logging on to the AUDIX syste, the AUDIX system voices the subscriber's name and annouces any new messages. On fully integrated systems, AUDIX also provides unified messaging by reporting unaccessed messages on other message services [such as EDC (Electronic Document Communications), MCS (Message Center Service), or switch LWC (Leave Word Calling) messages].

Hardware Requirements

For an overview of AUDIX hardware, see the ***AUDIX Reference*** manual (585-300-201). For complete AUDIX system maintenance information, see the appropriate AUDIX maintenance manual:

- ***AUDIX-L Maintenance*** manual (585-300-102)
- ***AUDIX Enhanced III Maintenance*** manual (585-304-102)
- ***AUDIX Maintenance*** manual (585-300-106)
- ***AUDIX Enhanced III Maintenance*** manual (585-304-106)
- ***AUDIX Enhanced III Forms Reference*** manual (585-304-202).

Switch Interfaces

AUDIX connects to the switch through the following interfaces:

- ***Alarms Link:***

  Major and minor alarm connections run from AUDIX to existing alarm facilities on the switch. The alarms alert service personnel who normally monitor the system about possible problems with AUDIX.

- ***Data Link:***

  A fully integrated AUDIX system includes a data link, which is used to exchange nonvoice information between the switch and AUDIX such as LWC messages, extension identification, automatic message-waiting lamp status, and call connect / disconnect information. The System 85 and Generic 2 data link is a DCIU link. AUDIX Stand-alone systems do ***not*** use a data link.

- ***Maintenance Link:***

  Service perosonnel can access AUDIX remotely through a modem (usually a 2212 or equivalent). Usually the modem is connected to AUDIX and an analog port on the switch which has an extension number accessible by a CO line leading to the remote site. The AUDIX maintenance connector is also used for local service as needed.

- ***Networking:***

  For AUDIX Networking, a special DCP (Digital Communications Protocol) link is set up that is used to transfer files between the AUDIX adjuncts involved. This is done by the ACC (AUDIX Communications Controller) board on the AUDIX adjuncts. These links are switched and look like data trunk calls to the switch. As such, Modem Pooling conversion resources may be required, depending on the trunk facilities used for the call.

  Release 1, Version 3 and early Version 4 AUDIX Adjuncts were equipped with TN366 ACC boards for AUDIX networking. These TN366 ACC boards support four ACC links when connected to a traditional module (System 85 or Generic 2 switches) but only two ACC links when connected to a universal module on a

Generic 2 switch. On later V4 AUDIX Adjuncts, the TN366 ACC was replaced by the TN366B ACC Prime board which supports four ACC links on either type of module.

- ***Voice Links:***

  AUDIX may have from 2 to 32 voice ports connected to an equivalent number of analog voice ports on the switch. Each group of eight ports on the AUDIX requires one 25-pair cable connection to the switch or cross-connect field. The switch ports are set up in one or more call-distribution or hunt groups which route calls to idle ports on AUDIX as calls are received.

  — ***For Traditional Modules:***

  One or more of the following analog line circuits are required to support the AUDIX Outcalling feature.

  SN222 Analog Line Circuit Pack

  SN228B Analog Line Port Circuit

  SN229 Analog Line Circuit Pack

  — ***For Universal Modules:***

  One or more TN742 analog line circuits are required.

Subscriber Interfaces

Each subscriber accesses the AUDIX system using touch-tone voice terminals. Touch-tone signals are used to enter all commands on AUDIX

Feature Administration

Many AUDIX features must be administered on both AUDIX and the switch. Features which are administered on the AUDIX only are covered in the AUDIX administration manuals (see the next section). Steps for administering a switch to work with AUDIX are covered in the AUDIX installation manuals as follows:

- ***AUDIX-L Installation*** manual (585-300-101), which covers R1 V2 and R1 V3 software. Addendum 1 covers R1 V4 enhancements.
- ***AUDIX-S Installation*** manual (585-300 -105), which covers R1 V2 software on early one-cabinet AUDIX systems.
- ***AUDIX Installation*** manual (585-302-105), which covers the one- and two-cabinet AUDIX cofiguration and R1 V3 software.
- ***AUDIX Enhanced III Installation*** manual (585-304-105), which covers the one- and two-cabinet AUDIX models and R1 V4 software.

The following features should be administered using the appropriate AUDIX installation manual to enable AUDIX operation.

a. **ACD:**

The ACD feature (previously called EUCD) serves as the gateway to the AUDIX system. ACD administration is covered in the ACD chapter of this manual.

NOTE: An AUDIX adjunct can have from 2 to 32 voice ports. Therefore, an ACD gateway split should be configured to contain up to 32 members.

b. **Call Vectoring:**

For R2 V4 System 85 and Generic 2, the Call Vectoring feature can be used in conjunction with the ACD feature on a per-system basis. When this is done, some of the standard ACD functions are replaced by the more flexible vector processing. Call Vectoring administration is covered in the Call Vectoring chapter of this manual.

c. **Touch-Tone Dialing:**

The Touch-Tone Dialing feature serves as the subscriber interface to the AUDIX system. The administration for touch-tone voice terminals is covered in the Touch-Tone Dialing chapter of this manual.

d. **Leave Word Calling:**

The Leave Word Calling feature is available on the AUDIX adjunct. Leave Word Calling administration is covered in the Leave Word calling chapter of this manual.

e. **Call Coverage:**

The AUDIX system can be assigned as the final point in a coverage path. Call Coverage administration is listed in the Call Coverage chapter of this manual.

f. **Call Forwarding:**

The AUDIX system can be assigned as the destination for Call Forwarding. Call Forwarding administration is covered in the Call Forwarding—Follow Me chapter of this manual.

DCIU Administration

Refer to Appendix H of this manual for detailed information about DCIU administration. This appendix includes some recommendations for typical administration of the DCIU.

System 85/Generic 2 DCIU Administration

		ADMINISTRATION PROCEDURES	
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
256	1	Administers the DCIU link characteristics.	No
256	2	Adminsters level 2 timers and counters for a DCIU link.	No
256	3	Administers level 3 timers and counters for a DCIU link.	No
257	1	Administers the components and priority status of a DCIU channel.	No
257	2	Administers DCIU ports for the DCIU channels.	No
257	5	Reserves a DCIU port for use by an AUDIX system (R2 V4 and Generic 2 software).	No
257	6	Sets up external adjunct characteristics for passing ES messages for AUDIX in a DCS network (R2 V4 and Generic 2 software).	No
258		Swaps (R2 V4 or Generic 2) or copies translation changes made using Procedures 256 and 257.	No
258	2	Refreshes the DCIU temporary translation tables before using Procedures 256 and 257.	No
261	1	Provides the translation between an internal AP or AUDIX number and the network AP or AUDIX number.	Yes
261	2	Assigns the associated extension for each network number.	Yes
275	1	Assigns DCIU to the system class of service.	Yes
350	2	Assigns the Transfer Into AUDIX dial access code. The applicable encode is: 58 Call Transfer to AUDIX.	No

AUDIX Administration

AUDIX system administration is summarized for all main features in the ***AUDIX Reference*** manual (585-300-201). For detailed AUDIX system administration procedures, see the following documents:

- ***AUDIX Administration*** manual (585-302-501) for AUDIX R1 V2 and R1 V3 software
- ***AUDIX Enhanced III Administration*** manual (585-304-501) for AUDIX R1 V4 software
- ***AUDIX Enhanced III Forms Reference*** manual (585-304-202) for screen forms in AUDIX R1 V4 software.

Authorization Codes

Description

Voice terminal users and attendants can use authorization codes in conjunction with the FRL (Facilities Restriction Level) feature. These codes add flexibility and improve the effectiveness of the FRL feature. Authorization codes are normally used with the Remote Access feature on an incoming basis and with the networking features (AAR [Automatic Alternate Routing], ARS [Automatic Route Selection], and WCR [World Class Routing]) on an outgoing basis. However, authorization codes can also be used for billing purposes as a "movable" personal identifier.

The FRL (Facilities Restriction Level) Feature

The FRL feature protects network facilities from unauthorized use and limits access to critical facilities. An FRL is assigned to each originating facility (class of service, attendant group, and incoming trunk group). This is the primary FRL (also known as the default FRL). When a user places a call, the switch selects a routing pattern based on the dialed digits and other parameters. Then, the switch compares the FRL associated with the call (the default FRL of the originating facility) to the FRL of the first preference [a trunk group with certain characteristics (including FRL) defined by the pattern where it is assigned]. A call can only access preferences that have FRLs equal to or lower than the FRL of the call. Within the routing pattern, the switch searches the preferences until an accessible preference with an available trunk is found. If no preferences with available trunks are accessible to the default FRL, the switch prompts the caller (with recall dial tone) for an authorization code. For incoming intertandem trunk calls, the user is never prompted for an authorization code.

Authorization Code Functions

To Change the FRL of a Call

When authorization codes are used, they provide an option to the default FRL. The switch requests an authorization code only once during a call. If the caller enters an authorization code with an FRL that is higher than the default FRL, the switch uses the FRL assigned to the authorization code. Using the new FRL, the switch makes a second attempt to route the call. If this attempt fails and queuing is assigned for an accessible preference, the switch tries to queue the call on the accessible preferences within the preference depth for queuing (Pattern Queuing assigned) or on the best choice trunk group (regular Queuing assigned).

Private-network trunk groups often have low FRLs. Public-network trunk groups and authorization codes often have higher FRLs. This is based on the assumption that long-distance calling over public-network facilities is more expensive than if the same call were placed over private-network facilities. [This may not always be true when MEGACOM® WATS Service, FX (Foreign Exchange) Service, or WATS (Wide Area Telecommunications Service) trunks are available.] Therefore, most employees are allowed to place private-network calls, but only employees that have access to a high enough FRL can access the public toll network.

High FRLs limit access to premium facilities such as expensive public-network trunks, international service facilities, or sensitive private-network trunk groups (such as high-speed digital trunks to data processing centers). Incoming trunk groups [such as tie trunks (except intertandem tie trunks) and remote access trunks] can be set up so that the switch always requests an authorization code. An authorization code allows a caller to override the FRL assigned to the terminal that the caller is using temporarily, for instance, another person's terminal or a general use terminal.

For Remote Access

Authorization codes can also be used as an alternative to Barrier Codes with the Remote Access feature. If the Authorization Code option is used for Remote Access, and an authorization code is later needed for FRL purposes, the authorization code does not need to be dialed again. The switch uses the same authorization code entry for FRL purposes.

Network Access Flag

The network access flag indicates whether or not the authorization code can be used ***from an off net location*** (such as through the Remote Access feature). When an Authorization Code is used for control of the Remote Access feature, and when the Remote Access feature is used to gain access to outgoing trunks, the network access flag (administered for each authorization code in Procedure 282, Word 1, Field 3) becomes a significant consideration.

For an authorization code to be useful from any off-net location (for example with a Remote Access or APLT call), the network access flag must be set to "1." If the nework access flag is set to "0," that authorization code cannot be used to gain access to the Remote Access feature or on an attendant extended call from a public or APLT trunk. Also, if a barrier code is used to access the Remote Access feature and the network access flag is set to "0," that authorization code cannot be used to gain access to an outgoing network trunk via the Remote Access feature through a networking features (AAR, ARS, or WCR). An authorization code with the network access flag set to "0" can only be use from a local station or main/satellite trunk (these calls are considered on net by the switch Software).

Alternate FRLs

Each primary FRL can be assigned an alternate FRL value. To activate or deactivate the alternate FRLs, the attendant simply presses the ALTERNATE FRL button on the attendant console. When alternate FRLs are in effect, they replace default FRLs. This is true for the FRLs assigned to authorization codes as well as default FRLs. The alternate FRLs can be more restrictive, less restrictive, or the same as the default FRL. Alternate FRLs are normally used during "off hours" when a different set of facility restrictions and controls may be desirable. The FRLs assigned to AAR, ARS, and WCR routing patterns, and the UCC (Unauthorized Call Control) FRL do not change.

Feature History and Development

The Authorization Code feature was first available in System 85, Release 1. Subsequent changes and enhancements have included the following:

- ***Increase in Authorization Codes Available***

  The number of authorization codes available in a single switch was increased from 9,000 to 90,000. This enhancement was first available with Release 2, Version 3.

- ***Validation Algorithm for Authorization Codes***

  In switches prior to Release 2, Version 3 (Release 2, Version 2 and earlier), the switch validates authorization codes using a predefined algorithm.

  Seed Digits, Check Digits, and Check Sum Digits:

  Each authorization code contains either three or four "seed digits" and from one to three "check digits." Additionally, a "check-sum digit" is defined for each check digit.

  Using this algorithm, the seed digits are always the ***leading digits*** within a code, while the check digits always ***follow*** the seed digits. Each check-sum digit is the modulus 10 sum of a designated subset of seed digits and a check digit. Table 15-A shows the available options for assigning a switch's authorization codes.

TABLE 15-A. Authorization Code Parameters

Number of Digits in Authorization Codes	Number of Seed Digits	Number of Check Digits	Number of Check-Sum Digits Needed
4 Digits	3	1	1
5 Digits	4 3	1 2	1 2
6 Digits	4 3	2 3	2 3
7 Digits	4	3	3

- ***Random Selection of Authorization Codes***

  In Release 2, Version 3 and later switches, the Validation Algorithm is no longer used. Instead, users are free to establish their own 4-, 5-, 6-, or 7-digit authorization codes on a random basis, or on whatever basis best suits their needs. For example, the last four to seven digits of an individual's social security number, payroll account number, birth date, or personal extension number could be used. Coding schemes of this sort make it easier for users to remember their individual authorization codes. However, these switches are somewhat less secure than the Validation Algorithm or a pure random number system.

User Operations

The following are the user operating procedures for this feature

To Enter an Authorization Code

An attendant or voice terminal user should:

1. Place the call in a normal fashion. [Call-progress tone]

   [If an Authorization Code is needed, recall dial tone (three bursts of tone, followed by dial tone) is returned.]

2. Enter the assigned Authorization Code. [Call-progress tone continued.]

To Escape the Authorization Code Sequence

If an authorization code is not available, after the authorization code has been requested by the switch the attendant or voice terminal user should:

1. On a System 85 or Generic 2.1 switch, dial the escape character "1" or wait for 10 seconds. [The switch skips the authorization code step if a 1 is dialed or times out after 10 seconds, and then proceeds with call processing, using the default FRL.]

2. On a DEFINITY Generic 2.2 switch, dial an escape character "1" or "# ." [The switch skips the authorization code step and proceeds with call processing, using the default FRL; 10-second time out is not used.]

Considerations

10-Second Timer

As the switch requests an authorization code with recall dial tone, it also sets a 10-second timer. An authorization code must be dialed before the timing elapses for the authorization code FRL to route the call. Otherwise, if an escape character (a 1 on System 85 or Generic 2.1 switches, or either a 1 or # sign on Generic 2.2 switches) is dialed, or the timer elapses, the switch retries the call using the original FRL.

Digits Required

Authorization codes are required to have at least four digits. Up to seven digits can be used. Every authorization code assigned to a particular switch (or to all nodes in a DCS) must have the same number of digits.

Hard Processor Swaps

Authorization codes and their corresponding FRLs are stored in a translation portion of switch memory. Therefore, these codes and their assigned FRLs will endure a hard processor swap.

Intertandem Tie Trunk Calls

Normally, for incoming calls from intertandem tie trunks that are tandeming through the local switch (to an outgoing trunk), the tandeming switch does not prompt for an authorization code. This is because the calling party may have already entered an authorization code at the previous switch and the FRL TCM is then the highest available for the call. Also, the list of valid authorization codes may be different at each switch within the network, and the calling party would have no way of knowing which switch is requesting the authorization code.

Normally, when the FRL for these calls is too low to access an available trunk, the tandem switch either:

- Queues the call to busy accessible preferences in the preference depth (when Queuing is assigned)
- Returns reorder tone if the accessible preferences are busy (when Queuing is not assigned)
- Returns intercept tone if the FRL is too low to access any preference.

However, a Generic 2.2 switch (translated with encode "0" in field 3 of Procedure 103, Word 1) could prompt for an authorization code if a higher FRL is needed for route selection. If this translation is used, care must be taken to ensure that valid authorization codes from the originating switch will be recognized and handled properly at the tandem switch. If Procedure 103, Word 1, field 3 is translated with either a "1" or "2," the switch will not prompt for an authorization code.

Limits

In System 85s, prior to Release 2, Version 3, there can be up to 9000 authorization codes.

Beginning with Release 2, Version 3, a System 85 can have up to 90,000 authorization codes.

The first digit of an Authorization Code can be any digit except the digit "1" The digit "1" is used to direct the switch to try again with the same FRL. If this digit is used at the beginning of an authorization code, the switch will "try again" when the first digit is dialed, and the rest of the authorization code will not be read.

Rotary Dialing Terminals

On internal calls (calls originated on a local switch or within a DCS or Main/Satellite arrangement), there are no problems with using either a rotary or touch-tone dialing terminal.

However, if an incoming call [over an automatic CO (central office), DID (Direct Inward Dialing), or Remote Access trunk] requires an authorization code, a rotary dialing terminal cannot be used. A touch-tone dialing terminal or equivalent (such as a data terminal using the keyboard dialing function) must be used to enter an authorization code on incoming calls. Otherwise, the call must be placed using attendant assistance. A rotary dialing terminal cannot dial an authorization code over an automatic CO, DID, or Remote Access trunk.

Reviewing the Assigned Authorization Codes

For switches that use the validation algorithm (R2 V2 and earlier), the assigned authorization codes can be reviewed in Procedure 282, Word 1. When this isdone, the administration device displays these authorization codes in numerical order.

However, beginning with R2 V3, the method for storing authorization codes in switch memory changed. The assigned authorization codes can still be reviewed in Procedure 282, Word 1, but the sequence of these displayed codes is no longer numerical. Instead, the Manager II, MAAP, or SMT displays the codes in the ***same order*** that they were entered. Displaying the authorization codes by order of entry gives the display an appearance of randomness.

Release 2, Version 3, Authorization Codes

Beginning with Release 2, Version 3, the authorization codes are assigned by the switch administrator rather than provided by the switch. An authorization code can be from four to seven digits. On a single switch, all authorization codes must have the same number of digits. Procedure 281 is used to enter the number of digits that are to be used and which digits, if any, to delete. Table 15-B shows recommended parameters for authorization codes.

TABLE 15-B. Recommended Digits for Authorization Codes

Number of Digits	Number of Possible Codes	Recommended Number of Actual Codes
4 Digits	9,000	1 to 90
5 Digits	90,000	91 to 900
6 Digits	900,000	901 to 9,000
7 Digits	9,000,000	9,001 to 90,000

When a user dials an authorization code (during a calling sequence), the switch searches for that code and its associated FRL and Network Access Flag. Three parameters are used by the switch to store and search for valid authorization codes. If the authorization codes are set up properly, these parameters help locate a valid authorization code.

Procedure 281 is used to tell the switch processor how many digits are to be dialed, and if more than four are used, what digits to delete. If a number scheme is used that inherently has common digits (for example, payroll numbers that include department numbers or personal extension numbers that include a common prefix), it is best to delete the most duplicated digits as this reduces the chances of duplicating authorization codes.

Security

There should be enough digits within the authorization codes to expect probable failure for a person attempting to find a valid code by trial and error. A good margin of safety would be provided if the ratio of ***possible codes*** to ***actual codes > = 100:1.*** For this ratio, table 15-B shows the recommended digit assignment for authorization codes.

Traffic Measurements

Whenever an invalid authorization code is received from a voice terminal, remote access

Interactions With Other Features

The following System 85 Generic 2 features affect or are affected by the operation of this feature.

FRL (Facilities Restriction Level)

An FRL is assigned to each authorization code. This FRL can then be used instead of the default FRL assigned to the terminal being used.

IPA (Interpartition Access)

For switch security, strict partitioning is applied to Authorization Codes. Authorization Codes can only be assigned to an extension partition (not to a partition group), and these codes cannot be shared by the other extensions in a different extension partition.

Last Number Dialed

For switch security, the Last Number Dialed features does not store or redial Authorization Codes.

Look-Ahead Interflow

The Authorization Codes feature does not apply to Look-Ahead Interflow calls. The R2 V4 System 85 or DEFINITY Generic 2 does not return recall dial tone to request an Authorization Code for Look-Ahead Interflow calls.

If the VDN associated with the sending switch's vector does not have a high enough FRL to access any preference in the routing pattern determined by the destination digits of the "route to" step, the Look-Ahead Interflow software considers the "route to" step as having an invalid destination. (If the invalid "route to" step is the final effective step in the vector, the step is treated as a "stop" step. Otherwise, the sending switch continues vector processing with the next sequential step.)

Queuing

When the Authorization Code feature is active and Queuing is provided, routing feature calls (AAR, ARS, or WCR) may queue.

Remote Access

The Remote Access feature can use an authorizatition code instead of a barrier code to control access. If the Enhanced Remote Access Security Speaker Verification option is used (Release 2, Version 3 or later), an authorization code must be used for access to the Remote Access feature, a barrier code cannot be used. The Speaker Verification adjunct uses feature dial access codes (encodes 103 and 104 in Procedure 350, Word 2) to communicate the status of authorization code entries to the switch.

When an authorization code is used, no further request is made for an authorization code. That is, if an authorization code is used to gain access to the Remote Access feature, and if an authorization code is needed later to complete the call the same entry used for access to Remote Access, is used to complete the call. If a barrier code is used to gain access to the Remote Access feature, an authorization code may be requested later to complete the call. In this instance, the authorization code must be entered separately. In either case, if the network access flag for the authorization code is set to "0," the call is denied. If the network access flag is set to "1," the call precedes normally (based on the FRL assigned to the authorization code).

Tenant Services

In Procedure 281, Word 1, the amount of digits used in authorization codes is assigned on a system-wide basis. Therefore, every partition in a partitioned switch must use the same amount of digits.

In Procedure 282, Word 1, each authorization code is assigned to an extension partition. Users in different extension partitions are not allowed to share an authorization code, and the same code cannot be duplicated for use in more than one partition.

When a voice terminal user in a specific partition dials an authorization code, call-processing software checks to determine whether the code's associated partition matches the partition of the voice terminal used to place the call. If these partitions do not match, the switch returns intercept treatment to the calling party.

WCR (World Class Routing)

The Authorization Code feature is used with the World Class Routing feature in the same way it was with the earlier networking features, AAR and ARS. An authorization code provides a means of possibly raising the FRL of a call (on an individual caller basis) when the default FRL for the call is not high enough to access additional routes that may be available to a higher FRL.

When the World Class Routing feature is being used, an escape entry is available for the case when the caller is prompted for an authorization code to try and raise the FRL of the call. If the caller does not have an authorization code or for some reason doesn't want to use it, dialing a "1" or a "#" will cause the switch to skip that step and avoid the 10-second time out that would otherwise occur if no authorization code is dialed.

Hardware Requirements

None.

Feature Administration

Assignment of the Authorization Codes feature is on a per-switch basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or FM (Facilities Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable MAAP and SMT procedures for Release 2, Version 2, and earlier switches.

SYSTEM 85 RELEASE 2, VERSION 2 AND EARLIER MAAP AND SMT PROCEDURES — AUTHORIZATION CODES			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
103	—	Administers network trunk group parameters including authorization code requirements for incoming tie trunks.	Yes
281	1	Administers the authorization code algorithm.	No
282	1	Assigns an FRL and network access flag to a single authorization code.	Yes
282	2	Assigns an FRL and network access flag to a group of consecutive authorization codes.	Yes
282	3	Displays and changes the FRL for a group of up to ten consecutive authorization codes. The FRLs do not have to be the same.	Yes
282	4	Administers the public (off-net) to private (on-net) network access flags for up to ten consecutive authorization codes.	Yes
283	1	Displays the authorization codes, extension numbers, and trunk groups associated with a specific FRL.	Yes

The following are the applicable administration procedures for System 85, Release 2, Version 3 and 4, and DEFINITY Generic 2.

ADMINISTRATION PROCEDURES
AUTHORIZATION CODES FEATURE

PROCEDURE	WORD	PURPOSE	SMT
103	1	Administers network trunk group parameters including FRL and authorization code requirements for incoming tie trunks.	Yes
281	1	Administers the authorization code algorithm parameters. This includes the number of digits in an authorization code and which digits, if any, to delete.	No
282	1	Assigns a single authorization code, its FRL, network access flag and extension partition (if applicable).	Yes
282*	2	Displays the number of authorization codes that have been entered in the switch.	Yes
283	1	Displays the authorization codes, extension numbers, and trunk groups associated with a specific FRL.	Yes
285	1	Assigns system class of service features including authorizatior code requirements for network access and for the Remote Access feature, symmetrical routing depth, and WCR routing digit collection characteristics.	
350	1&2	Assigns feature dial access codes as required. Encodes pertaining to the Authorization Code feature (used with Speaker Verification) are: 101 Speaker Verification — Accept** 102 Speaker Verification — Fail 103 Unadministered Authorization Code Entered 104 No Authorization Code Entered. These dial access codes are used on R2 V4 and Generic 2 switches, when the Enhanced Remote Access Security/Speaker edification feature is used to screen Remote Access calls. They are used by the Speaker Verification adjunct to communicate its results to the switch.	Yes

* Procedure 282 was changed for R2 V3 and later switches. The MAAP and SMT procedures for R2 V2 and earlier switches are shown in a separate table.

** Speaker Verification acceptance implies that a valid authorization code was entered by the Remote Access caller.

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — AUTHORIZATION CODES	
PATH NAME	PURPOSE
terminal-change system parameters (select the Access-Codes option)	Enables the use of Authorization Codes for AAR/ARS.

The following is the applicable FM path name used with the AP 16.

FM SCREEN — AUTHORIZATION CODES	
PATH NAME	**PURPOSE**
facilities-mgmt facility-rest authorization-codes	Displays and changes the FRL and network access status associated with an authorization code(s). A printed report can also be generated.

Notes:

Automatic Alternate Routing

Description

The Automatic Alternate Routing (AAR) feature provides alternate routing for private-network calls. With the Automatic Alternate Routing feature, more than one route can be available for a call to reach its find destination.

Feature History and Development

The AAR feature was first available on System 85, Release 1. The following enhancements have been made to this feature since its introduction:

- During Release 1, System 85 switches allowed for as many as 180 AAR routing patterns. In System 85, R2 V1, the maximum number of patterns was increased from 180 to 255. In System 85, R2 V3, the maximum number of patterns was increased from 255 to 640.
- In System 85, R2 V3, the maximum number of preferences in a pattern was increased from 4 to 16.
- In System 85, R2 V3, Conditional Routing, the first implementation of Generalized Route Selection was added to limit the number of satellite links (or other "restricted use" facilities) used in routing a private-network call.
- In System 85, R2 V3, Pattern Queuing was added so that a call can queue on any preference in a pattern.
- In System 85, R2 V3, subnetwork trunking was enhanced to provide IXC Access (customer-defined access to any long-distance vendor). The following are included in this enhancement.
  - Seven digits can be deleted from the front of the destination code (no change).
  - Twenty digits can be inserted in the place of the deleted digits (increased from 12).
  - Digits can be sent in up to four groups (no change).
  - Each group can hold a maximum of 15 digits (increased from 7).
- In System 85, R2 V4, Generalized Route Selection was expanded to allow AAR routing for calls according to each call's Bearer Capability Class (BCC). Also, AAR calls were allowed to route according to an ISDN trunking services designation: ISDN required, ISDN preferred, or any facility accepted.

- In DEFINITY Generic 2, BCC routing by the AAR feature was expanded to allow routing according to as many as 256 Bearer Capability Classes of Service (BCCOSs). Also for data calls, the preference-selection process for BCC routing is enhanced. The AAR preference-selection software chooses better kinds of trunk facilities and inserts Modem Pooling conversion resources (when necessary) to route data calls.
- In System 85, R2 V4, Issue 2.0 and DEFINITY Generic 2.1, Issue 2.0, Selective International (011) Routing is added. This enhancement supports customers with international private line service by allowing the switch to examine "01X" plus the digits that follow in order to make routing decisions.

In Generic 2.2 the AAR feature is discontinued as it has been replaced by the World Class Routing (WCR) feature.

Routing Structures

Patterns

For calls between two points (switches) possible routes are grouped into routing patterns. ***Routing patterns*** are ordered lists of preferences (trunk groups) thatthe switch can use to complete a particular call. There can be up to 640 AAR patterns.

Preferences

Each routing pattern has from 1 to 16 preferences. A ***preference*** is a trunk group assigned. to a specific pattern, along with specific characteristics (such as FRL and BCCOS). For patterns with more than one preference, the preferences are arranged in order of desirability (first-choice, second-choice, etc.).

Typical AAR Pattern Arrangement

Figure 16-1 illustrates a typical AAR arrangement for private network calls between distant cities (in this example, Los Angeles, Phoenix, and San Diego).

The first-choice preference in a routing pattern is usually the most direct and economical route available. For a call from Los Angeles to Phoenix, this would be Trunk Group 41. If this route is busy, the switch has another chance to extend the call using an alternate route, Trunk Group 46 to San Diego and then to Phoenix via Trunk Group 47. If this route is also busy, there is yet another alternative, Trunk Group 22 to the public network.

The more alternate choices available in a pattern, the better the chance that a call will complete on the first calling attempt. During periods of peak demand, AAR lowers the probability that calls must be redialed because of busy trunk facilities.

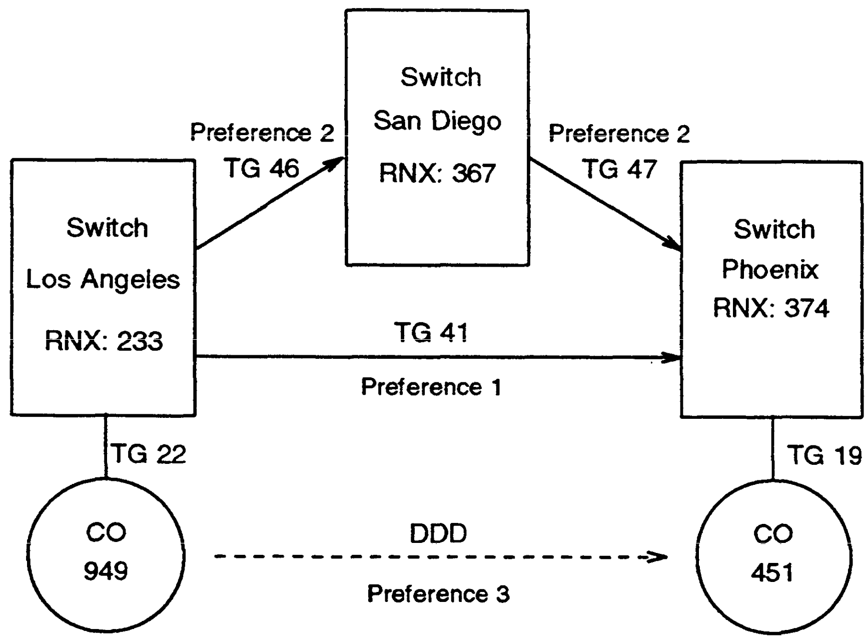


Figure 16-1. Automatic Alternate Routing

Facilities Restriction Level (FRL)

Each preference in a routing pattern has a Facilities Restriction Level (FRL). When a voice terminal user places a call, the switch selects a routing pattern based on the location code (usually the first two or three digits). The switch then compares the FRL associated with the call (the default FRL) to the FRL of the first-choice preference. A call can use a preference only if the FRL of the call is equal to or higher than the FRL of the preference. If the call can access the first-choice preference, the switch checks that preference for an available trunk. If every trunk is busy, the switch checks the next preference, again assuming that the FRL of the call is high enough to gain access to that preference.

If all accessible preferences are busy and a higher FRL would allow access to additional preferences, the switch can prompt the caller (with recall dial tone) for an authorization code. The Authorization Code feature (discussed later) provides a means of overriding the default FRL assigned to a specific voice terminal and replacing it with an FRL assigned on an individual (caller) basis.

Multiple FRLs

Each trunk group is assigned its own FRL; however, this FRL is not used by the AAR feature. Each AAR preference is also assigned its own FRL in Procedure 321, Word 1. When calls are routed using the AAR feature, the FRL assigned to the preference (not the FRL assigned to the trunk group) is used for call routing. The same trunk group can reside in more than one pattern and have a different FRL in each preference. In this way, the preference's FRL can more closely correspond to the cost of the call, as well as the cost of the trunk facility.

Figure 16-2 shows an example of such an arrangement. This example uses a 7-digit Uniform Numbering Plan.

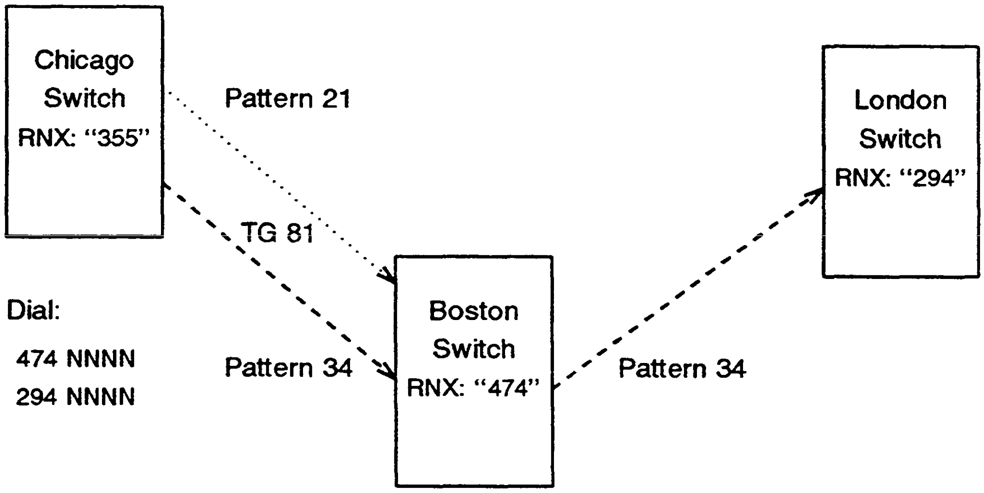


Figure 16-2. Multiple FRLs Assigned by Preference

In this example, the destination code of RNX "474" is a tandem switch in Boston, and the destination code of RNX "294" is a main switch in London. Trunk group 81 connects the switch in Chicago to the switch in Boston and appears in both Pattern 21 and Pattern 34 on the Chicago switch. As a preference in Pattern 21, it routes calls between the Chicago and Boston switches. As a preference in Pattern 34, it carries calls between Chicago and London that are routed by way of Boston. RNX "474" points to Pattern 21, and RNX "294" points to Pattern 34. Trunk group 81, serving in two patterns, has an FRL of "3" in Pattern 21 and an FRL of "6" in Pattern 34.

The caller in Chicago dialing "294 NNNN" needs a higher FRL to call London than to call Boston even though both calls route over the same trunk group.

Authorization Codes

If the dialed authorization code has an FRL that is higher than the default (previous) FRL, the switch replaces the default FRL with the ***authorization code's*** FRL. Using this new FRL, the switch makes another attempt to find an available trunk. If the new FRL is still too low, the switch tries to queue the call on the first-choice preference. The caller's FRL (either the default FRL or the authorization code FRL) must allow access to the first-choice preference before the call can be queued. The trunk group's queue length (number of calls allowed in queue) must also allow access to queuing. If queued, the switch checks the first-choice preference every 2 seconds for an idle trunk. If no trunk becomes idle before the time-in-queue limit is exceeded, the switch makes a "last try" on all accessible preferences (as determined by the FRL) in the pattern.

Pattern Queuing

The Pattern Queuing option allows any number of preferences in the pattern to be checked for an idle trunk during the entire queuing process. The switch may check the first-choice preference only (ignoring Pattern Queuing), or the first two preferences, etc. If Pattern Queuing indicates that three of ten preferences are to be checked, the switch still checks all ten during the "last try." When a call is placed in queue, it still queues on the first-choice preference and is restricted by that trunk group's queuing parameters (queue length, time-in-queue, etc.).

> **CAUTION:** *Care must be exercised when setting the number of preferences to be included in Pattern Queuing. Increasing the number of preferences to be checked means an increase in processing time. If this added processing time does not produce a significant increase in calls seined, queues could begin to overflow.*

FRL Raising

Just before the "last try," when the time-in-queue limit elapses for an AAR call, the switch can raise the call's FRL to help provide an allowable trunk facility for the call. FRL Raising is assigned on a per-system basis for both AAR and Automatic Route Selection (ARS) in Procedure 330, Word 1.

FRL Raising first compares the timed-out call's current FRL with the assigned Threshold FRL (Field 3). If the call's current FRL is greater than or equal to the Threshold FRL, this call is qualified for FRL Raising. At this time, the switch ***considers*** substituting the assigned Raised FRL (Field 4) for the timed-out call's current FRL. Substitution is made if the Raised FRL will be higher than the current FRL.

Uniform Numbering

Uniform numbering, to draw an analogy, is the mortar that binds a private network together. Each terminal within an Electronic Tandem Network (ETN) is assigned a 4-, 5-, 6-, or 7-digit destination code that uniquely identifies that terminal within the network.

Every switch within the ETN network must conform to the selected numbering plan. For example, when a 7-digit numbering plan is used at one switch, a 7-digit numbering plan must be used at every switch within the ETN. The AAR feature does not route calls to private-network destination codes of mixed lengths.

The number of terminals within a network determines how many digits are needed. In small networks, a destination code may only need four digits, while large networks may require as many as seven digits. Each destination code has a *2*- or 3-digit location code, plus an extension number of from 2 to 5 digits.

> **NOTE:** System 85s, DEFINITY Generic 2 switches, and DIMENSION® System Feature Package (FP) 8, Issue 3 switches cannot have 2-digit extension numbers. These switches must have at least 3-digit extension numbers.

Location Code

The location code has the form RNX, where R equals any digit 2 through 9 except the Call Detail Recording (CDR) Account Code Prefix and the Reserved Digit (Procedure 285, Word 1, Fields 6 and 7), and X equals any digit 0 through 9. To reach a terminal assigned to a distant switch in the private network, the caller dials the AAR access code, plus the location code, plus the extension number. To reach another terminal on the same switch, a caller only has to dial the extension number.

When a Direct Distance Dialing (DDD) or International DDD number follows the AAR access code, the call is passed to the ARS feature for routing.

Overflow to the Public Network

Typically, the first-choice preference in an AAR pattern is a direct (private-network) path to the destination switch. Alternate routes, however, can require the use of public-network facilities. At the customer's option, an AAR routing pattern may contain public network trunk groups (for example, Foreign Exchange, Wide Area Telecommunications Service, or local Central Office).

Normally, public-network trunk groups are either the last choice or the only choice in an AAR routing pattern. The switch routes an AAR call via a public-network trunk group only if every private-network trunk group in the routing pattern is either busy or the routing pattern has no private-network trunk groups. When a call is rerouted from a private-network pattern to a public-network pattern (or vice versa), the switch can be administered to apply *warning tone* to the connection. Also, the destination code is changed to conform to the uniform numbering plan of the new network. The method used to accomplish this is called subnetwork trunking.

Subnetwork Trunking

The switch uses software routines to modify the format of the dialed number. These software routines can insert digits, delete digits, regroup digits, or insert pauses between digits as necessary to change the destination code to conform to a new uniform numbering plan. The destination code modification capabilities of subnetwork trunking available beginning with the System 85, Release 2, Version 3 switch areas follows:

- Up to seven digits can be deleted from the front of a destination code.
- Up to 20 digits can be inserted in place of the deleted digits.
- Digits can be sent in up to four groups.
- Each group can have up to 15 digits.

NOTE: The switch does not perform AAR subnetwork trunking on an outgoing trunk group if the trunk group's "Main/Tandem" field is assigned to "1" in Procedure 103, Field 4.

Both public and private-network trunk groups can be accessed through subnetwork trunking routines. Transparent access to any long-distance vendor, called Interexchange Carrier (IXC) Access, is also available. Trunk groups that involve a toll charge can be set up to give a 1-second warning tone. After hearing the tone, the caller can either let the call proceed or hang up and try again later when a less expensive trunk facility is available.

NOTE: When subnetwork trunking is used as an overflow to a private-network switch ***(with*** DID facilities) over the public network, the inserted digits in Procedure 321, Word 3 should specify a complete (at least 7 digits) Listed Directory Number (LDN) at the receiving private-network switch. As the subnetwork trunking software encounters this LDN, the software can modify the digit stream in one of two ways:

- If the called number corresponds to an extension at the receiving switch, then only part of the digit stream is modified. In this case, the subnetwork trunking software deletes the number of leading digits specified in Field 7 of Procedure 321, Word 1 and then inserts the number of digits specified in Field 4 of Procedure 321, Word 2.
- However, if the called number corresponds to an attendant at the receiving switch (for example, the last four digits are "0111" or "0XXX"), then the entire digit stream is modified. In this case, subnetwork trunking ignores the number of leading digits to delete (Procedure 321 Word 1 Field 7). Instead, the switch deletes the entire digit stream (RN(X) + 0XX(X)) and then inserts the entire LDN that was previously specified in Procedure 321 Word 3.

NOTE: Also, when subnetwork trunking is used as an overflow to a private-network switch ***(with*** DID facilities) over the public network, Field 9 of Procedure 321, Word 1 can be assigned in one of two ways according to extension numbering at the receiving switch.

Field 9 is set to "0" when the digit "0" ***is not used*** as the first digit for extension numbers at the receiving switch. When Field 9 is "0," the subnetwork trunking software converts the last four digits to the LDN whenever the thousandths digit is "0."

Field 9 is set to "1" when the digit "0" ***is used*** as the first digit for extension numbers at the receiving switch. When Field 9 is "1" the subnetwork trunking software converts the last four digits to the specified LDN ***only when*** the last four digits are "0111. "

NOTE: When subnetwork trunking is used as an overflow to a private-network switch ***(without*** DID facilities) over the public network, Procedure 321, Word 1, Field 7 should be assigned to delete the entire digit stream (RN(X)-XXX(X)), and then a non-DID LDN should be inserted (with at least 7 digits). In turn, the attendant(s) at the receiving switch can extend these overflowed calls to the originally called party.

NOTE: Also, when overflowing to private-network switches without DID facilities, a *pseudo-DID) capability can* be provided by deleting the RN(X) and then inserting a 7- or 10-digit Remote Access number associated with the distant switch.

If the distant switch uses a barrier code or authorization codes for Remote Access security, then subnetwork trunking must also be assigned to send a valid 4-digit barrier code or a 4- to 7-digit authorization code between the Remote Access number and the originally dialed extension. Also, whenever the pseudo-DID capability is used, insert a pause just before the trailing extension.

For pseudo-DID, after subnetwork trunking acts on the original RN(X) + XXX(X), the final digit stream should have the form:

7- or 10-Digit Remote Access Number
+ [Barrier or Auth Code] + Pause + Dialed XXXX.

Routing to Private Network Locations in Foreign Countries

Another convenience AAR offers is private-network routing to locations in foreign counties. The caller dials a private-network number of the form RN(X)-XXXX, where RN(X) is the location code for the destination (private network) switch and XXXX is the final four digits of the International Direct Distance Dialing (IDDD) number. Subnetwork trunking converts the RN(X) to the appropriate country and city codes for the overseas central office that serves the private-network location. For System 85 switches prior to Release 2, Version 4, Issue 2.0, an IDDD route is always the only choice in this type of routing pattern.

For System 85, Release 2, Version 4, Issue 2.0 and DEFINITY Generic 2.1, Issue 2.0 switches, the switch can examine the digits following the IDDD designator (01X) to make routing decisions. With this enhancement, users can dial numbers in the form "0011+XXXXXXXXXXXXXXX" and have the switch choose either *private facilities* or *least cost (public) services* when they are available. With proper administration, users can dial public (01X) numbers and the switch can capture on-net traffic for private facilities or off-net traffic that is closer to an international destination for tail-end hop-off. (See the ARS feature for a discussion of tail-end hop-off.)

To use selective international call routing customers assign selected digit strings to a "01X exception list." These digit strings point immediately to an ARS routing designator (pattern) containing a list of preferences that can include either private or public facilities. This allows customers to administer selected international dialing sequences to point to a pattern with private facilities first, while other international dialing sequences (where private tie trunks may not be available) point to the default international routing pattern defined in Procedure 311. This administration uses the same digit translation tables as 10- to 7-digit conversion. These tables can contain "01X" entries. The digit strings can be from 7 (minimum) to 18 digits in length and must begin with "01." The corresponding routing designator number (1 through 64) represents an existing ARS routing patterns already defined in Procedure 309.

NOTE: Administration for selective international routing is accomplished through the ARS feature; however, private network facilities can be used for call routing, and either the AAR or ARS access code can be used to access these routing patterns. New administration is done using Procedure 312, Word 3, which is available on the Maintenance and Administration Panel (MAAP) or the DEFINITY, Manager II.

Generalized Route Selection

Conditional Routing is the first implementation of Generalized Route Selection and is available only on System 85, Release 2, Versions 3 and 4 and DEFINITY Generic 2 switches. Generalized Route Selection uses a set of call attributes for routing selection rather than just the RN(X) dialed by the user. Each different set of attributes defines a different call category.

Conditional Routing

Conditional Routing, also known as Satellite Hop Control, is the only attribute currently available. Conditional routing counts the number of satellite links a call has used to reach the local switch (hence the name satellite hop control). It ranges from 0 to 2. The three possible call categories are:

- Call Category 0—Satellite count = 0
- Call Category 1—Satellite count = 1
- Call Category 2—Satellite count = 2.

If two satellite links have already been used, the use of additional satellite links may introduce signaling delays and signal degradation. Therefore, AAR must select a pattern that will not use more than two satellite links.

The satellite count associated with the call is sent as a second Traveling Class Mark (TCM) following the Facilities Restriction Level TCM when used. The local switch may receive a destination code followed by the callers FRL TCM and then by the Conditional Routing TCM. The Conditional Routing TCM is matched against a call category that in turn is used to select a pattern. Patterns not assigned to the call category cannot be used for routing for example, a call with a satellite count of 2 can be prevented from accessing a pattern containing any satellite links by assigning the pattern to Call Category 0 and/or Call Category 1. It is not necessary at this time to convert from satellite count to call category. However, as new attributes are added in future releases, the need for distinct call categories will develop.

Bearer Capability

Bearer capability applies to all calls and call support facilities but is of primary significance to data calling requirements. Bearer capability was first used on the System 85 with the Release 2, Version 4 ISDN—PRI feature. With the introduction of the ISDN—BRI feature in ***DEFINITY Generic 2,*** the scope and function of bearer capability has been expanded

In ***System 85, Release 2, Version 4,*** bearer capability is assigned to ISDN facilities as a part of the class of service and is used with AAR to select routes for ISDN calls. Five bearer capability codes are used to identify the types of calls that a specific facility can support.

Bearer Capability Code	Type of Traffic Supported
0	Voice and Voice Grade Data
1	Mode 1 data, 56 Kbps allowed

2	Mode 2 data, 64 Kbps allowed
3	Mode 3 data
4	Mode 0 data.

In ***DEFINITY Generic 2,*** the power and versatility of ISDN messaging is used to determine bearer capability needs for call routing by the AAR feature. Identification of call type and resource requirements are based on the best available information, obtained as follows:

- ***Call Setup Messages***

  Call control information contained in the ISDN call setup message associated with each specific call is the primary source of information on protocol and call handling facility requirements. Call control setup messages (originating from ISDN facilities) contain IEs that indicate the type of call (for example, voice, Mode 0 data), protocol used, data rate, and other information needed to identify required resources.

- ***Optional Query***

  Data modules (both ISDN and DCP) have the ability to respond to requests for additional information from the switch. For information that is needed but not available in a specific call setup message this optional query ability is used.

- ***Default Values***

  The last resort for determining resources needed for a specific call is the customer administered ***BCCOS*** (Bearer Capability Class of Service) assigned to trunk groups, extensions, and AAR/ARS preferences. This BCCOS provides default requirements and characteristics for specific ISDN facilities. The default BCCOSs are associated with the facility (trunk, extension, etc.) and not a specific call.

If call processing must, for any reason, use only the BCCOS default values (unchanged), the resulting DEFINITY Generic 2 call handling will be consistent with call handling provided by a System 85, R2 V4 switch for the same call. In effect, the default processing for DEFINITY Generic 2 equates to the basic bearer capability handling provided by the System 85, R2 V4 switch.

Switch actions based on BCCOS are specified in administration (Procedure 014, Word 1, Fields 4 through 13). These switch actions determine how a call with a specific BCCOS will be handled by each preference in each AAR/ARS pattern. Three specific switch actions are used:

- Circuit switch the call
- Insert a Modem Pooling conversion resource
- Block the call.

With Bearer Capability, the AAR search algorithm operates essentially as follows:

1. ***Preferred Option***

   First the search looks for a preference that calls for the switch action ***"circuit switch the call."*** If a preference is found that provides this action, the FRL allows that preference to be used, and a trunk is available, then that preference is selected for routing the call.

2. ***Acceptable Option***

   While looking for a preference that calls for circuit switching the call, the search algorithm also checks for a preference that calls for ***insert a Modem Pooling conversion resource.*** If a preference is encountered that calls for the action ***insert a Modem Pooling conversion resource,*** that preference is recorded for future reference if needed.

3. ***Exercising the Alternative***

   If the search for the ***preferred option*** is not successful (no available trunk with a usable FRL is found), the algorithm tries to connect the call to an ***acceptable option*** trunk if one is available.

4. ***Unacceptable Option***

   Blocking the call is an unacceptable option. All other alternatives for routing the call (Authorization Code, FRL raising Queuing, etc.) must first be exhausted. No attempt will be made to connect a call to a trunk when the switch action specified is ***block the call.***

A more detailed description is provided under the Bearer Capability feature.

Flow Diagram

To help conceptualize this complex feature, a flow diagram is provided in Figure 16-3. This diagram does not show all of the decisions made in the AAR software. The diagram does, however, unify many of the different AAR functions.

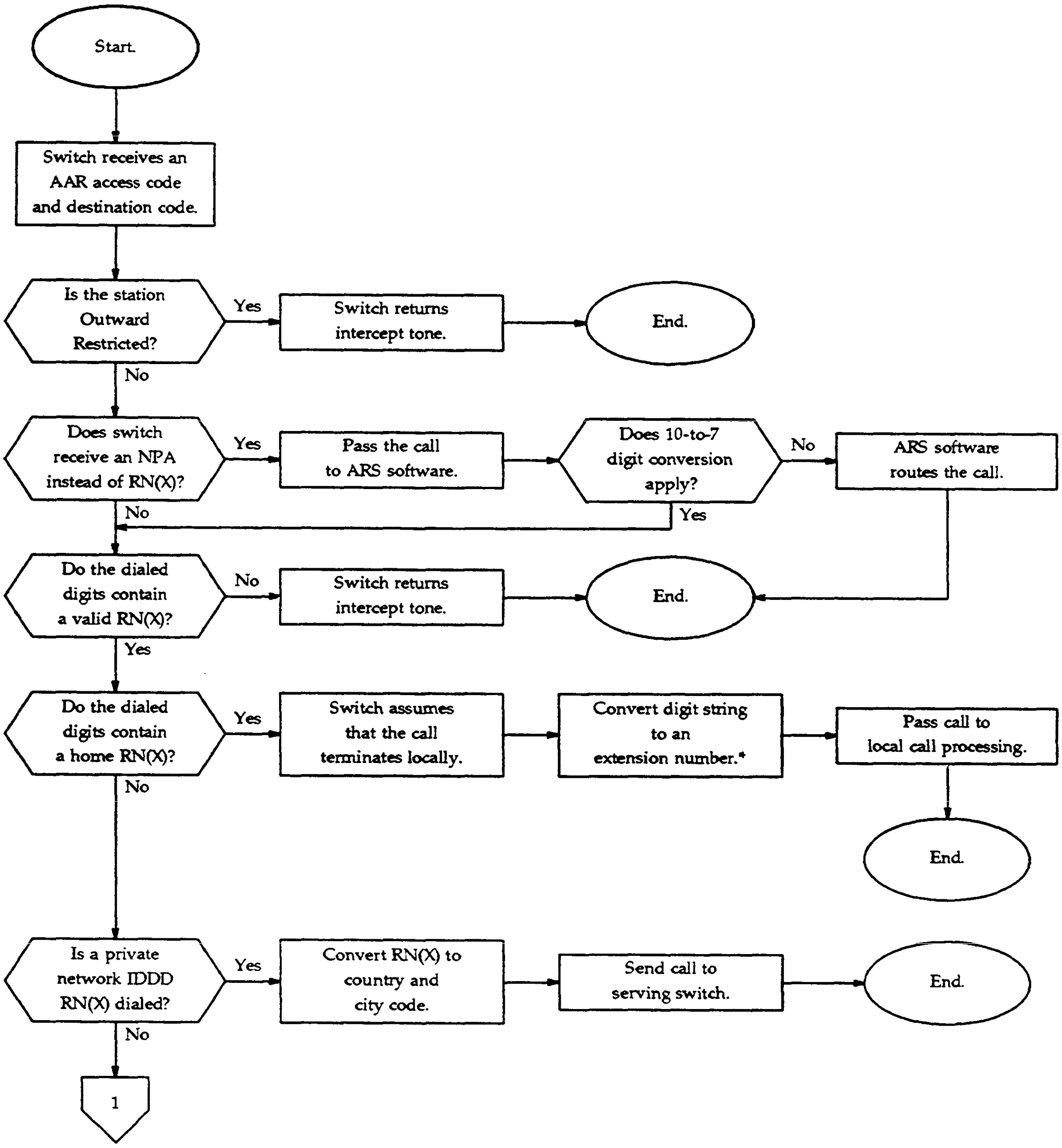


Figure 16-3. AAR Feature Flow

* Subsequent processing of this extension number could result in subsequent routing by the AAR feature or other features (for example, Main/Satellite or Extension Number Portability).

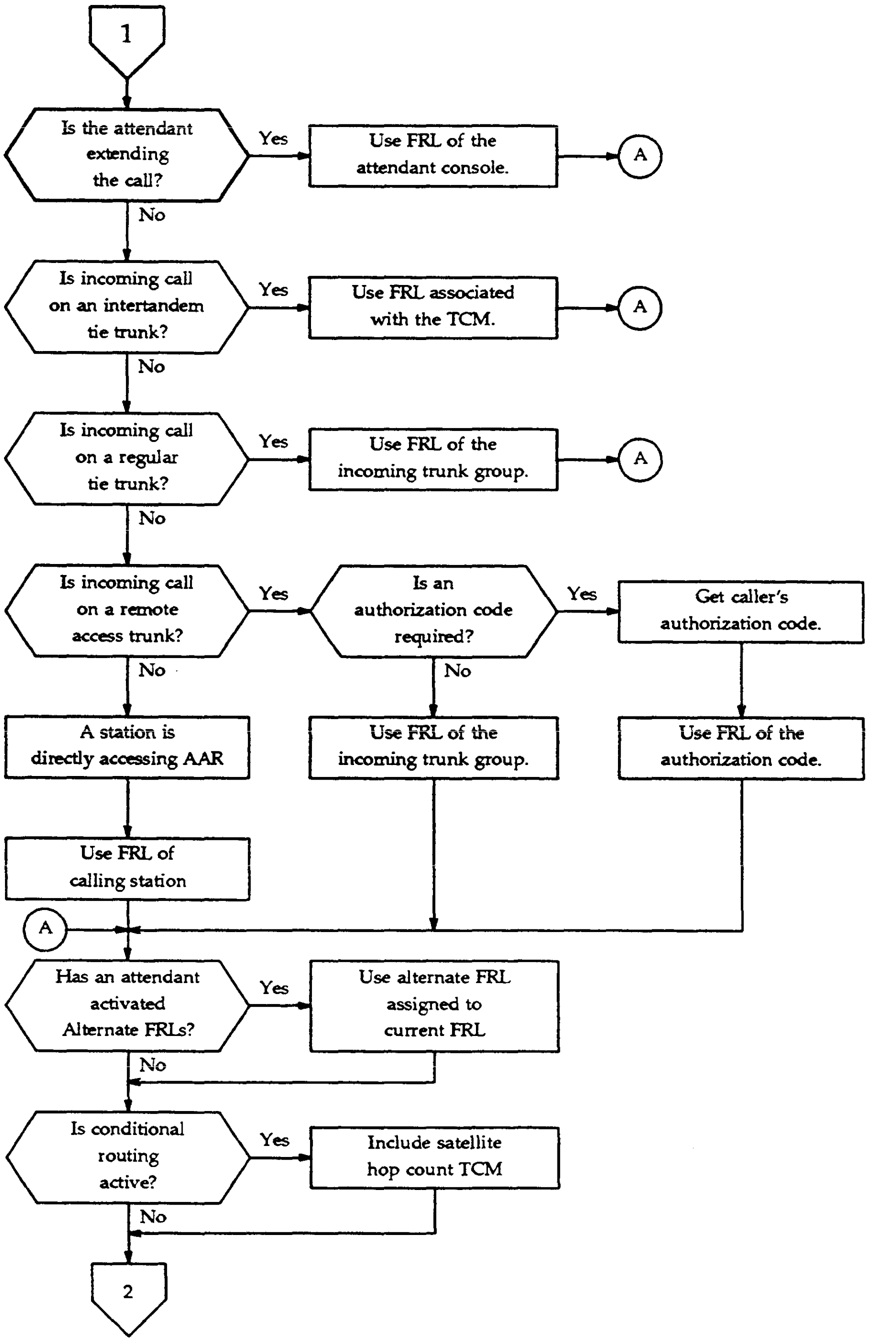


Figure 16-3. AAR Feature Flow (Sheet 2 of 5)

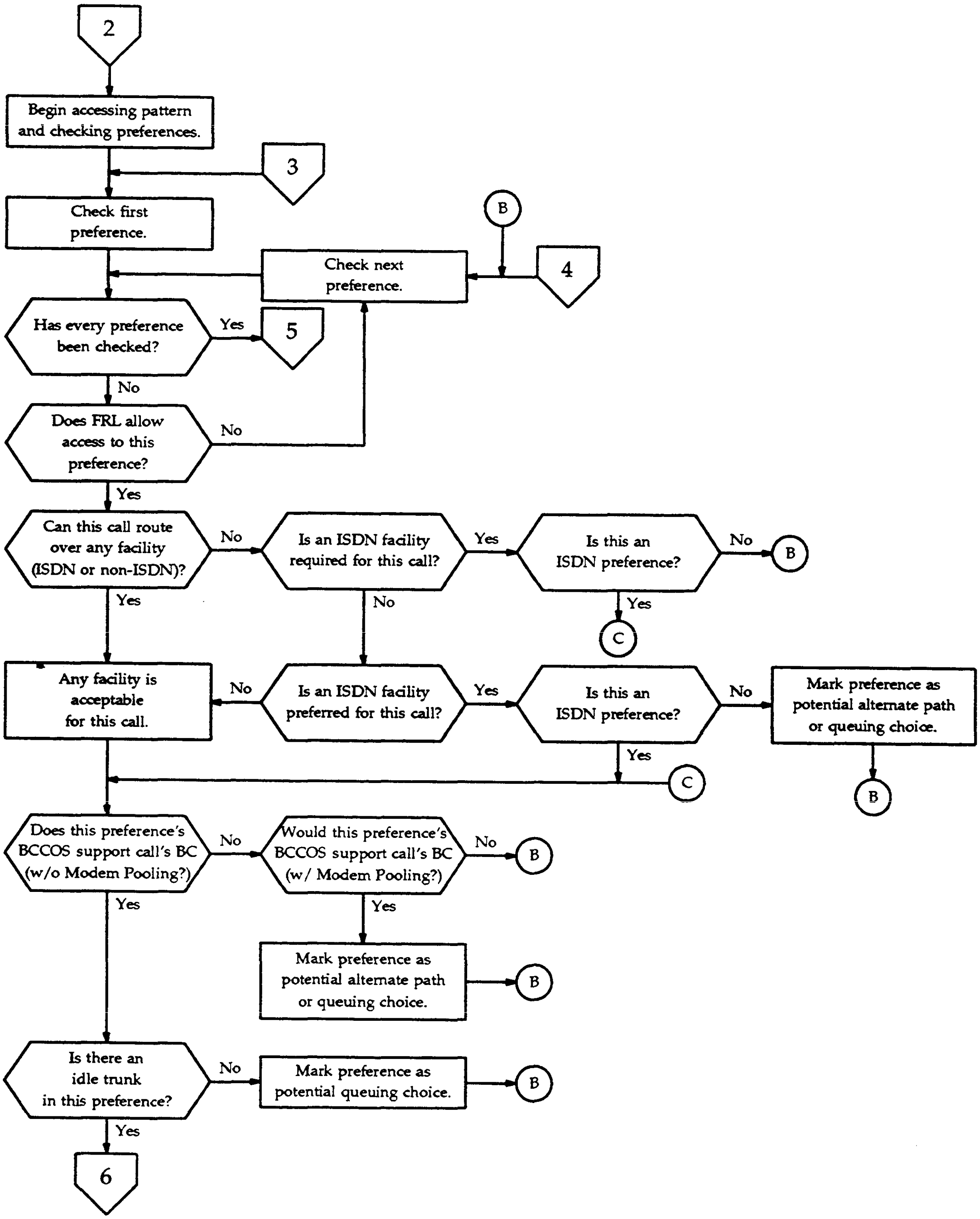
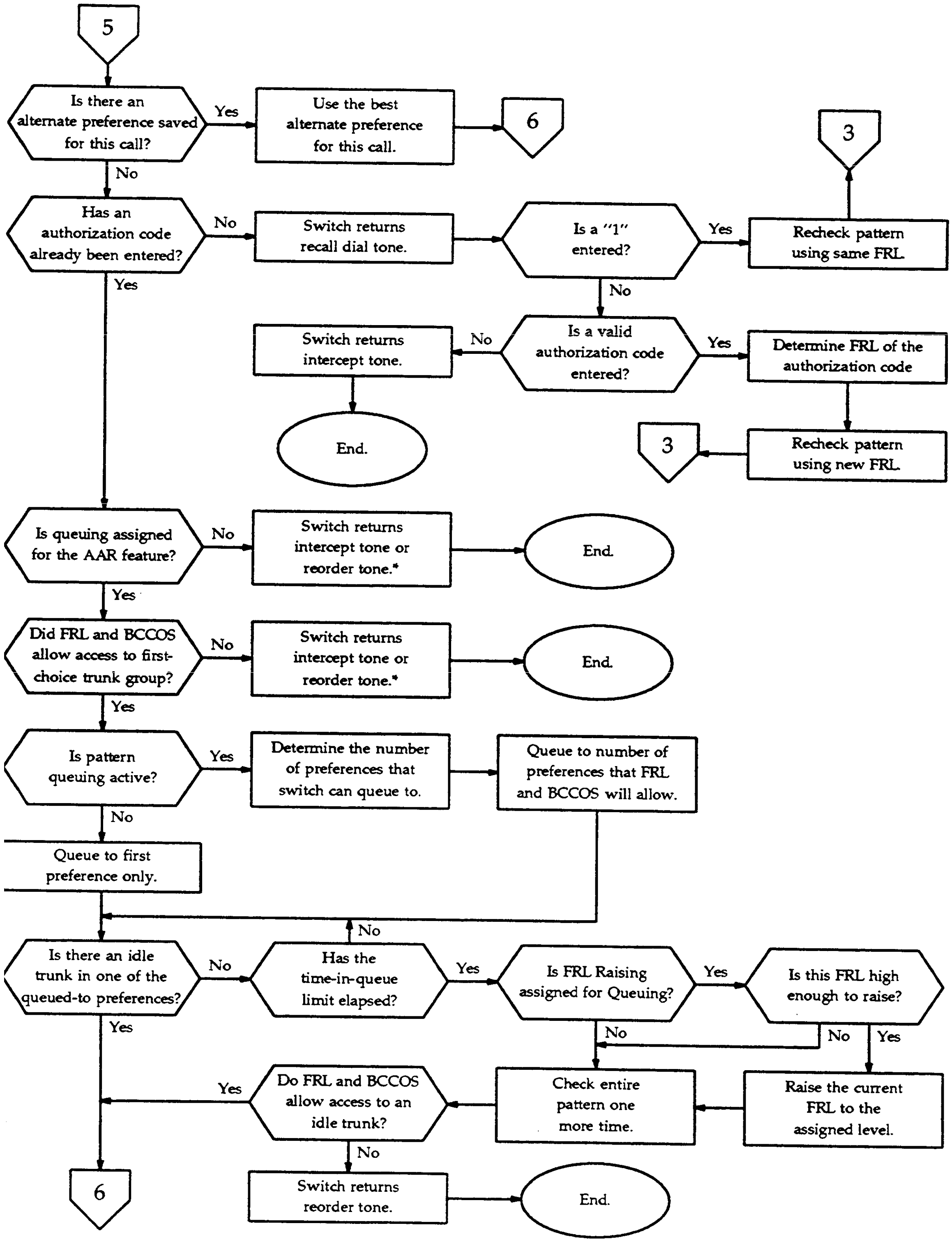


Figure 16-3. AAR Feature Flow (Sheet 3 of 5)



* Intercept tone if the call is not allowed access to any preferences. Reorder tone if the accessible preferences are busy.

Figure 16-3. AAR Feature Flow (Sheet 4 of 5)

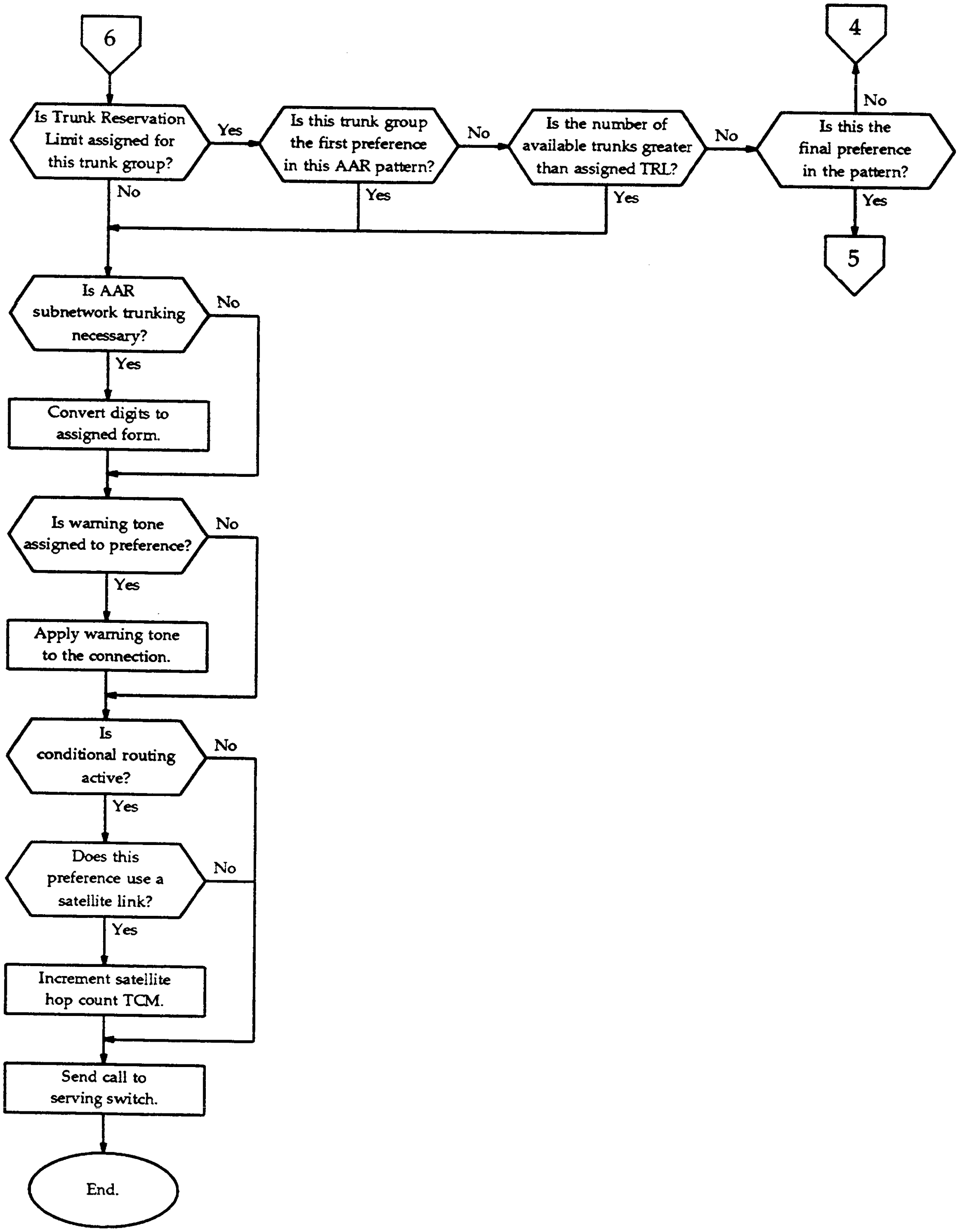


Figure 16-3. AAR Feature Flow (Sheet 5 of 5)

Considerations

"Standard Network" Field in Procedure 276

Field 1, the Standard Network field, in Procedure 276 ***must*** be assigned to provide the AAR feature. In addition to AAR, this field is assigned to provide a private-network Uniform Numbering Plan for tandem switches in an electronic tandem network (ETN).

"Multipremise" Field in Procedure 276

Field 2, the Multipremise field in Procedure 276, is assigned to enable the Main/Satellite feature. The Main/Satellite software is a prerequisite for the ETN capability (including AAR and Uniform Numbering Plan) on tandem switches.

Single-Digit AAR Access Code

For System 85 and DEFINITY Generic 2, the AAR dial access code must be a single digit between "0" and "9." This dial access code, which is usually assigned as "8," cannot be assigned as "*" or "#." This limitation exists because the AAR Prefix Digit in Field 9 of Procedure 103 is limited to a single digit.

Inferred Routing

Certain tie trunks can be administered (Procedure 103) to always route incoming calls to AAR or ARS. These calls are said to "infer" AAR or ARS routing, meaning that the switch automatically inserts the private-network access digit ahead of the received digits and then routes the call. The trunk types that commonly infer AAR or ARS routing are as follows:

- 2-way intertandem tie trunk: wink start in/delay dial or wink start out (Trunk Type 41*)
- 1-way incoming intertandem tie trunk wink start (Trunk Type 42*)
- 1-way outgoing intertandem tie trunk or bypass access tie trunk: delay dial or wink start (Trunk Type 43)
- 2-way access tie trunk (main-tandem connection): dial repeating in/ delay dial or wink start out (Trunk Type 46*)
- 2-way access tie trunk dial repeating delay dial in/ delay dial or wink start out (Trunk Type 47*).

FRLs and Intertandem Tie Trunks

The preceding flowchart shows that receiving System 85s and DEFINITY Generic 2 tandem switches always use the FRL associated with the Traveling Class Mark (TCM) for

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***** Inferred routing must be used with this trunk type.

incoming calls over intertandem tie trunks to select an outgoing AAR preference. However, this flowchart presents a simplified picture of the preference-selection process at the receiving tandem. The following paragraphs are a more detailed description of this process.

The FRL TCM is the last (or the next to last) digit that the receiving tandem gets, and the System 85 or DEFINITY Generic 2 AAR software does not wait for this digit to begin its preference selection process. During the *first* check of the preferences in the assigned pattern, the receiving tandem actually uses the FRL assigned to the incoming intertandem trunk group in Procedure 103, Field 2. (If this FRL is not assigned, the FRL value for the trunk group defaults to "0.")

If the AAR software ***can select*** a preference using the trunk group's FRL, it does. However, when the digits are sent over the next outgoing intertandem trunk group, the System 85 or DEFINITY Generic 2 uses the FRL TCM that was subsequently received.

If the AAR software ***cannot select*** a preference using the trunk group's FRL, the AAR software needs to check the pattern's preferences again. By this time, the TCM FRL has arrived, and the System 85 or DEFINITY Generic 2 unconditionally replaces the trunk-group FRL with the TCM FRL. From this point on*, the AAR feature uses the TCM FRL to select a preference and sends the TCM FRL with the outgoing digits.

Authorization Codes and Intertandem Tie-Trunk Calls

For calls where an authorization code has not already been entered, the preceding flowcharts show that a receiving tandem switch (System 85 or DEFINITY Generic 2) always prompts for an authorization code (when assigned) if the default FRL cannot access an available facility and a higher FRL would allow access to additional preferences. However, the flowcharts present a simplified picture of AAR authorization-code prompting.

For incoming calls received over intertandem tie trunks (Procedure 103, Fields 3 and 4 assigned as "1") where the AAR feature tries to select an outgoing preference, an ETN tandem switch does not prompt for an authorization code. First, the calling party may have already entered an authorization code at the previous tandem. Also, the list of valid authorization codes could be different at each tandem within the network, and the calling party would have no way of knowing which tandem is requesting the code.

Therefore, when the FRL for these calls is too low to access an available trunk facility, the ETN tandem does one of the following:

- Queues the call to busy accessible preferences in the preference depth (when AAR Queuing is assigned)

* Until the time-in-queue limit elapses, and FRL Raising can be invoked.

- Returns reorder tone if the accessible preferences are busy (when AAR Queuing is not assigned)
- Returns intercept tone if the FRL is too low to access any preference or the first queuing preference.

Symmetrical Routing Depth and Intertandem Tie Trunks

For calls where the next preference in a pattern needs to be checked, the preceding flowchart (in Positions "4" and "B") shows that a System 85 or DEFINITY Generic 2 switch always determines whether every preference has been checked. However, this flowchart presents a simplified picture of the AAR preference-selection process.

For incoming calls from over intertandem tie trunks (Procedure 103, Fields 3 and 4 assigned as "1") where the AAR feature is selecting outgoing preference, an ETN tandem switch checks the Symmetrical Routing Depth (Procedure 285, Word 1, Field 5). If the Symmetrical Routing Depth is set to "0," the switch will always ask whether every preference has been checked (as shown in the flowchart). However, if the Symmetrical Routing Depth is set to a value between "1" and "9," the switch will compare the current preference number to the assigned value. If the current preference number is ***less than or equal to*** the assigned value, the switch will check this preference. If the current preference number is ***greater than*** the assigned value, the switch will act as though every preference had been checked and continue AAR processing at Position 5.

This Symmetrical Routing Depth parameter provides a simple method for controlling circular routing through ETN tandem switches.

Data-Only Routing Patterns

For data-only applications, specifically where data calls are originated over DS1 alternate voice data (AVD) trunks, AAR data call routing patterns must be separate from patterns used for voice calls. Routing patterns for data calls should minimize tandem connections. All trunk groups in a data-only pattern must be suitable for data transmission. DS1 AVD trunk groups must not overflow to voice-grade trunk groups because the introduction of modems (through the Modem Pooling feature) at the destination end will result in handshake failure.

Hard and Soft Processor Swaps

The contents of the AAR routing patterns are stored in a translation portion of switch memory. Therefore, these patterns will endure a hard processor swap.

AAR queues are stored in a status portion of memory. If an AAR call is queued to a pattern when a hard swap occurs, the call is not routed. Instead the queue is cleared.

Stable AAR calls will endure a hard processor swap. However, an AAR call cannot be placed during a hard swap.

The AAR feature operates normally during a soft processor swap.

Interactions with Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Control of Trunk Group Access (ACTGA)

The ACTGA feature takes precedence over AAR. A call directed by AAR to a controlled trunk group is routed to an attendant.

Authorization Codes

The Authorization Codes feature is used to provide personal calling privileges (transportable from one station to another) that can be used instead of the default FRL when placing an AAR call. An authorization code can be used when placing any outgoing station originated call. An authorization code can also be used when placing a Remote Access call unless on-network access to off-network callers is denied (the network access flag is set to "0" for the specific authorization code).

Automatic Route Selection (ARS)

If a System 85 or DEFINITY Generic 2 caller dials the AAR dial access cock and then dials a public-network telephone number of the form "NPA-NXX-XXXX," the switch does not automatically deny the call. Instead, the AAR software passes the 10-digit public network number to the ARS software for routing.

Dining this process, the caller does not specify either toll or nontoll access to the public-network facilities (by dialing the toll or nontoll ARS access code). In this situation, the ARS software selects a preference for the call as if the calling party had dialed the ***toll*** ARS access code.

When selective international call routing (or 01X screening) is in effect, it applies to all calls using the "01X" prefix, whether the AAR or ARS access code is used. The 01X exception list is administered only once, and cannot be administered separately (different lists) for the separate features (AAR and ARS).

Bearer Capability

In System 85, Release 2, Version 4, bearer capability defines the type of calls that a trunk group or service facility (such as modem pool or host access port) can support. The AAR feature searches for a preference that supports the bearer capability requirement of the call being routed. Five bearer capabilities are recognized:

Bearer Code	Type of Traffic That Can Be Carried
0	Voice and voice-grade data allowed
1	Mode 1 data, 56 Kbps allowed
2	Mode 2 data, 64 Kbps allowed
3	Mode 3 data
4	Mode 0 data.

If a request is made for a trunk group to route a Mode 2 call, only preferences that can support Mode 2 data are searched.

Bearer Capability Class of Service (BCCOS)

In DEFINITY Generic 2, AAR routing is based on BCCOS. Two methods are available for routing the call:

1. The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (for example, Mode 2 data, 1200 bps, restricted channel, etc.). If a match is found and a trunk is available, other factors such as FRL permitting, the action taken is to circuit-switch the call.
2. If a match is not found, the algorithm attempts to connect the call to a preference for which the action to take is **not** ***block the call.*** With currently available options, this would be a preference where the action is to insert a modem pool.

Call Detail Recording

When optional account codes are used, a private-network caller can dial an account code after dialing the AAR access code. When this operation is used, the System 85 or DEFINITY Generic 2 needs a way to discriminate between an account code and a private-network telephone number. Therefore, the Account Code Prefix (assigned in Procedure 285, Word 1, Field 6) ***cannot match*** the first digit of any of the RN(X)s (location codes) within the private network.

When Forced Entry of Account Codes (FEAC) is used, there is no need to discriminate between account codes and RN(X)s. Using this operation, the private-network caller (after dialing the Account Code access code) dials the account code before dialing the AAR access code.

Call Vectoring

In Procedure 010, Word 3, an FRL can be assigned to vector directory numbers (VDNs) for use with the "route to" command. The FRL of a VDN is used to determine whether the call is allowed to route over available private-network facilities.

Data Call Setup

The AAR feature can be used to improve the routing of data calls just as it is for voice calls. However, without ISDN, separate routing patterns are required to prevent voice calls from terminating on data extensions and vice versa.

Distributed Communications System (DCS)

A DCS subnetwork can be administered as either a Main / Satellite arrangement (4-digit dialing plan) or as an Electronic Tandem Network (ETN). If a DCS is administered as an ETN, the AAR feature is used for internode call routing.

However, DCS feature transparencies ***are not*** provided when a user (within the DCS subnetwork) dials the AAR dial access code + RNX + XXXX to call a party in another DCS node. DCS feature transparencies are ***only*** provided when the calling party dials an extension number [including "soft" Call Coverage extension numbers, DID LDNs, associated extension numbers, and VDNs (Vector Directory Numbers)] or an attendant dial access code.

Extension Number Portability (ENP)

The AAR feature must be used to route calls inside a portability subnetwork. Instead of the usual AAR assignment of an RN(X) to a pattern, the ENP feature associates a pattern with a node number. Extension Number Portability may require network routing which will result in additional call-setup time (over local calls).

Information Systems Network (ISN) Interface

ISN data stations can use the circuit-switch feature AAR when placing private-network calls through the circuit switch. When this is done, Modem Pooling is required if the trunk groups to be accessed contain analog trunks.

ISDN—PRI (Primary Rate Interface)

Calls placed over JSDN facilities use the AAR or ARS features for ISDN access. To work effectively for ISDN, bearer capability classes must be assigned. In this way, placing an ISDN call is transparent to the user. AAR and ARS patterns are selected according to the calling party's class of service (COS) and the trunk-group bearer capability. Class of Service:

The AAR and ARS pattern searches work differently for ISDN calls based on the calling station's COS. For ISDN, the calling party's COS specifies one of three options as follows:

ISDN Facilities Required

If ISDN facilities are required by a user's class of service, only ISDN end-to-end facilities should be used for AAR or ARS routing. If ISDN end-to-end connections are not available, the call will either queue or receive reorder tone.

ISDN Facilities Preferred

If ISDN facilities are preferred but not required by the COS, ISDN facilities are checked first, followed by a check for non-ISDN facilities.

Any Facilities Available

Any available facilities can be used to complete the call. In this case AAR and ARS pattern searches work like they do for a non-ISDN call.

Interexchange Carrier Access (IXC)

The subnetwork trunking function of the AAR feature is used to implement the IXC selection automatically. An IXC code (10XXX) cannot be dialed by a station user, and the IXC digits cannot be passed over the ETN network.

Last Number Dialed (LND)

When the LND feature is used to redial an AAR call, the LND feature sends all of the originally dialed digits (including the AAR access code) to the AAR feature for routing. If an AAR function such as AAR subnetwork trunking or Interexchange Carrier (IXC) Access needs to modify the digit stream, the AAR feature performs the necessary modifications (internally) each time the number is redialed.

For switch security, the LND feature does not store or redial Authorization Codes.

Look-Ahead Interflow

The AAR feature is required to route Look-Ahead Interflow calls through the private network. In order to provide the AAR feature for a switch within the ETN configuration, the "Standard Networking" field (in Procedure 276) must be assigned.

When the AAR feature is used to route Look-Ahead Interflow calls, the digit contents of a vector-group list item for a "route to" step must conform to the uniform numbering plan (UNP) within the ETN network. When System 85s, DEFINITY Generic 2 switches, and DIMENSION FP 8, Issue 3 switches reside in the ETN, this UNP format for the ETN can have one of four forms:

1. RNX (3-Digit Location Code) + XXXX (4-Digit Extension Number)
2. RN (2-Digit Location Code) + XXXX (4-Digit Extension Number)
3. RNX (3-Digit Location Code) + XXX (3-Digit Extension Number)
4. RN (2-Digit Location Code) + XXX (3-Digit Extension Number).

Besides conforming to the UNP for the ETN, the vector-group list items for Look-Ahead Interflow "route to" steps must be prefixed by the single-digit AAR dial access code.

Besides conforming to the dialing plan for the private network, an AAR pattern must be translated for the RN(X) specified within the destination digits of a "route to" step. When this is not done, the "route to" step is treated as having an invalid destination. If the "route to" step is the final effective step in the sending (or tandeming) vector, vector processing treats the step as a "stop" step. Otherwise, vector processing continues with the next sequential step in the vector.

At a sending (or tandeming) switch, Queuing (including Pattern Queuing and FRL Raising, which is invoked by Queuing) does not apply to Look-Ahead Interflow calls. Instead, if every preference in the first-choice AAR preference is busy, Look-Ahead Interflow software will either retry routing the call at 2-second intervals (if final effective step) or continue vector processing with the next sequential vector step.

For voice calls, the Bearer Capability Class of Service (BCCOS) is not a significant consideration. This is because voice calls are usually compatible with any carrier facility. However, the AAR feature does check the BCCOS of calls that are diverted outside the switch by a "route to" step. Therefore, when applicable, the BCC of outgoing (first-choice) preference must be compatible with the BCC in the local calling party's COS or the BCC assigned to the incoming trunk group.

Since a Look-Ahead Interflow "route to" command always selects the first preference of the AAR pattern needed to route the calls, the AAR Trunk Reservation Limit (assigned in Procedure 103, Word 1) does not prevent interflow calls from accessing the preference. Rather, assigning an AAR Trunk Reservation Limit to the trunk group has the effect of reserving trunks in the preference to ensure the muting of Look-Ahead Interflow calls.

The Look-Ahead Interflow feature is compatible with AAR Conditional Routing when assigned. For Look-Ahead Interflow calls, the AAR software will increment the satellite hop count whenever a "route to" step diverts a call over an AAR preference with a satellite link. Also, the AAR software will send the current value of the satellite hop count as the second TCM for Look-Ahead Interflow calls.

The Look-Ahead Interflow feature is compatible with AAR subnetwork trunking. For Look-Ahead Interflow calls that are not routed over intertandem tie trunks, the AAR subnetwork trunking function can internally modify the acceptable digit formats for vector-group list items so that the next ETN switch receives the expected digits.

As part of the Look-Ahead Interflow SETUP message, an intervening private-network switch is always requested to route the interflow call on ISDN-Preferred basis. Then, acceding to its AAR software, the intervening switch gives ISDN routes first preference during its route-selection process. If the private-network intervening switch cannot find an available ISDN route, the intervening switch will return a "Private-Network Interworking" message to the sending switch. For private-network calls, the sending switch will accept this message and allow the call to route on a non-Look-Ahead basis.

Main/Satellite/Tributary

The AAR feature can be assigned to a main location producing what is known as an "intelligent main." This is done when the Main / Satellite / Tributary configuration accesses a tandem switch. Assigning AAR to a main location gives the main the private-network muting capabilities that apply to the main itself. Some of these capabilities include:

- Direct Distance Dialing (DDD) overflow for outgoing calls when the tie trunks between the main and the tandem are heavily used
- Authorization-code screening by the local main for outgoing calls
- Traveling class marks (FRLs and / or Call Categories) generated locally by the main and passed on to the ETN tandem for outgoing calls
- Filtering a partial thousand's group of extension numbers to decide which extension numbers for incoming calls terminate to the Main / Satellite / Tributary configuration and which numbers should be routed to the local CO as tail-end hop off calls.

Precedence Calling

To ***enable*** the routing of incoming precedence calls within a DCS, the Standard Networking field (in Procedure 276) and the AAR dial access code (in Procedure 350, Word 2) must be assigned for every node in the AUTOVON access DCS network.

However, AUTOVON (Precedence Capable) trunk groups ***must not*** be included in AAR patterns. AAR is never used to route incoming precedence calls within a DCS. AUTOVON routing patterns are separately (from AAR routing patterns) assigned within the DCS in Procedure 305, Word 1.

Queuing

An incoming tie trunk that infers AAR routing cannot have ringback queuing.

Remote Access

The Remote Access feature provides a means for external callers to access System 85 and Generic 2 features (such as from public network telephones). Remote access trunks are assigned default FRLs that are used by the AAR feature to determine calling privileges in the same way that default FRLs for local stations are used. The Authorization Code feature can also be used with Remote Access to raise the default FRL in the same way as for local station calls except, authorization codes are assigned a network access flag that defines whether or not that authorization code can be used with the Remote Access feature.

Restriction—Code Restriction

The Code Restriction feature has no effect on calls placed using the AAR feature. When a voice terminal user dials the AAR access code (followed by RN(X) + XXX(X)) to access the private network, this access is usually limited by the user's FRL (not by the Code Restriction feature).

Restriction—Miscellaneous Trunk Restrictions

The Miscellaneous Trunk Restrictions feature does not restrict access to trunk groups in an AAR routing pattern.

Restriction—Toll Restriction

The Toll Restriction feature has no limiting effect on AAR calls placed using the AAR access code. Toll Restriction denies toll calls placed over specific trunk groups using the ***trunk-group*** access code. To restrict an AAR user from placing expensive calls, a low FRL should be assigned to the user's class of service in Field 23 of Procedure 010, Word 3.

Route Advance

The Route Advance feature has no effect on the way the Automatic Alternate Routing (AAR) feature selects a preference within an AAR pattern. If a trunk group in an AAR pattern is also the first trunk group in a Route Advance sequence, the AAR software ignores the alternate Route Advance trunk groups while selecting an available (and accessible) trunk group in the AAR pattern.

Tenant Services

The AAR feature is not partitioned. So, a partitioned System 85 or DEFINITY Generic 2 ***cannot*** serve as a tandem node for any tenant's private network. However, with proper assignment, a partitioned System 85 or DEFINITY Generic 2 ***can*** serve as an endpoint for one or more tenant's private networks.

To allow an extension partition to serve as an endpoint in a private network, the partition should be administered as a satellite in a Main / Satellite configuration (where the main also serves as a tandem node in the private network). The tie trunks connecting the main and the "satellite partition" must be dedicated to the partition to prevent unexpected access by other partitions. Under this arrangement, outgoing calls route from the satellite partition to the rest of the private network according to the partitioning assignments. Incoming calls route from the main to the satellite partition according to Extension Number Steering.

Touch-Tone Calling Senderized Operation

When the Automatic Alternate Routing feature uses subnetwork trunking on non-ISDN—PRI trunk groups, each call requires a touch-tone calling sender. If a sender is not available, the switch denies the call.

Restricting Feature Use

The attendant can restrict access to AAR or change FRL values with the following features:

- Attendant Control of Voice Terminals
- Alternate FRLs
- Attendant Control of Trunk Group Access.

Fixed restrictions that deny terminal users access to AAR include:

- Origination Restriction
- Outward Restriction
- Terminal-to-Terminal Only Calling.

Hardware Requirements

For Traditional Modules:

The AAR feature requires touch-tone calling sender (SN252) circuit packs. These circuits reduce call-completion time.

For Universal Modules:

The AAR feature requires tone detector (TN748C) circuit packs. These circuits reduce call-completion time.

Feature Administration

Assignment of the Automatic Alternate Routing feature is on a per-switch basis.

On System 85 switches, this feature is assigned using the Maintenance and Administration Panel (MAAP). The customer can partially administer this feature using the System Management Terminal (SMT), Terminal Change Management (TCM) feature, or Facilities Management (FM) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II. This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

Administration Procedures — Automatic Alternate Routing

Procedure	Word	Purpose	SMT
012	1, 2, 3	Assigns distinctive names to trunk groups.	**Yes**
014	l & 2	Defines a BCCOS and the default Bearer Capability requirements.	N/A
100	1	Assigns a trunk-group dial access code and trunk type to a trunk group.	No
101	1	Administers the characteristics of trunks assigned to a trunk group.	No
103	1	Assigns network trunk-group parameters including minimum FRL, authorization code requirements, incoming tie trunk access to AAR and ARS, Conditional Routing attributes, and the number of trunks that are reserved for first-choice AAR and ARS preferences. If Field 3 equals "1" incoming calls on this trunk group (Field 1) infer AAR or ARS routing.	Yes
110	1	Assigns restricted dial code entry numbers to trunk-group dial access codes.	No
111	1	Administers restricted dial code entry number associated with trunk groups for tandem tie trunk and trunk-to-trunk restrictions.	No

(Continued)

Administration Procedures
Automatic Alternate Routing ***(Continued)***

Procedure	Word	Purpose	SMT
150	1	Assigns trunks (via equipment location) to a trunk group.	No
176	1	Prior to System 85, R2 V4, displays the equipment locations of trunks associated with a specific trunk-group number.	No
177	1	Prior to System 85, R2 V4, displays the equipment locations of the set dial access code.	No
178	1	Beginning with System 85, R2 V4, displays the equipment locations, trunk types, and signaling types of the set of trunks (or specific trunks) associated with a specific trunk-group number or trunk-group dial access code.	No
200	1	Administers attendant console features including Direct Trunk Group Selection feature, class of service display, trunk test, and FRL for the consoles as a group.	No
202	1	Administers the Direct Trunk Group Select buttons (attendant console) and the BUSY/WARNING level for a trunk group(s).	No
275	3	Assigns miscellaneous attributes to the system class of service including the home NPA (area code).	Yes
276	1	Assigns Standard Networking to the feature-group class of service.	No
285	1	Assigns network parameters to the system class of service including network type (hierarchic or symmetric routing plan), barrier code and authorization code requirements, automatic circuit assurance, number of location code [RN(X)] and terminal digits, and the designated extension number for trunk verification. With symmetrical routing, tandem-to-tandem calls route only on the first-choice trunk group.	Yes
312	3	Assigns "011 Screening" for International Call Routing. Used to create or change the digit translation table that converts selected "011" digit strings to point to international routing patterns that can contain both private and public facilities.	No

(Continued)

Administration Procedures Automatic Alternate Routing *(Continued)*			
Procedure	**Word**	**Purpose**	**SMT**
320	1	Assigns routing attributes to call categories.	Yes
321	1	Administers routing-pattern parameters for AAR including trunk-group number, FRL, warning tone, off net (to a DDD or IDDD location), number of digits deleted for an on-net route, whether network has 0XXX extension numbers or not, and the IXC identifier (0 - no IXC).	Yes
321	2	Defines the digit grouping and dialing format (touch-tone or rotary) for a trunk group in an AAR pattern that uses subnetwork trunking.	Yes
321	3	Defines the digits to insert for a trunk group in an AAR pattern that uses subnetwork trunking.	Yes
321	4	Associates a location code [RN(X)] and call category with an AAR pattern number.	Yes
321	5	Assigns a BCCOS and network specific facility to an AAR preference.	N/A
330	1	Assigns the number of trunk groups in a pattern used for queuing. (A "1" indicates the first choice only. A "7" indicates the first seven.) This procedure also assigns FRL Raising for AAR.	Yes
350	1	Assigns the first digit of a dial access code, or extension number. The first digit is defined in terms of the number of digits the switch expects to receive and the call type.	No
350	2	Assigns the AAR dial access code. The applicable encode is as follows: 61 AAR Call.	No

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Automatic Alternate Routing	
Path Name	**Purpose**
terminal-change class-of-service attributes	Assigns an FRL to an extension class of service.
terminal-change system parameters	Assigns an authorization code for access to AAR routing patterns.
terminal-change names trunk-group-names	Assigns distinctive names to trunk groups.

The following are the applicable FM path names used with the AP 16. A printed report of the displayed information can also be generated.

FM Screens — Automatic Alternate Routing	
Path Name	**Purpose**
facilities-mgmt routing alternate-routing patterns	Displays and changes attributes associated with an AAR pattern including: ● Pattern number ● Preference [first- to fourth-choice group (System 85, R2 V1 and R2 V2)] [first- to sixteenth-choice (System 85, R2 V3)] ● Trunk-group number ● Minimum FRL ● Off-network signaling, international destination, DID at destination, inserted digits ● On-network digits deleted and warning tone.
facilities-mgmt routing alternate-routing location-codes	Displays and changes an AAR pattern and associated call category assigned to a location code.
facilities-mgmt routing alternate-routing call-category	Assigns attributes to call categories.
facilities-mgmt routing alternate-routing rearrangement	Displays and changes the trunk groups that make up an AAR pattern. If the pattern does not use subnetwork trunking, trunk groups can be removed or reordered.
facilities-mgmt routing conversion	Displays and changes the correspondence between a private-network location code and the public-network telephone number that routes to that location code (see the ARS feature description for an explanation of 10-Digit Conversion). Either the location code or the public-network destination code (telephone number) can be entered, and the corresponding values are displayed.

Automatic Call Distribution

Description

The ACD (Automatic Call Distribution) feature permits incoming trunk calls, local voice terminal calls, and attendant-extended calls to terminate to the most idle voice terminal in a group of terminals. The answering positions appear as normal voice terminals to the switch and are also able to place and receive calls in the usual manner. Selected terminal users (agents) can be organized into a split (collection of agents) to allow for balanced call distribution to the agents.

The ACD feature can also distribute calls to Message Center, AUDIX (Audio Information Exchange), and ISDN Gateway. These are software applications that run on an adjunct processor. From a switch administration perspective, these software applications are ACD split types, and like regular ACD splits, can be assigned appropriate ACD functions such as recorded announcements. ACD split types are described later in this section.

Feature History and Development

This feature was first available in Release 2, Version 3 to replace and enhance the functionality provided by the EUCD (Enhanced Uniform Call Distribution) feature that was included in Release 2, Version 2. Likewise, the EUCD feature replaced the UCD (Uniform Call Distribution) and DDC (Direct Department Calling) feature that were first available in Release 1.

The ACD feature has several major enhancements for R2 V4. These enhancements include:

- Call Vectoring Compatibility

  When the Call Vectoring feature is used in conjunction with ACD, significant flexibility and power are added to ACD's call-handling capabilities.

- Sixty ACD splits

  The maximum number of ACD splits was increased from 30 to 60 in R2 V4. With this increased maximum, ACD call answering operations can be subdivided into more functional categories. As an example, the System 85 and DEFINITY Generic 2 limit of 1024 ACD agents in 60 splits can be reached with the following configuration:

  — 4 splits of 32 agents, and

  — 56 splits of 16 agents.

- Queue-Status Display

  This function provides ACD agents (equipped with display voice terminals) with periodic updates of the number of calls in the split's queue and the amount of time the oldest queued call has waited.

- Multiple Call Handling

  Primarily for Message Center agents, this function allows ACD agents to put calls on hold (using the HOLD button) and make themselves "available" to receive ACD calls.

- Malicious Call Trace Compatibility

  Using the Malicious Call Trace feature, an ACD agent can quickly and easily activate a trace that can identify the calling party of a malicious call.

- Improved REPEAT Button Operation

  Using an R2 V4 ACD, an agent can press the REPEAT button to repeat any origin announcement that the agent just heard. VDN-, city-, or queue-of-origin. Using a call distributor prior to R2 V4, the REPEAT button only repeats a city-of-origin announcement.

- Improved Agent Call Pegging for CMS (Call Management System)

  Beginning with Issue 1.4 of R2 V3 and Issue 1.1 of R2 V4, the switch will peg agent calling activity more accurately. Before pegging a call placed by an ACD agent, the switch will review the dialed digits and, for internal calls, monitor the call's progress. In this way, the CMS will not be notified when feature access codes, many invalid numbers, and many calls resulting in busy tone or reorder tone are dialed by ACD agents.

  In the earlier issues of the R2 V3 and R2 V4 software, the switch notifies the CMS whenever an agent goes off-hook on an idle appearance. This method of pegging the calling activity of ACD agents can distort the corresponding CMS statistics by reporting agent state changes, feature requests, and many invalid or misdialed numbers as completed outgoing calls.

- Look-Ahead Interflow

  Beginning with R2 V4, Issue 1.3 and DEFINITY Generic 2, the Look-Ahead Interflow feature can be used in conjunction with ACDs that ***also*** use the Call Vectoring and the ISDN (Integrated Services Digital Network)/PRI (Primary Rate Interface) features. (Refer to the Look-Ahead Interflow chapter of this manual for a detailed description of this operation.)

Beginning with Issue 3.0 of R2 V4 System 85 and DEFINITY Generic 2.1, 3-burst zip tone is no longer given for interflow calls (except Look-Ahead Interflow calls) that route over ETN trunk groups.

Biginning with Issue 3.0 DEFINITY Generic 2.1, an RLT (Release Link Trunk) can terminate to an ACD split, a VDN (Vector Directory Number), or an attendant console. Before this enhancement, an RLT could only terminate to an attendant console.

The following enhancements apply to the ACD feature beginning with DEFINITY Generic 2.2:

- 2048 Agents Per System

  The maximum number of ACD agents per system increases from 1024 to 2048.

- 255 Recorded Announcement Trunks

  The number of recorded announcement trunks has been increased from 84 to 255.

- Split Size Restrictions Removed

  Split size restrictions have been removed. That is, the size of an ACD split does not have to be specified in multiples of 16. An ACD split may contain any number of agents from 1 to 1024 (1023 if the split is measured by CMS).

- Splits Administered as Measured by CMS

  ACD splits are administered as either measured by CMS or not. As many as 1023 extension numbers can be measured by CMS. Prior to DEFINITY Generic 2.2, individual extension numbers or a range of extension numbers is administered as measured by CMS.

- 106B Display Unit Assignments Based on Extension Number

  The assignment of agents to 106B display units (used to monitor calling activity) is based on extension number. Prior to DEFINITY Generic 2.2, 106B assignments are based on agents' split and member numbers. If an agent assigned to a 106B moves from one split to another, the 106B assignment has to be changed. Beginning with DEFINITY Generic 2.2, 106B assignments do not have to be changed when agents move from one split to another.

- Lamp Indication for Stroke-Count Buttons

  If the stroke count function is assigned to a button with a status lamp, the status lamp lights for 2 seconds if a message can be sent to CMS when the button is pressed. If a message cannot be sent, the flash rate of the status lamp is set to broken flutter for 2 seconds.

Refer to Appendix B for a tabular comparison of the various call distributors provided in System 85, Release 2 and DEFINITY Generic 2.

Related Features

The following System 85 and DEFINITY Generic 2 features are commonly used with the ACD feature.

- ASAI Gateway Interface
- AUDIX
- Call Vectoring
- Call Work Codes
- Expert Agent Selection
- Look-Ahead Interflow

Related Documents

The following documents contain information about the *HOME AGENT*™ application. The *HOME AGENT* application runs on AT&T's *Conversant*® Voice Information System and works with the ACD feature on R2 V4 System 85 and DEFINITY Generic 2.1 and later switches. As the name implies, the *HOME AGENT* application enables the ACD feature to distribute incoming calls to agents who work at home. Home agents can be assigned to any ACD split and can provide the same services as local agents.

- *DEFINITY Communications System and System 85 HOME AGENT Application, Installation and Operations* (555-035-501)
- *DEFINITY Communications System and System 85 HOME AGENT Application, Trainer's/Supervisor's Guide* (555-035-751)
- *DEFINITY Communication System and System 85 HOME AGENT Application, Home Agent's Instructions* (555-035-705)

Call Distribution

A published number is linked to an ACD split by associating the published number with the extension number of the first terminal in the split. The controlling or primary terminal for the split (the split supervisor's terminal) is the first terminal in the split's list.

The switch directs incoming calls for a published number to the associated split's queue. Calls are distributed from the queue to available agents according to the call distribution (hunting) type assigned to the split. Three types of call distribution are available: linear hunting circular hunting and MIA (most idle agent) distribution.

Linear Hunting

Linear hunting (also called Direct or Terminal Hunting) always starts with the first agent (split supervisor) and hunts toward the last member. Linear hunting is used for applications where a priority series of answering positions is desired.

> **NOTE:** Linear hunting provides the call-distribution algorithm that was formerly provided by the DDC (Direct Department Calling) feature.

Circular Hunting

Circular hunting starts where the hunting process left off during the previous scan and continues through the list of agents. After checking the final member of the list, circular hunting again returns to the first member of the list and continues in a circular fashion. Circular hunting is useful for applications, such as order taking or Message Center, where a more evenly balanced call distribution is necessary.

> **NOTE:** Circular hunting provides the call-distribution algorithm that was formerly provided by the UCD (Uniform Call Distribution) feature.

MIA (Most Idle Agent) Distribution

The MIA type of call distribution distributes an ACD call to the available agent for whom the longest period of time has elapsed since the agent has finished an ACD call. Agents who are in the Staffed mode of operation and are not handling an ACD call are placed in an agent queue. The call from the head (top) of the split's incoming call queue is distributed to the agent at the head of the agent queue. The MIA type of call distribution, like circular hunting, is used for applications where an evenly balanced call distribution is necesary. This type of call distribution is especially useful when there is substantial variation in the duration of ACD calls.

NOTE: The MIA call-distribution algorithm provides what is commonly known in the trade as "true" ACD.

Agents are marked as "unavailable" with regard to the agent queue when they are in the Aux-Work mode of operation, or are active on a personal call. In this way, agents, although unavailable for another ACD call, are moved ahead in the agent queue. If an unavailable agent moves to the head of queue, the agent is bypassed for distribution of ACD calls until becoming available. When available, the next ACD call quickly routes to that agent.

For some ACD applications, outgoing calling is an integral part of the agents' duties. For other applications, an agent rarely places an outgoing call. The MIA method of distribution allows outgoing calls either to be considered as work related (agent is removed from the agent queue) or personal (agent remains in queue and is marked as unavailable). This choice is made on a per-split basis, allowing the choice to be based on the primary application of each split.

Each terminal (including the split supervisor's terminal) in an ACD split can receive calls either as a split member or as an individual terminal. For internal calls, unique ACD split extension numbers, called associated numbers, identify the split. For incoming calls, the ACD split is associated with incoming trunks. For automatic-in type trunks, the call routes to the assigned ACD split. For dial-repeating type trunks, the call routes to the split dialed. This is similar to the way DID calls complete to individual terminals (including individual ACD split members).

Priority Queuing

One of the associated numbers and any number of the incoming trunk groups may receive a priority designation. Priority calls are placed at the head of the queue or behind priority calls which are already in queue. A nonpriority call enters the queue behind all other calls.

An example of priority queuing is shown in Figure 17-1. Within this figure, the priority calls are ***lettered,*** and the nonpriority calls are numbered. One associated extension number (the maximum) and two trunk groups are delivering incoming calls to the end of the priority portion of the queue. Meanwhile, two associated extension numbers and two trunk groups are delivering incoming calls to the end of the nonpriority portion of the queue. Each call is distributed from the head of queue (currently, Call A) to an available agent in Split 29.

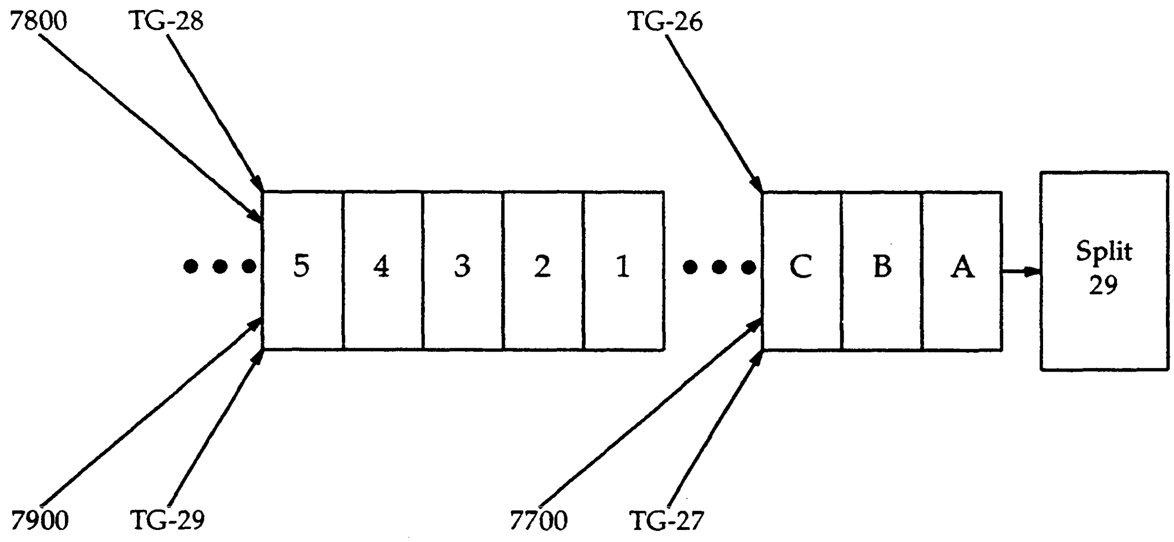


Figure 17-1. ACD Priority Queuing

ACD Split Types

An ACD split can be administered (Procedure 026, Word 1) as one of the following types:

- Regular

  A normal ACD split staffed by agents equipped with a voice terminal, a voice terminal and a data terminal, or a work station that combines the capabilities of both voice and data terminals.

- Message Center

  Similar to a regular ACD split, but the split is staffed by message center agents. Message Center is a software application that runs on an adjunct processor.

- AUDIX

  AUDIX is a message-handling or "Voice Mail" system that runs on a adjunct processor. The AUDIX system uses recorded prompts and announcements to guide callers through the messaging operations. Unlike regular and message-center splits, AUDIX splits are not staffed by live (human) agents. The "agents" are switch ports that are connected to the adjunct processor.

- ISDN Gateway (number only)

  ISDN Gateway is a software application that runs on an adjunct processor. The ISDN Gateway adjunct receives call-related information from the switch, converts the information into a form that a customer's application software can use, and then sends the converted information to the application. The customer's application software can, for example, use the call-related information to retrieve information from a data base (on the ISDN Gateway adjunct or a separate host computer) or accumulate call-related statistics. Information retrieved from a data base can be sent to an answering position and displayed on a data terminal. For this split type, the switch sends the calling party's telephone number to the ISDN Gateway adjunct.

- ISDN Gateway (name and number)

  This split type is similar to ISDN Gateway (number only) except that the switch also sends the calling party's name to the customer's application software.

Automatic Available Splits

An ACD split can be administered as automatic available. Before administering a split as automatic available, make sure all agents assigned to the split are unstaffed. Field 6 (Field 5 in R2 V3, beginning with Issue 1.4) of Procedure 026, Word 2 assigns "automatic availability" to every agent position in an ACD split. Automatic availability should ***only*** be assigned to an ACD split that is serving as an unmeasured or measured gateway to a VRU (Voice-Response Unit) or other automatic answering device.

Many VRUs cannot automatically restaff or login their agent positions after a hard processor swap or after a commercial power failure. When automatic availability is assigned to the split, the switch runs a regularly scheduled audit to automatically restaff the agent positions

Automatic availability ***should not*** be assigned to traditional ACD splits with human agents. If this is done, ACD agents lose the ability to maintain their requested unavailability. For example, if an agent enters the Aux-Work mode to go on break, the request is accepted. However, as soon as the scheduled audit runs, the agent is returned to the available mode, and ACD calls are distributed to the agent's unoccupied position.

Also, automatic availability ***should not*** be assigned to an AUDIX split. The AUDIX system has its own method of restaffing the gateway ports after a hard processor swap or after a commercial power failure.

When assigned to an ACD split, automatic availability does not apply to Member 0 (the "supervisor") of the split. Since the switch does not restaff Member 0 of an automatic available split, this agent position can be equipped as a regular voice terminal. In this way, the access codes can be dialed on this voice terminal (in addition to using Procedure 026, Word 2) to add agent positions to and remove agent positions from the automatic available split.

> **NOTE:** As an agent position is added to or removed from a measured automatic available split, the switch automatically notifies the CMS with the appropriate messages.

Methods of Routing Incoming ACD Calls

The two ways to route incoming ACD calls to the System 85 or DEFINITY Generic 2 switch include:

- Dial-repeating type routing

  Using this method, an associated extension number's digits are passed through the serving switch (usually, the serving Central Office) and to the local switch in a similar manner to the way that DID calls are routed. As the associated extension number's digits are analyzed by the switch's call-processing software, the dialed number is recognized as a number that terminates to a specific split's queue. In turn, the call-processing software gives control of the incoming call to ACD processing.

- Automatic-in type routing

  Using this method, an ACD call is recognized by the serving switch (usually, the serving Central Office) as a call that is routed to the local switch over a specific trunk group. In turn, the local System 85 or DEFINITY Generic 2 accepts the call from over the incoming (or 2-way) trunk group, and recognizes the call as assigned to terminate to a specific split's queue. This method of routing resembles "non-DID routing" to the attendant-queue. In fact, "attendant-completing" trunk groups are often used for automatic-in type routing of ACD calls. Automatic-in type trunk groups are assigned as "attendant-completing in" or "automatic in" in Procedure 100, Word 1 and then assigned to terminate to an ACD split in Procedure 115.

  The trunk types that can be assigned to terminate to an ACD split in Procedure 115 include:

  - 16 = CO 1-way in attendant-completing
  - 19 = CO 2-way attendant-completing in / DOD out
  - 20 = CO 2-way with party test attendant-completing in / DOD out
  - 21 = FX 1-way in attendant-completing
  - 24 = FX 2-way attendant-completing in / DOD out
  - 25 = FX 2-way with party test attendant-completing in / DOD out
  - 26 = WATS l-way in attendant-completing
  - 35 = TIE l-way in automatic
  - 38 = TIE 2-way automatic in / dial repeating out
  - 39 = TIE 2-way automatic in and out
  - 50 = Remote Access 2-way*
  - 66 = CAS release link trunk 1-way incoming at main.

* Used only for Remote Access speaker verification.

Outgoing Calling by ACD Agents

This ACD feature description primarily focuses on ACD from the perspective of incoming calls. For example, ACD can distribute high volumes of "incoming calls to agents in a fair and cost-effective manner. The ACD feature can also provide recorded announcements and / or music for a desirable calling party interface. Moreover, several management tools are available (most notably, CMS and service observing) to hold costs in line and to ensure effective agent performance.

However, especially when ACD is applied to address the needs of a telemarketing organization, agents can also devote significant effort placing outgoing calls (e.g., to solicit sales). In this environment, an ACD manager is also concerned with the costs of outgoing calling. Although the ACD feature does not directly address these concerns, a number of coresident features on System 85 and DEFINITY Generic 2 do.

As previously mentioned, ACD agents in addition to receiving calls distributed from a split's queue, are able to place and receive calls in the usual manner. Given this ability, ACD agents have the same access as other users to the least-cost-routing capabilities of the ARS (Automatic Route Selection), AAR (Automatic Alternate Routing), WCR (World Class Routing), and IXC (Interexchange Carrier Access) features. Indeed, the switch administrator can control the costs of ACD outgoing calling with the same methods that costs are generally addressed on System 85 and DEFINITY Generic 2.

Call Management System

A System 85 or DEFINITY Generic 2 can send information about ACD activity over a DCIU (Data Communications Interface Unit) link to a 3B2 or 6386 computerrunning CMS software. The CMS software collects, stores, and formats this information for real-time system-status displays and scheduled or on-demand historical printed reports. Beginning with Issue 1.3 of Release 2 CMS, the optional graphics feature displays real-time data in bar-graph form.

Real-time displays show the current performance and status of individual agents, splits (or other defined agent groups), trunks, trunk groups, calls, and call vectors For example, the following performance information can be displayed for splits or VDNs:

- Number of calls waiting
- Oldest call waiting
- Average speed of answer
- Number of calls abandoned
- Average time to abandon
- Number of ACD calls
- Average talk time (ACD calls and outgoing calls)
- Average after-call-work time.

When pre-set system thresholds are exceeded, for example too many calls in queue, supervisors can be notified by way of real-time exception displays.

Stored information can be printed on demand or automatically on a daily, weekly, or monthly basis. These reports and a sophisticated forecasting capability help managers make intelligent decisions and forecasts based on current and past usage, seasonal variations, trends, and growth factors.

CMS can also be used to partially administer agent positions, trunk groups, call vectors, and parameters associated with each ACD split. System administration can be done on demand or automatically at predetermined times.

To help conceptualize this complex feature, a sample ACD/CMS system is shown in Figure 17–2. The contents of the dashed box describe the ACD functions provided by the System 85 or DEFINITY Generic 2.

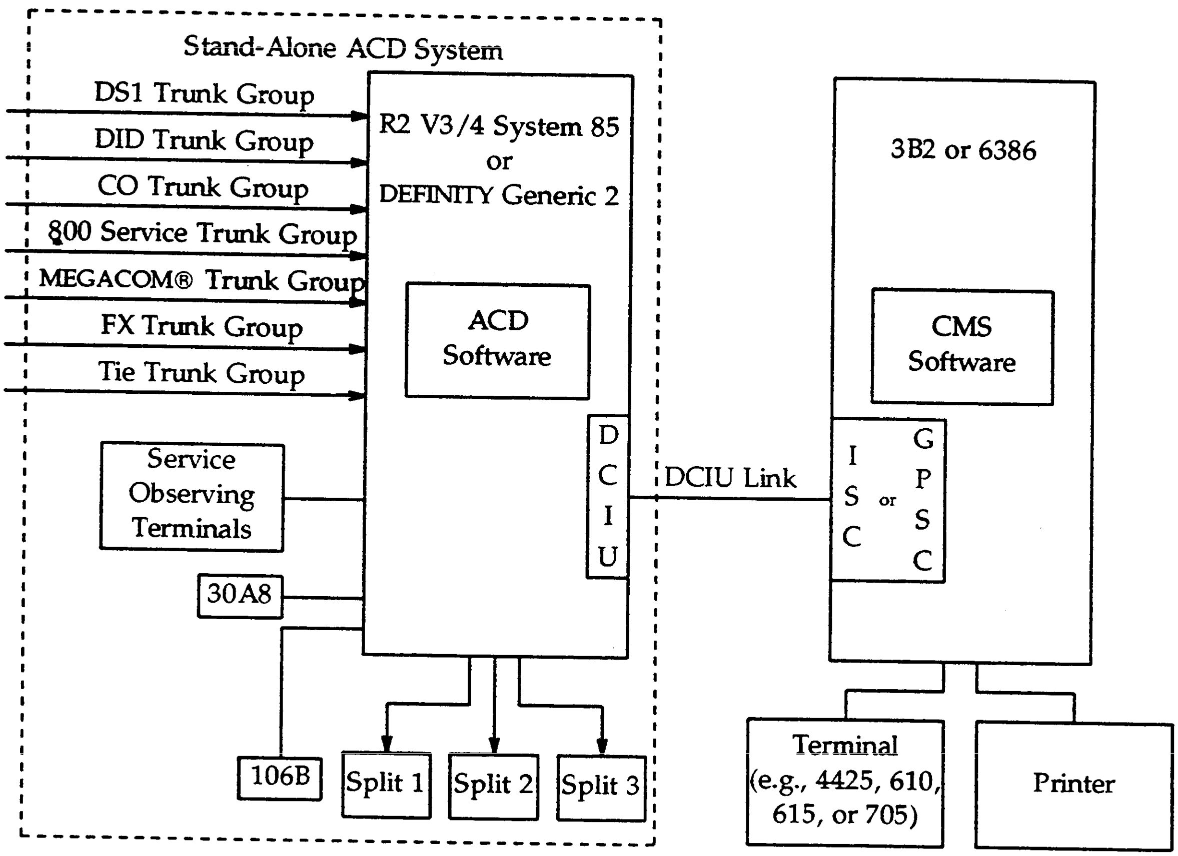


Figure 17-2. Simplified Pictorial of an ACD/CMS System

Release 3.0 CMS

To collect information about ACD activity on a DEFINITY Generic 2.2, Release 3.0 CMS is required. The following are some of the enhancements Release 3.0 CMS offers:

- ACD calls transferred or conference by agents who are measured by CMS. Calls are classified as internal or external and the dialed digits are sent to CMS.
- Personal Calls placed or received by measured ACD agents. When available, the dialed digits are sent to CMS.
- Calling-party identification for incoming-trunk calls and internal calls that encounter a measured trunk group, VDN, or split. Calling-party identification is only sent for ISDN and DCS calls for which calling-party information is available. For other types of calls, no calling-party identification is sent.
- Calls that are abandoned before they are queued to an ACD split whether or not they encounter a measured trunk group or VDN.
- For systems that have the ASAI Gateway Interface feature, calls initiated by the call-center software on behalf of an agent.
- For systems that have the Look-Ahead Interflow feature, successful and unsuccessful attempts to route a call to another switch.

On DEFINITY Generic 2.1, CMS has to be busied out (Procedure 28, Word 2) whenever administration changes to ACD splits are made using Procedure 026, Words 1 through 3.

On DEFINITY Generic 2.2, CMS only has to be busied out (Procedure 28, Word 1) when the following administration changes are made:

- Using procedure 26, Word 3, a previously unmeasured agent extension is changed to measured (added to a measured split), or a previously measured extension is changed to unmeasured (removed from split 0).
- Using Procedure 26, Word 2, a previously unmeasured split is changed to measured, a previously measured split is changed to unmeasured, or a new split is added as a measured split.

The following are examples of ACD administration on DEFINITY Generic 2.2 that do not require CMS to be busied out (CMS does have to be busied out for earlier versions of switch software):

- Adding an agent to or removing an agent from a measured split.
- Adding a new unmeasured split.
- Assigning multiple call handling to a split.
- Assigning automatic available to a split.
- Changing the hunt type of a split.
- Changing the inflow threshold or the queue warning lamp (outflow) threshold.

Refer to the following manuals for more detailed information about CMS.

CMS Administration (585-215-511)

CMS Custom Reports (585-215-513)

CMS Installation and Maintenance for WGS Computers (585-215-112)

CMS Installation and Maintenance for 3B2 Computers (585-215-111)

Lamp Monitoring of ACD Agents

Display units (106B) can be provided to show the current activity of 20 ACD agents. For each agent, the unit displays one of the following agent states:

- Available for ACD call. [First lamp (top)]
- On ACD call. [Second lamp]
- After call work. [Third lamp]
- Aux work. [Fourth lamp]
- On non-ACD call (e.g., personal call). [Fifth lamp (bottom)]

  NOTE: This state includes any incoming- or outgoing-calling activity (from the perspective of the agent's voice terminal) that is considered non-ACD related.

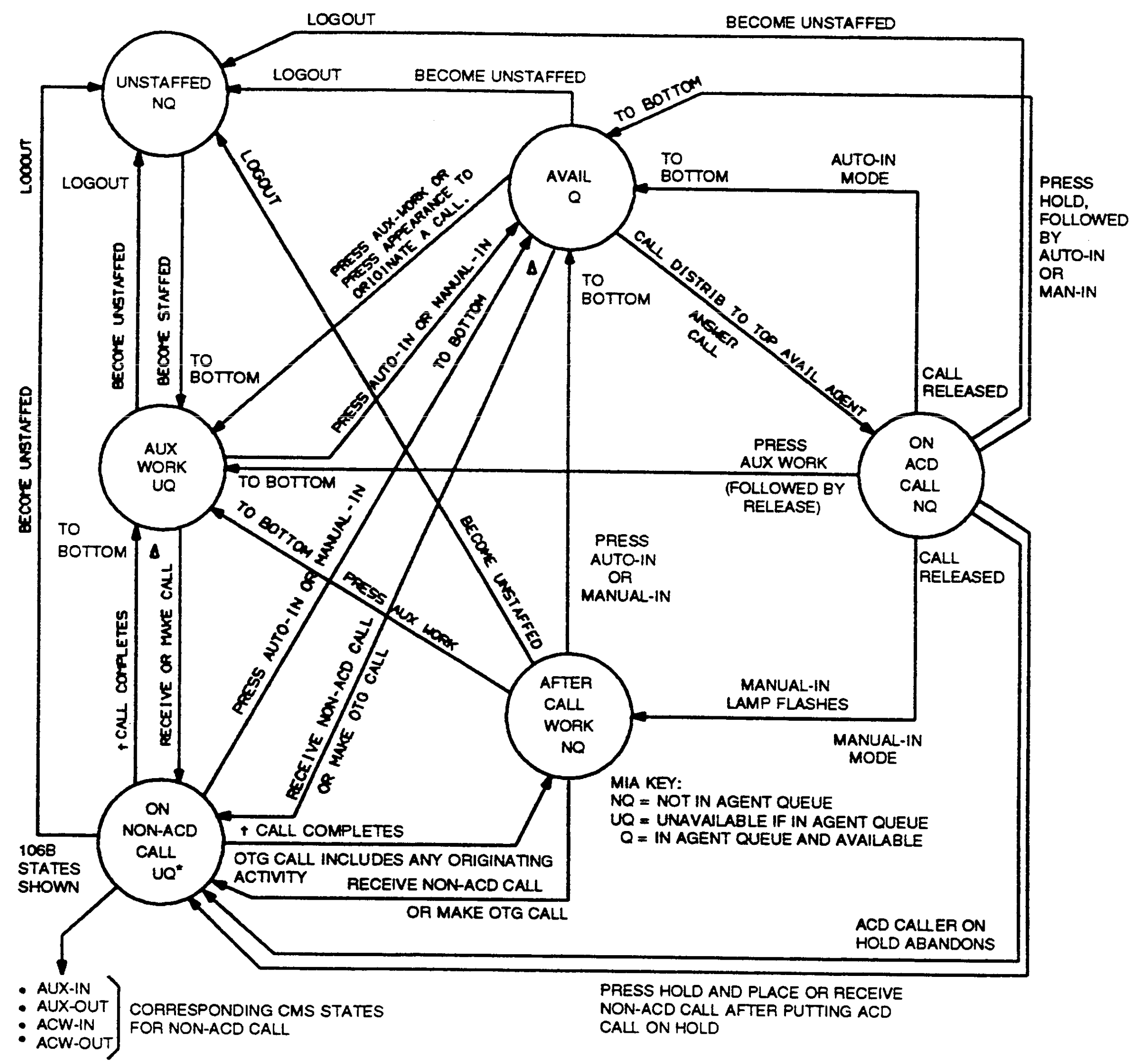
- Unstaffed. [All lamps off]

An agent's current state (activity) can only be displayed on one display unit.

Prior to DEFINITY Generic 2.2, display-unit assignments are based on agents' split and member numbers. If an agent assigned to a display unit moves from one split to another, the display-unit assignment must be changed. Beginning with DEFINITY Generic 2.2, display-unit assignments are based on extension number and, consequently, display-unit assignments do not have to be changed when agents move from one split to another.

Agent State Diagram

Figure 17-3 is a state diagram describing the interrelation of these six agent states and the queuing details of the MIA call-distribution algorithm. The diagram shows the transitions of an ACD agent fropm one state to another during routine calling operations. As an example, an agent in the After Call Work state who presses the MANUAL-IN button is retumed to the end (bottom) of the agent queue as an available agent.



NOTES:

∆ If not already in the agent queue.

* Administration is available for outgoing call treatment. The agent can either keep the same position in the agent queue or return to the end (bottom) of the agent queue when the call finishes.

† Agent returns to the previous state after non-ACD call is finished (if the agent does not manually change states using a button or an access code).

Figure 17-3. Agent State Diagram Including MLA-Distribution and CMS Interactions

Answer Supervision

Answer supervision is a signal sent by switch to the serving CO (Central Office) indicating that an incoming call has been answered. Upon receiving this signal, the originating CO (generally) begins tracking toll charges for the call (if charges apply). For ACD calls to System 85 or DEFINITY Generic 2 which enter a queue before being answered, answer supervision must be returned (at the latest) either just before an agent actually answers or just before the first recorded announcement (if provided), whichever comes first. This is the ***preferred operation,*** and is provided automatically by R2 V3 (beginning with Issue 1.4) System 85, R2 V4 System 85 and DEFINITY Generic 2.

For earlier issues of R2 V3, the preferred operation is provided on a system-wide basis when Field 12 of Procedure 275, Word 4 is set to "1." When this field is left with the default entry "0," System 85 returns answer supervision for ACD calls at the time each call first enters queue. This operation is not recommended, because it results in higher toll charges for the customer (when 800 Service is used) and/or for each calling party.

Answer Supervision and Incoming ISDN—PRI Trunks

Beginning with R2 V4, the switch can still return answer supervision at the preferred times for incoming calls from over ISDN—PRI trunk groups. However, the answer-supervision protocol is different with these incoming trunk groups.

As the System 85 or DEFINITY Generic 2 receives an incoming ISDN—PRI call destined for an ACD split's queue, the switch returns a CALL PROCEEDING message to the serving switch. When the billing switch receives this message, this switch infers that the ISDN call has successfully negotiated for a B channel in the ISDN—PRI trunk group, but that answer supervision is not yet being returned. Then, just before an agent answer (or the first recorded announcement plays), the System 85 or DEFINITY Generic 2 returns an "Accept" (CONNECT, ALERTING, or PROGRESS) message to the serving switch. When the billing switch receives a CONNECT message, billing begins for the ISDN—PRI call.

Answer Supervision and Abandon Call Search

As discussed, the switch returns answer supervision to the serving CO just prior to connecting the call to the first recorded announcement. However, after answer supervison has been returned, "ghost calls" can occur if the calling party abandons the call.

A ghost call occurs whenever an agent answers a call ***after*** the calling party hangs up, but ***before*** the CO returns the disconnect signal to the System 85 or DEFINITY Generic 2. Primarily depending on the type of CO, the delay in returning disconnect is on the order of 2 to 25 seconds. If a ghost call were to occur, an ineffective call would be distributed to an agent. Also, as if the agent had received an actual ACD call, the agent would be inappropriately credited with an ACD call and bypassed for distribution of subsequent ACD calls.

To minimize this problem, abandon call search can be assigned to the switch (Procedure 275, Word 4). When abandon call search is assigned to System 85s prior to

R2 V4, Issue 1.2, the switch checks incoming CO, 800 Service, and FX trunks just before ringing an idle ACD agent. If the trunk is found to be on-hook at the CO, the switch releases the trunk. If the trunk is active, the switch distributes the call to the idle agent.

Separation of Trunk and Signaling Types

This algorithm was modified to conform with separate trunk and signaling types. Beginning with R2 V4, Issue 1.2, the switch checks incoming trunks with the following signaling types just before ringing an idle agent:

1. Ground start
2. Ground start with party test.

In a similar manner to the previous algorithm, if the ground-start trunk is found to be on-hook at the serving switch, the local switch releases the trunk. If the trunk is active, the local switch distributes the call to the idle agent.

NOTE: It is recommended, whenever processor occupancy allows, that abandon call search be assigned to the switch.

Split Supervision

Supervision of an ACD split is primarily provided by a split supervisor. The split supervisor, Member 0 of the split, may be an experienced agent who performs agent training duties, serves as a consultant to the group of agents, and regulates the split's operation. In addition to the usual agent capabilities, the split supervisor is able to:

- Observe agent performance using agent override or service observing
- Add or remove agents to/from the split
- Verify the split's recorded announcement
- Activate or deactivate Call Forwading for the split
- Activate or deactivate Overload Balancing for the split
- Turn off the reload warning lamp after a tape reload by the switch.

Also, one local attendant-console operator can be designated as the system supervisor. The supervisor cannot perform agent duties from this attendant console, but may perform the following tasks:

- Activate or deactivate Call Forwarding for a split
- Turn off the reload warning lamp after a tape reload by the switch.

Intraflow and Interflow

Intraflow, diversion of ACD calls to a local destination, and interflow, diversion of ACD calls outside the System 85 or DEFINITY Generic 2, are available to increase call-distribution flexibility. Call Forwarding is primarily used as the software mechanism for intraflow (local redirection), while overload balancing is usually chosen as the mechanism for interflow (distant redirection). Both intraflow and interflow can operate in two ways: ***threshold*** redirection and ***unconditional*** redirection (redirection of ***all*** calls). Threshold redirection diverts ACD calls during a heavy call load. Unconditional redirection diverts all calls when an ACD split is inactive.

NOTE: When interflowed ACD calls use the AAR, ARS, or WCR feature to route these calls outside the local switch, the FRL of the split's supervisor is used to determine whether these calls can use the available network facilities.

Intraflow

"Chaining" of Destinations for Intraflow—All

When Intraflow-All is activated, the split can have as many as three local destinations (e.g., an extension, the attendant group, or another ACD split) for unconditional direction of ACD calls. These destinations are arranged in a priority scheme by order of activation. If the first destination is unavailable, the switch checks the second and third destinations (while the call remains in the first split's queue). Of these three destinations, only the last destination should either be another ACD split or the attendant group (i.e., be accessed by a queue).

NOTE: When a destination before the last destination has a queue, the call unconditionally enters the queue, and subsequent destinations are never checked.

If an unconditionally intraflowed call enters an ACD queue, and this queue ***also*** has Intraflow-All active, the call will recursively intraflow again. This time, the call intraflows to one of the new split's destinations.

If an unconditionally intraflowed call enters an ACD queue, and this queue has Intraflow-Threshold active, the call will not intraflow again. Instead, this call is marked as "Intraflow-Threshold Not Allowed," and must be answered by an agent in this split.

No "Chaining" of Destinations for Intraflow—Threshold

There can also be three local destinations (e.g., an extension, the attendant group, or another ACD split) for threshold redirection of ACD calls. These destinations are arranged in a priority scheme by order of activation. If the first destination is unavailable, the switch checks the second and third destinations (while the ACD call remains in the first split's queue). All three of these destinations can be ACD splits (i.e., accessed by a queue). When the inflow parameter at a receiving split is met, the switch executes the intraflow operation and marks the call as "Intraflow—Threshold Not Allowed." These calls are not allowed to intraflow again. ACD calls are not allowed to conditionally intraflow throughout the switch in a recursive manner.

NOTE: Intraflow-Threshold operates to attenuate heavy call loads while the split has active ACD agents to answer the calls. So, when the parameters for the first intraflow operation are carefully chosen, the need to redirect the same ACD call again is minimal.

If a conditionally intraflowed call enters an ACD queue, and this queue has Intraflow-All active, the call will recursively intraflow again. This time, the call intraflows to one of the new split's destinations.

Parameters for Intraflow Threshold Redirection

Threshold (overflow) redirection operates when the number of calls that are waiting in a queue is greater than or equal to the preset overflow level. A split's overflow level can be set to any value between 1 and 99. However, if the destination for redirection is an ACD split, the call to be diverted will only be accepted into queue if the number of calls in queue is less than or equal to the destination split's inflow level, and there is at least one ***staffed agent*** in the destination split. A split's inflow level can be set to any value between 0 and 98. When overflow redirection does occur (to one of the destinations that is checked at 2-second intervals), the switch diverts the first call in queue.

NOTE: When an inflow level is set to "0," a call will not be accepted unless there is ***at least one available agent*** in the destination split.

NOTE: The same field that assigns the overflow level (Procedure 026, Word 1, Field 5) also determines the threshold amount of queued calls that causes the split's queue warning lamp (on the 30A8 panel) to light.

For example, split A has an overflow level of 21 and split B has an inflow level of 5. Every agent in split A is busy on a call and the queue contains 20 waiting calls. If split B has five or fewer calls in queue, the incoming call will enter split A's queue and the first call in split A's queue will redirect to split B. If split B has six or more calls in queue, redirection can occur to an alternate destination (if assigned), but not to split B. If every destination for redirection of ACD calls is busy, the incoming call enters the original queue as the 21st call.

NOTE: Assigning the overflow level for split A to "1" would have the effect of emptying the queue. This operation for diverting ACD calls would be applied to the first call attempting to enter the queue.

NOTE: For R2 V4 System 85 and DEFINITY Generic 2, the Call Vectoring and Look-Ahead Interflow features can provide a more flexible and powerful way of handling queue overflow conditions. Refer to the Call Vectoring and Look-Ahead Interflow features for a description of these capabilities.

Interflow

Each split can have one destination outside the local switch for diversion of ACD calls. This destination is usually provided via Overload Balancing. Overload Balancing is always automatically implemented as the last priority. The Overload Balancing destination can be used alone or after the list of local destinations. As such, there can be as many as four possible destinations.

Threshold interflow diversion is always provided using the Overload Balancing—Overflow access code (Encode 85). Whereas, unconditional interflow diversion is usually provided using the Overload Balancing—All access code (Encode 84).

Using AAR/ARS/WCR With Overload Balancing

The AAR, ARS, and WCR features are used in conjunction with Overload Balancing to divert ACD calls outside the switch. Any private network or public network telephone number can be specified as an Overload Balancing destination as long as an

AAR/ARS/WCR pattern has been translated for the destination. (Normally, Overload Balancing is used to divert ACD calls over an ETN trunk group to an associated extension number of an ACD split residing in a different ETN node.)

As the interflow software sends Overload Balancing calls to the AAR/ARS/WCR features for routing, these Overload Balancing calls can be allowed to route over any preference (trunk group) within the routing pattern. (The preferences actually allowed are determined by the split supervisor's FRL.)

To allow interflowed calls to route without calling-party intervention, the following network muting parameters are ignored:

System 85 and DEFINITY Generic 2.1

- Authorization Code — The calling party will not be prompted to enter an authorization code if the FRL associated with the call is not high enough to access any preference in the selected AAR/ARS pattern.
- CDR Account Code — The FEAC (Forced Entry of Account Codes) parameter can be assigned to a trunk group in an AAR or ARS pattern, to the system class of service (for ARS calls), or to a line class of service. If FEAC is assigned to a trunk group and the Overload Balancing destination does not contain an account code, the trunk group will be skipped. If FEAC is assigned to all trunk groups in the selected pattern and the Overload Balancing destination does not contain an account code, the call will fail. If FEAC is assigned to the split supervisor's line class of service, the assignment is ignored.
- Warning Tone — If the Warning Tone parameter is assigned to the selected trunk group, Warning Tone will not be returned to the calling party.

DEFINITY Generic 2.2

- Authorization Code — The calling party will not be prompted to enter an authorization code if the FRL associated with the call is not high enough to access any preference in the selected WCR pattern.
- CDR Account Code — The FEAC (Forced Entry of Account Codes) parameter can be assigned to a trunk group in a WCR pattern, to a WCR network, or to a line class of service. If FEAC is assigned to a trunk group and the Overload Balancing destination does not contain an account code, the trunk group will be skipped. If FEAC is assigned to a WCR network and the Overload Balancing destination does not contain an account code, the call will fail. If FEAC is assigned to the split supervisor's line class of service, the assignment is ignored.
- Warning Tone — If the Warning Tone parameter is assigned to the selected trunk group, Warning Tone will not be returned to the calling party.

If every accessible preference within the pattern is busy, Overload Balancing returns the interflowed call to the head of the queue.

Activating Overload Balancing

The digit strings for Overload Balancing destinations can take one of the following forms:

System 85 and DEFINITY Generic 2.1

- 1-Digit AAR Access Code + RN(X) (Location Code) + XXX(X) (Extension Number)
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + NXX (Office Code) + XXXX (Extension Number)
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + NPA (Area Code) + NXX (Office Code) + XXXX
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + International Telephone Number.

DEFINITY Generic 2.2

- 1- to 4-Digit WCR Network Access Code + ("1") (Prefix Digit) + WCR address string.

The digit strings for Overload Balancing destinations ***cannot*** be:

- An extension number within the DCS network (for DCS destinations, see "Using Call Forwarding—Follow Me as the Mechanism for Unconditional Interflow")
- 1- to 4-Digit Trunk-Group Access Code + Destination Telephone Number.

As the split supervisor activates Overload Balancing the switch thoroughly validates the digits of the destination telephone number (except for international telephone numbers). As three examples, if the digits are outside the appropriate numbering plan, if the ARS/WCR digits do not conform to the Dial "1" requirements of the switch (Procedure 275, Word 3) (Procedure 312, Word 1 beginning with DEFINITY Generic 2.2), or if an AAR/ARS/WCR pattern has not already been translated to route interflow calls to this destination, the switch returns intercept tone to deny the activation.

The split supervisor can divert Overload Balancing calls to a ***default*** destination. After establishing the default destination, the split supervisor need only dial "#" during the Overload Balancing activation to divert calls to this telephone number.

Refer to the "User Operations" section of this ACD feature description to review the step-by-step Overload Balancing operations for the split supervisor.

Overload Balancing at the Receiving ETN Node

If the AAR or WCR feature routes an interflow call to an ACD split residing in a different ETN node, the inflow threshold of the receiving split is ***not*** checked. These interflowed calls are unconditionally accepted. Also, the original priority level of the ACD call is not passed with the interflowed call, and the delay recorded announcement of the originally called split is not played for the calling party (unless the announcement was already played at the sending switch).

If an interflow call routes over an ETN trunk group (Trunk Types: 41, 42, 43, 46, and 47), the receiving switch provides the answering agent with a 3-burst zip tone to indicate an interflowed call. (The receiving switch infers that incoming ACD calls that arrive on an ETN tie-trunk group were interflowed from another ETN switch.) Beginning with Issue 3.0 of R2 V4 System 85 and DEFINITY Generic 2.1, 3-burst zip tone is no longer given for interflow calls (except Look-Ahead Interflow calls) that route over ETN trunk groups.

A city-of-origin announcement can be assigned to an incoming tie-trunk group at the receiving switch. If this announcement is combined with the 3-burst zip tone for an incoming ETN trunk group, the zip tone and the announcement will identify incoming calls from over the trunk group as interflow calls with a specific source.

Interflow calls routed over ETN trunk groups can be given priority queuing at the receiving switch by activating Overload Balancing to divert the interflow calls to the priority associated extension number at the sending switch.

Also, when automatic tie-trunk types are used, priority queuing can be assigned to these incoming tie-trunk groups at the receiving switch to give these interflowed calls preferential treatment within the ACD queue.

Using Call Forwarding — Follow Me as the Mechanism for Unconditional Interflow

Beginning with R2 V3, Call Forwarding—Follow Me can be used as another mechanism for unconditional interflow. This form of unconditional interflow diversion is provided using the Call Forwarding—Follow Me feature button or dial access code (Encode 1).

The Call Forwarding—Follow Me feature can be used to divert all ACD calls to an extension number of a remote switch within the DCS network. However, only an AUDIX or an MCS split is recommended as the remote destination for these forwarded ACD calls. If any other type of remote destination is desired (for example, a distant voice terminal, a distant attendant queue, a distant ACD split, or a public-network telephone number), the Overload Balancing function should be chosen to divert the calls.

When Call Forwading—Follow Me is used as the mechanism for interflow, the distant destination should be (for better functionality) the final destination activated.

Although this form of unconditional interflow and Overload Balancing can be activated at the same time, the Call Forwarding method of interflow does not operate in conjunction with Overload Balancing. Once Call Forwarding—Follow Me is activated to divert ACD calls outside the local switch, these calls will unconditionally forward outside the switch and can no longer be acted on by the Overload Balancing software.

Since the Call Forwarding form of interflow does not work in conjunction with Overload Balancing and since this form of interflow should be the final destination in the list of forwarding destinations, the destination should be activated as the final destination of up to three destinations (not four).

AAR, WCR, Main/Satellite, and Unconditional Call Forwarding

The AAR, WCR, or the Main / Satellite feature is used in conjunction with DCS Call Forwarding to divert ACD calls to another switch in the DCS network. Any extension number within the DCS network (preferably the associated extension number of a Message Center or AUDIX split) can be specified as a Call Forwarding destination as long as an AAR or WCR pattern (or Main / Satellite trunk group) and a DCIU link have been translated for the destination.

Using AAR or WCR With Unconditional Call Forwarding

For System 85 and DEFINITY Generic 2.1, as the Call Forwarding software prepares to send interflow calls to the AAR feature for routing these diverted calls will only be allowed to route over the ***first*** preference (trunk group) in the selected routing pattern. If every trunk in the preference is busy, the Call Forwarding software rechecks the first preference for an idle trunk every two seconds while the call ***remains*** at the head of the original split's queue. After the Call Forwarding software finds an available trunk in the first preference, the call is passed to the AAR software for digit outpulsing, DCS messaging, and subsequent routing.

For DEFINITY Generic 2.2, Call Forwarding passes the call to the WCR software for route selection, digit outpulsing, and DCS messaging.

During the AAR / WCR routing of a DCS forwarded call, the split supervisor's FRL is used by the AAR / WCR software to determine whether calls can access available facilities.

To allow interflowed calls to route without calling-party intervention, the following network routing parameters are ignored:

System 85 and DEFINITY Generic 2.1

- Authorization Code — The calling party will not be prompted to enter an authorization code if the FRL associated with the call is not high enough to access the first preference in the selected AAR pattern.
- CDR Account Code — The FEAC (Forced Entry of Account Codes) parameter can be assigned to a trunk group in an AAR pattern or to a line class of service. If FEAC is assigned to the first trunk group in the selected pattern and the forwarding destination does not contain an account code, the call will fail. If FEAC is assigned to the split supervisor's line class of service, the assignment is ignored.
- Warning Tone — If the Warning Tone parameter is assigned to the selected trunk group, Warning Tone will not be returned to the calling party.

DEFINITY Generic 2.2

- Authorization Code — The calling party will not be prompted to enter an authorization code if the FRL associated with the call is not high enough to access the first preference in the selected WCR pattern.
- CDR Account Code — The FEAC (Forced Entry of Account Codes) parameter can be assigned to a trunk group in a WCR pattern, to a WCR network, or to a line class of service. If FEAC is assigned to the first trunk group in the selected pattern and the forwarding destination does not contain an account code, the call will fail. If FEAC is assigned to a WCR network and the forwarding destination does not contain an account code, the call will fail. If FEAC is assigned to the split supervisor's line class of service, the assignment is ignored.
- Warning Tone — If the Warning Tone parameter is assigned to the selected trunk group, Warning Tone will not be returned to the calling party.

Activating Remote Call Forwarding

The digit strings for these external destinations can take the form of a 3-, 4-, or 5-digit extension number within the DCS network.

The split supervisor must have Call Forwarding—Follow Me in the class of service to activate remote Call Forwarding. If not, the switch returns intercept tone to deny the activation.

For System 85 and DEFINITY Generic 2.1, as the split supervisor activates remote Call Forwarding, the switch compares the split supervisor's FRL with the FRL of the first preference in the selected pattern. If the split supervisor's FRL is not high enough to access the first preference, the switch returns intercept tone to deny feature activation. For DEFINITY Generic 2.2, the split supervisor's FRL is one of the parameters used by the WCR software to select the best-choice route. If a route cannot be selected, the switch returns intercept tone to deny feature activation.

As the split supervisor activates remote Call Forwarding, the switch validates the digits of the destination's extension number. As two examples, if the digits are outside the appropriate numbering plan or if an AAR/WCR pattern has not already been translated to route the calls to this destination, the switch returns intercept tone to deny feature activation.

Refer to the "User Operations" section of this ACD feature description to review the step-by-step Call Forwading operations for the split supervisor.

Remote Call Forwarding at the Receiving DCS Node

If the AAR, WCR, or the Main/Satellite feature routes an interflow call to an ACD split residing in a different DCS node, the inflow threshold of the receiving split is ***not*** checked. These interflowed calls are unconditionally accepted. Also, the original priority level of the ACD call is not passed with the interflowed call, and the delay recorded announcement of the originally called split is not played for the calling party (unless it was already played at the sending switch).

If a forwarded call routes over an ETN trunk group (Trunk Types: 41, 42, 43, 44, 45, 46, and 47), the receiving switch provides the answering agent with a 3-burst zip tone to indicate an interflowed call. (The receiving switch infers that incoming ACD calls that arrive on an ETN tie-trunk group were interflowed from another ETN switch.) Beginning with Issue 3.0 of R2 V4 System 85 and DEFINITY Generic 2.1, 3-burst zip tone is no longer given for interflow calls (except Look-Ahead Interflow calls) that route over ETN trunk groups.

If a forwarded call routes over a Main/Satellite trunk group (Trunk Types: 74, 75, 77, and 78), the receiving switch provides the answering agent with 2-burst zip tone to indicate an incoming call.

A city-of-origin announcement can be assigned to an incoming ETN or Main/Satellite trunk group at the receiving switch. If this announcement is combined with the 3-burst zip tone for an incoming ETN trunk group, the zip tone and the announcement will identify incoming calls from over the trunk group as interflow calls with a specific source.

Interflow calls routed over ETN or Main/Satellite trunk groups can be given priority queuing at the receiving switch by activating Call Forwarding to divert the interflow calls to the priority associated extension number at the sending switch.

Look-Ahead Interflow

Beginning with R2 V4, Issue 1.3 and DEFINITY Generic 2, the Look-Ahead Interflow feature can be used in conjunction with ACDs that ***also*** use the Call Vectoring and the ISDN (Integrated Services Digital Network)/PRI (Primary Rate Interface) features.

With this arrangement, vector programming (at the switch that initially received an ACD call) first decides whether the interflow operation is necessary. After the need to interflow is determined, the interflow destinations are specified by one or more "route to" steps within the same vector. However, before an ACD call is redirected to the VDN of another switch within the private network, the "sending switch" queries the "receiving switch" over the D channel of the appropriate ISDN—PRI tie-trunk group. This process allows the receiving switch to decide whether it can adequately handle the redirected call. The receiving switch makes this decision according to vector programming within its own vector assigned to the same VDN specified in the "route to" step at the sending switch.

If the receiving switch can handle the interflow call, the receiving switch accepts the call with a D-channel message, and the sending switch sends the call. If the receiving switch cannot handle the call, the receiving switch refuses the call with a different message. At this time, the sending switch either queries another switch (according to subsequent "route to" steps) or executes the alternative action programmed within its own local vector.

If the Look-Ahead Interflow feature is assigned to both the sending and receiving switches, the answering agent (at the receiving switch) hears 3-burst zip tone for calls redirected by the Look-Ahead Interflow feature.

NOTE: Refer to the Look-Ahead Interflow chapter of this manual for a detailed description of this operation.

Split-Membership

The split supervisor is able to regulate membership in the supervisor's own split by adding and removing agents to/from the split. Furthermore, an agent's extension can be moved from one split to another, provided the two split supervisors coordinate the move. Before adding or removing an agent to/from a split, the agent's extension must be placed in the unstaffed mode.

Agent Override

The agent override function allows a local terminal user to enter an agent's call. Agent override is useful as a tool to observe the call-handling performance of the agents. However, the agent must be actively engaged in a call.

Agent override is activated by a dial access code. An optional warning tone is available and is provided by using a different dial access code. Using either agent override access code, an audible 2-way connection is always provided by the switch.

Service Observing

The service observing function allows a local multiappearance terminal used to monitor the work performance of an agent (who must also be equipped with a multiappearance terminal) for extended periods of time. The observer (usually, the split supervisor or a voice terminal user outside the split) is able to listen closely to the way an agent handles successive ACD calls. The SERVICE OBSERVE button is used to activate the service observing function.

Service observing can be used for training purposes. As a training device for inexperienced agents, service observing is useful in two ways. First, the new agent can be allowed to use a voice terminal equipped with a SERVICE OBSERVE button. In this way, the new agent can unobtrusively observe the way an experienced agent handles ACD calls and learn by example. Second, the split supervisor can monitor the performance of the developing agent. This enables the split supervisor to spot mistakes and refine the agent's call handling abilities.

Warning Tone

An optional warning tone is available (assigned on a per-system basis) to alert the agent to the presence of an observer in the connection. If this option is chosen, the switch returns a 2-second burnt of 440-hertz tone as the observer enters an established call and repeats the tone (for 0.5 seconds) at 15-second intervals for the duration of the observer's presence. When an observer associates with an agent's line between calls, the switch returns the first warning tone 5 seconds after the beginning of the next call.

Muting (Silent Observing) and 2-Way Observing

On a switch with the optional warning tone disabled, observers begin observation with "muting" active. In this way, when desired, silent observation can proceed normally. However, after observation has started, the observer can establish an audible 2-way connection and participate in an agent's call.

The muting function is deactivated (i.e., a 2-way connection is established) by pressing the SERVICE OBSERVE button during observation. Another press of the SERVICE OBSERVE button would restore muting for the connection.

For a switch with the warning tone ***enabled,*** an audible 2-way connection is always active during observation.

* The 7401D is an acceptable voice terminal for an observer who resides outside an ACD split. Also, ACD agents with 7401D voice terminals can be observed. However, for other reasons, 7401D voice terminals are ***not*** recommended for ACD agents (see the consideration "7401D Terminals and ACD").

ACD From the Calling Party's Perspective

Recorded Announcements for Caller

When there is a delay in answering the call, either one or two optional recorded announcements inform the calling party of the delay before the call completes. The first recorded announcement can provide a unique message for each ACD split. The second recorded announcement is system-wide and assures the caller that the call has not failed.

The timing interval preceding the first recorded announcement is set from 2 to 30 seconds (by 2-second increments). The calling party is always connected to the beginning of an announcement. Therefore, the timing between placing the call in queue and actually hearing the announcement can be as much as the timing interval chosen (from 2 to 30 seconds) plus the duration of the announcement.

The second recorded announcement is also available for each split that uses a first recorded announcement. The timing interval between the end of the first announcement and the beginning of the second announcement is set from 2 to 30 seconds (by 2-second increments). The calling party is always connected to the beginning of the announcement. Therefore, the actual timing between announcements can be as much as the timing interval chosen (from 2 to 30 seconds) plus the duration of the second announcement.

If an agent becomes available at any time before, during or after a recorded announcement, the switch removes the announcement (if necessary) and sets the call-distribution process in motion.

> **NOTE:** The switch only provides a delay announcement for incoming ACD calls when there is at least one staffed agent in the associated ACD split. The calling party continues to hear ringback. Otherwise, the delay announcement would encourage calling parties to wait when no agents are available to answer their calls.

Music-on-Hold

Music-on-Hold is also optional and may be provided to callers who are waiting in queue. A music signal provides continuous audible feedback indicating that the connection is still in effect.

When Music-on-Hold is provided for ACD queues, music is also provided (on a per-switch basis) for the Hold, Conference-Three Party, and Transfer features.

Tabular Representation of Available Choices

Table 17-A shows the calling party interfaces provided by ACD. Generally, each ACD split can use a different option. However, the Music-on-Hold option is assigned on a system-wide basis. If this option is assigned, callers hear music after the recorded announcement ends.

A split cannot be provided with the system-wide second recorded announcement unless the split is also provided with a first recorded announcement. An easy way to circumvent this problem, if desired, is to provide a first recorded announcement with similar wording to the system-wide announcement.

TABLE 17-A. Available Choices for Customer Interface

Option	Calling Party Hears*
No Announcement	Ringback
First Recorded Announcement (No music provided)	Ringback, Announcement, Silence
First and Second Recorded Announcements (No music provided)	Ringback, First Announcement, Silence, Second Announcement, Silence
First Recorded Announcement (Music provided)	Ringback, Announcement, Music
First and Second Recorded Announcement (Music provided)	Ringback, First Announcement, Music, Second Announcement, Music

* When an agent becomes available, the switch discontinues calling-party feedback (ringback, announcement, music or silence) and distributes the call to the agent.

NOTE: For R2 V4 System 85 and DEFINITY Generic 2, the Call Vectoring feature can provide a more flexible and powerful calling-party interface. Refer to the Call Vectoring feature for a description of these capabilities.

ACD From the Agent's Perspective

Zip Tone

Agents in an ACD split can receive zip tone before connecting to an ACD call or before hearing an origin announcement. A single burst of zip tone designates a call dialed directly to the agent's split (one burnt of zip tone is only provided for agents with automatic answering). Two bursts of zip tone designate a call redirected from a split within the local switch. Three bursts of zip tone designate a call which routed to the split via an ETN tie-trunk group (Trunk Types: 41, 42, 43, 46, and 47), for example, by overload balancing. The calling party does not hear zip tone. Beginning with Issue 3.0 of R2 V4 System 85 and DEFINITY Generic 2.1, 3-burst zip tone is no longer given for interflow calls (except Look-Ahead Interflow calls) that route over ETN trunk groups.

NOTE: Zip tone is not provided for non-ACD calls (calls that do not pass through a split's queue before terminating to an idle agent's voice terminal). Instead, non-ACD calls terminate to the first idle appearance on an agent's voice terminal with ringing.

City-of-Origin and Queue-of-Origin Announcements

An agent may receive either a city-of-origin or a queue-of-origin announcement. The calling party does not hear either announcement.

When implemented, the queue-of-origin announcement is heard when intraflow redirects an ACD call to an agent in another local split. When implemented, the city-of-origin announcement is heard when these announcements are assigned to incoming trunk groups. City-of-origin announcements are also heard when interflow redirects an ACD call to an agent at a distant node of the DCS or ETN network.

The city-of-origin and queue-of-origin announcements are 1.5 seconds long, run continuously, and are connected to an agent on a barge-in basis. In R2 V4, pressing an optional REPEAT button repeats the city-, or queue-of-origin announcement. (In R2 V3, pressing the optional REPEAT button only repeats the city-of-origin announcement.)

> **NOTE:** When recording "continuous" announcements (city-, and queue-of-origin), the message should be repeated for the full duration of the announcement.

Display Capabilities for Agents

The Display—Voice Terminal feature can be used by ACD agents to replace and/or supplement the city-of-origin and queue-of-origin announcements. The CALLMASTER™, 7406D With Display, 7407D, 7506, 7507, 7405D (with D401 display module), and ISDN Advantage provide display capabilities for ACD agents.

The display capabilities typically used by ACD agents areas follows.

- City-of-Origin Display

  When an ACD call forwards to a distant ACD split, the answering agent receives a visual display identifying the city of the originally called split. Also, when an ACD call arrives over an incoming trunk group, the answering agent receives a visual display identifying the source. The city-of-origin display is obtained by assigning an appropriate name (e.g., Chicago) to the incoming trunk group at the distant node.

- Queue-of-Origin Display

  When an ACD call forwards to a local ACD split, the answering agent receives a visual display identifying the originally called split. The queue-of-origin display is obtained by assigning an appropriate name (e.g., Sales Dept.) to the queue directory number or the split supervisors individual extension number for the originally called split.

- Calling Number Display

  When a call from within the DCS network is directed to an agent's individual extension number, the agent receives a visual display identifying the calling party or department. This is useful when an agent needs to correspond with other individuals or departments within the company.

- Dialed Number Display

  This function displays the dialed number when an agent places a call. This display helps to ensure accuracy in dialing.

- Elapsed Time Function

  This function displays the amount of time elapsed since the ELAPSED TIME button was pressed. The Elapsed Time function operates during calls and between calls. Therefore, an agent can use this clock as a tool for pacing the agent's handling of calls and after call work.

- Call-Duration Display on ISDN Advantage

  In addition to the display capabilities described above, the ISDN Advantage displays call-duration information for as many as four call appearances. This information is displayed for calls that terminate to or originate from a ISDN Advantage.

- Calculator Function on 7407D

  Calculator functionality is provided with the 7407D voice terminal that allows an agent to conveniently perform any necessary arithmetic computations. The calculator on the 7407D can operate during calls and between calls. Therefore, an agent can use the calculator both as a tool for quoting prices to customers and as an aid for after call work.

DNIS (Dialed Number Identification Service)

ACD agents equipped with a display voice terminal (e.g., CALLMASTER, 7405D, 7406D, 7407D, 7506, or 7507) can receive visual displays that specify the dialed number. The Call Forwarding—Follow Me feature or the Call Coverage feature (using the "All" criterion) provides the redirection needed to deliver this functionality.

> **NOTE:** When Call Vectoring is assigned to the System 85 or DEFINITY Generic 2, DNIS is administered differently. Refer to the Call Vectoring feature for a description of this DNIS configuration.

In traditional ACD arrangements, groups of agents are organized into "splits" (functional groups of answering positions). Under this approach, an agent is trained to answer calls for one specific purpose in an efficient and professional manner. However, ACD managers are recognizing the need to relax this concept of limiting each split to one call-answering task.

The alternative is to provide splits where each group of agents is proficient with several types of calls. The desired gain is to provide adequate service for the several call types with fewer agents and with less administrative intervention by the ACD manager. Using this approach, the changing staffing needs of the several call types are averaged in time, and enough agents are staffed to provide adequate service for the prevailing average load. Where 5 agents might be needed in each of 3 smaller splits (15-agent total) to handle 3 types of calls, only 11 or 12 agents might be needed in the single (more general) split.

This idea of averaging the call-handling load is sound for certain applications, but the goal of improved agent efficiency is more readily achieved with the DNIS capability. With DNIS, each answering agent knows the purpose of each incoming call as the call terminates to that agent's voice terminal. As a result, the natural efficiencies of the single split/single call type arrangement are not compromised. With the calling number display provided by DNIS, agents are aware of each call's purpose, and can answer each incoming call with the appropriate greeting. Agents need not invest time merely to determine the purpose of calls.

The following table shows sample displays that an ACD agent might receive.

TABLE 17-B. DNIS Display Information

Type of Call	Display
Inside call	a=R JONES to CLAIMS f
Outside call	a=OUTSIDE CALL to SALES f
ISDN call	a=212-281-7733 to SERVICE f

Call Forwarding—Me Configuration

A "dummy" extension (not a dummy split) is assigned in Procedure 000 (without assigning an equipment location) for each call purpose within a split. These extensions are, in turn, forward to the split's associated extension number. A calling party dials the number of the dummy extension, and the call is forwad to the split's queue.

> **NOTE:** Since Call Forwarding can be activated for the extension from an attendant console, an actual voice terminal and line circuit are not required to complete the configuration. After these Call Forwarding relationships are established from the console, a "Run Tape" operation should be performed (using Manager II, MAAP, or SMT) to make these relationships permanent.

Call Coverage (All) Configuration

A "dummy" extension (not a dummy split) is assigned in Procedure 000 (without assigning an equipment location) for each call purpose within a split. Also, these extensions are assigned to the appropriate coverage group in Procedure 000, Word 2. These extensions are, in turn, redirected using Cover All (Procedure 011, Word 1) to the split's associated extension number. A calling party dials the number of a dummy extension, and the call is redirected to the split's queue.

> **NOTE:** Since the Call Coverage configuration is an administered relationship, an actual voice terminal and line circuit are not required to complete the configuration. However, as a normal aspect of the Call Coverage feature, internal callers to the ACD split will receive coverage tone (followed by the system-wide caller response interval) before the call enters the split's queue.

Queue-Status Display

A queue-status display function is available with the ACD feature to provide agents (equipped with display voice terminals) with information about the status of the split's queue. This function is assigned to an ACD agent's class of service in Procedure 010, Word 1. When assigned, two queue-status items are automatically provided to the answering agent each time an ACD call terminates to an agent's voice terminal. These displays are also manually updated each time an agent presses the NORMAL MODE button. The queue-status items include:

- The number of calls in the split's queue

  This value (from 1 to 999*) is the current number of queued calls (at the time the ACD call begins to ring the agent). This value is right-justified in its 3-character field on the 40-character display, and when necessary, leading blanks are inserted.

- The duration of time the oldest queued call has waited.

  This value (from 001 to 999†) is the current number of seconds that the oldest queued call has waited (at the time the ACD call begins to ring the agent). This value is right-justified in its 3-character field on the 40-character display, and when necessary, leading zeros are inserted (to distinguish this field from the "number of queued calls" field).

With this information, an ACD agent would know at the beginning of every call whether to answer the call in a casual and courteous manner, to be especially efficient with the call, or to record the calling party's name and number and return the call later.

The following table shows sample queue-status displays.

TABLE 17-C. ACD Queue-Status Display Information

TYPE OF CALL	DISPLAY
Inside call	`a=R JONES to CLAIMS               5 009`
Outside call	`a=OUTSIDE CALL to SALES          25 064`
ISDN call	`a=212-281-7733 to SERVICE        17 045`

Multiple Call Handling

The multiple call handling function allows ACD agents to place calls on hold (using the HOLD button) and allow themselves to remain "available" to receive ACD calls. In software packages prior to Release 2, Version 4, an agent was not considered available to receive an ACD call unless every appearance of the agent's individual extension was idle.

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* If the number of queued calls exceeds 999, this portion of the display will show "***."

† If the oldest call has waited more than 999 seconds, this portion of the display will show "***."

Further, ACD calls were only allowed to terminate to the ***first appearance*** of an agent's voice terminal, and automatic answering could only be assigned to the first appearance.

> **NOTE:** For information about the step-by-step button presses to make multiple call handling work, refer to the "User Operations" section of this feature description.

When multiple call handling is assigned to a split, these constraints are relaxed. ACD calls can now terminate to other appearances of an agent's extension, and automatic answering can be assigned to as many as 12 "terminating/originating" appearances.

Multiple call handling (in conjunction with Principal Busy Indications) is primarily useful for Message Center agents. Commonly, these agents receive redirected calls from busy principals, and the calling party would like to hold until the principal is finished with his/her call. Multiple call handling allows these Message Center agents to handle additional ACD calls while these calling parties wait. Meanwhile, Principal Busy Indications provide these Message Center agents with the current busy/idle status of the principals' extensions.

Multiple call handling can also be used by traditional ACD agents to help "clear" the split's queue during peak periods of incoming calling. For some ACD applications, allowing ACD agents to clear the queue and to put these calls on hold provides what can be beneficial human contact with the calling parties. Once the queue is cleared, the split's agents can serve the held ACD calls one at a time.

> **NOTE:** Music-on-Hold is recommended to supplement multiple call handling as a way of clearing queues. Otherwise, the benefits of the initial human contact can be diminished during an extended period of silence for the held parties.

> **NOTE:** Since the agents who are clearing a queue are actually answering these queued calls, answer supervision (if not returned previously) is returned to the serving switch at the time an agent removes each of these calls from queue.

Refer to "User Operations" for a description of ACD agent operation using Multiple Call Handling.

Agent Convenience

Multibutton voice terminals are recommended for use with ACD. The feature buttons provide convenience and flexibility for the agents. An agent's mode of operation can be altered without interruption to callers, and the agent is freed from remembering numerous dial access codes. Furthermore, the status lamps on the terminal provide a visual reminder of the current agent mode. Refer to Table 17-D for a cross-reference of applicable feature buttons and dial access encodes.

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* The full software limit of 12 appearances can be reached using a 7507, a 7434D, or a 7405D voice terminal and a C401A or C401B coverage module. Otherwise, the limiting factor becomes the amount of 2-lamp appearance buttons provided on the specific voice terminal used

Automatic Answering

Automatic answering provides hands-free operation for ACD agents. With automatic answering assigned, incoming calls automatically connect to an agent's terminal without ringing and are automatically disconnected when the calling party disconnects. In the AUTO-IN mode, the agent is available for the next call immediately after the current call is disconnected.

> **NOTE:** Automatic answering can only be assigned to an appearance (or set of appearances) ***of one extension*** on an ACD agent's voice terminal. An automatic line appearance cannot be assigned (Procedure 052, Word 1, field 10) to an SLS (Straight Line Set) and should not be assigned to a single-appearance voice terminal.

When automatic answering is used, agents should be provided headsets. Automatic answering can be used with voice terminal handsets, but this result in inconvenient voice terminal operation. Going on-hook (hanging up the handset) after answering a call places an agent position in the unstaffed mode, and the agent must manually staff the position to receive another ACD call. To avoid this operation, the handset must be ***continuously held*** to the agent's ear.

The buttons that are available for use by ACD agents are as follows.

- AGENT SKILL

  Beginning with DEFINITY Generic 2-2, if the Expert Agent Selection feature is active, agents specify their call-handling skills by pressing this button and then entering as many as four skill codes. The AGENT SKILL button is an Abbreviated Dialing button with the Agent Skill Entry dial access code as the stored number. Refer to the Expert Agent Selection feature for more information.

- APPEARANCE BUTTONS

  Agents use these buttons to place and answer calls. Two status lamps (red and green) are adjacent to each appearance button. The red lamp lights when an agent presses the appearance button. The green lamp flashes to alert an agent to an incoming call.

  The recommended assignment of the appearance buttons for ACD agents is as follows:

  First appearance — Terminating and originating capabilities

  Second appearance — Terminating and originating capabilities

  Third appearance — Originating (only) capabilities.

  ACD calls always terminate to the first line appearance.

- ASSIST

  The ASSIST button is an Abbreviated Dialing button with the split supervisor's individual extension as the stored number. An agent uses this button to request help from the split supervisor. Pressing this button places a call from the agent to the split supervisor.

Each time an ASSIST button is pressed, the switch sends a message to the CMS so that the CMS can provide a count of assistance requests.

- AUDIO TROUBLE (Stroke Count Button, Code 0)

  The AUDIO TROUBLE button is pressed by an agent whenever there is poor transmission quality on a call (e.g., too much noise or not enough amplification). When this button is pressed, the switch sends information to the CMS to assist in the maintenance of defective facilities. This information identifies the agent's line, the trunk facility being used, and the time of day the difficulty occurred.

  Beginning with DEFINITY Generic 2.2, if the AUDIO TROUBLE (stroke count) function is assigned to a button with a status lamp, the status lamp lights for 2 seconds if a message can be sent to CMS when the button is pressed. If a message cannot be sent, the flash rate of the status lamp is set to broken flutter for 2 seconds.

- AUTO-IN

  This button is pressed when an agent wants to receive calls in the automatic mode. In the automatic mode, an agent is available to receive an ACD call immediately after disconnecting from the previous call. Activating this mode (pressing the AUTO-IN button) automatically releases all other modes. (To enter the Auto-In mode, the agent must first be in the STAFFED mode.) While active on an ACD call or between calls, an agent can change from the Auto-In mode to the Manual-In mode by pressing the MANUAL-IN button.

- AUX-WORK

  This button allows an agent to discontinue receiving ACD calls without unstaffing or logging out of the system. Often, this mode is active during breaks from the work schedule (e.g., lunch, coffee breaks, etc.). Pressing the AUX-WORK button automatically releases all other modes.

- WORK CODE

  Beginning with DEFINITY Generic 2.2, if the Call Work Codes feature is active, an agent who is measured by and logged into CMS can enter a work code during or after an ACD call. A Call Work Code is a customer-defined code such as an account code or a call activity code. This button can only be assigned to DCP voice terminals (CALLMASTER and the 7400 series are DCP voice terminals). Refer to the Call Work Codes feature for more information.

- EMERGENCY

  When an agent receives a malicious call, this button allows an agent to quickly activate a trace. An attendant traces the call. Refer to the Malicious Call Trace feature for more information.

- HOLD

  This button allows an agent to put an active call on hold. When Multiple Call Handling ***is not*** assigned to the split and a call is put on hold, the ACD feature will not distribute new calls to that agent until the held call is released or transferred. When Multiple Call Handling ***is*** assigned to the split and a call is put on hold, the ACD feature can distribute new calls to that agent.

- LWC

  This button activates the Leave Word Calling feature The ACD agents can quickly and easily leave standard messages with the Leave Word Calling feature. Pressing the LWC button leaves a "return call" message for a called party within the DCS network.

- LOGIN

  The LOGIN button is an Abreviated Dialing button with the Agent Log In access code as the stored number. When an agent presses this button, the access code to log into CMS is automatically dialed.

- LOGOUT

  The LOGOUT button is an Abbreviated Dialing button with the Agent Log Out access code as the stored number. When an agent presses this button, the access code to log out of CMS is automatically dialed.

- MANUAL-IN

  Pressing this button allows an agent to receive a single ACD call. In the Manual-In mode, an agent is considered unavailable to receive another ACD call upon disconnecting from a call. In this way, the agent can finish call-related paper work or do follow-up work before accepting another call. (To enter the Manual-In mode, the agent must first be in the STAFFED mode.) While active on an ACD call or between calls, an agent can change from the Manual-In mode to the Auto-In mode by pressing the AUTO-IN button.

- MUTE

  Pressing this button on the CALLMASTER voice terminal turns off the ACD agent's voice transmission and sidetone in the headset. (The ACD agent can still hear the calling party.) Another press of the MUTE button returns the agent's voice transmission and sidetone to the connection. Also, disconnecting the headset deactivates the muting function.

  Pressing the RELEASE button does ***not*** deactivate the muting function.

  Pressing this button on a BRI (Basic Rate Interface) terminal does ***not*** turn off voice transmission and sidetone in the headset. (The BRI MUTE button only affects transmission and sidetone in the handset.) Headsets for BRI voice terminals use headset adapters with their own muting capability. Headsets with adapters are muted by pressing and holding the QUIET button on the headset adapter.

- RELEASE

  Pressing this button quickly releases any type of call in progress at an answering position.

  > **NOTE:** The RELEASE button should not be assigned to an agent's (or observer's) voice terminal unless automatic answering (Procedure 052, Word 1) ***is also*** assigned to the voice terminal.

  Well trained agents routinely press the RELEASE button just after the closing salutation to the calling party. If the RELEASE button is not pressed, the duration of the automatic disconnect process is about 6 seconds. However, using the RELEASE button, the disconnect process completes in less than 1/10 of 1 second.

  The time savings accrued from regular use of the RELEASE button provides several advantages:

  — Reduced charges for toll trunk facilities

  — Quicker distribution of the next call to an agent in the AUTO-IN mode

  — More accurate timing of the duration of ACD calls and after-call work for an agent in the MANUAL-IN mode.

- REPEAT

  In R2 V4, pressing this button repeats the optional VDN-, city-, or queue-of-origin announcement (whichever was just heard) for an agent. In R2 V3, pressing this button only repeats the optional city-of-origin announcement for an agent.

- STAFFED

  This button is pressed by an unmeasured agent to notify the switch that the agent's voice terminal is occupied. Another press of the STAFFED button would return the unmeasured agent's position to the unoccupied mode. When the STAFFED button is pressed to staff a voice terminal, the position is placed in the AUX-WORK mode. From the AUX-WORK mode, pressing the AUTO-IN or the MANUAL-IN button allows an agent to receive an ACD call.

  For unmeasured agents using voice terminals equipped with headsets, plugging in the headset automatically places an agent in the occupied mode. Removing the headset automatically places an unmeasured agent in the unoccupied mode.

  Measured agents must log into CMS to enter the staffed mode of operation. A measured agent, who is logged in, can log out of CMS either by pressing the STAFFED button or dialing the Agent Log Out access code. Logging out of CMS places a measured agent in the unstaffed mode.

- STROKE COUNT BUTTONS (Codes 1 through 9)

  Stroke count buttons help the CMS system to monitor ACD activity. These buttons are used in conjunction with the CMS system to assist in tabulating events of current interest or concern. A stroke count button should be pressed whenever such an event occurs. When an agent who is off-hook and logged into CMS presses a stroke count button, the switch sends a message to the CMS. Upon receiving the message, the CMS increments a counter for the agent and a counter for the agent's split.

As many as ten stroke count buttons can be assigned to an agent's voice terminal. One of these buttons serves as the AUDIO TROUBLE button. The remaining nine buttons can be used as desired by the customer.

Beginning with DEFINITY Generic 2.2, if the stroke count function is assigned to a button with a status lamp, the status lamp lights for 2 seconds if a message can be sent to CMS when the button is pressed. If a message cannot be sent, the flash rate of the status lamp is set to broken flutter for 2 seconds.

The buttons that are available for use by split supervisors are as follows.

- ADD AGENT

  This button is an Abbreviated Dialing button with the Member Add access code as the stored number. When the split supervisor presses this button, the Member Add access code is automatically dialed.

- AGENT OVERRIDE

  This button is an Abbreviated Dialing button with an Agent Override access code (with or without warning tone) as the stored number. Using this button, the split supervisor or an observer can enter an agent's call to observe the agent's work performance.

- DELETE AGENT

  This button is an Abbreviated Dialing button with the Member Delete access code as the stored number. When the split supervisor presses this button, the Member Delete access code is automatically dialed.

- INTERFLO ALL

  This button is usually an Abbreviated Dialing button with the Overload Balance All access code as the stored number. When the split supervisor presses this button, the access code for unconditional distant redirection of ACD calls is automatically dialed.

  This button can also be a Call Forwading—Follow Me feature button. The split supervisor can use this button to activate unconditional distant redirection of ACD calls.

  NOTE: When Call Vectoring is used in conjunction with ACD, this button is not used.

- INTERFLO THRESHLD

  This button is an Abbreviated Dialing button with the Overload Overflow access code as the stored number. When the split supervisor presses this button, the access code for overflow distant redirection of ACD calls is automatically dialed.

  NOTE: When Call Vectoring is used in conjunction with ACD, this button is not used.

- INTRAFLO ALL

  The INTRAFLO ALL button can be a Call Forwarding—Follow Me feature button. The split supervisor uses this button to activate unconditional local redirection of ACD calls.

  This button can also be an Abbreviated Dialing button with the Call Forwarding—Follow Me access code as the stored number. When the split supervisor presses this button, the access code for unconditional local redirection of ACD calls is automatically dialed.

  NOTE: When Call Vectoring is used in conjunction with ACD, this button is not used.

- INTRAFLO THRESHOLD

  The INTRAFLO THRESHLD button can be a Call Forwarding—Busy and Don't Answer feature button. The split supervisor uses this button to activate overflow local redirection of ACD calls.

  This button can also be an Abbreviated Dialing button with the Call Forwarding—Busy and Don't Answer access code as the stored number. When the split supervisor presses this button, the access code for overflow local redirection of ACD calls is automatically dialed.

  NOTE: When Call Vectoring is used in conjunction with ACD, this button is not used.

- SERVICE OBSERVE

  This button is needed to activate service observing. Using this button, the split supervisor or an observer can monitor the work performance of an ACD agent.

  For switches with the optional warning tone disabled, pressing the SERVICE OBSERVE button (during observation) also deactivates/restores the muting function.

- VERIFY ANNCT

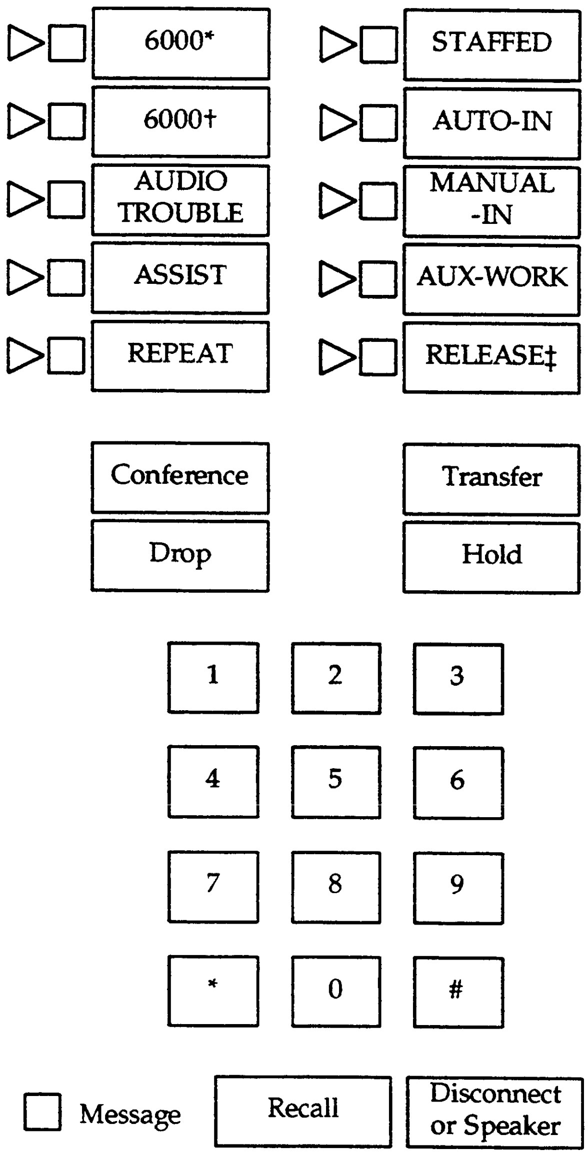
  This button is an Abbreviated Dialing button with the Announcement Verify access code as the stored number. When the split supervisor presses this button, the access code to verify the split's first recorded announcement is automatically dialed.

The recommended button layouts for ACD voice terminals are shown in Figures 17–4 through 17–1 7. These figures are provided to clarify the feature description and to assist in implementing the agents' sets.

Figures 17-4 through 17-10 show the recommended button configurations for convenient handling of ACD calls.

- Figure 17-4 shows the recommended layout for a 16-button voice terminal.
- Figure 17-5 shows the layout for a 40-button voice terminal.
- Figure 17-6 shows the layout for a CALLMASTER voice terminal.
- Figure 17-7 shows the layout for a 7407D Integrated Display Terminal.
- Figure 17-8 shows the layout for a 7406D With Display terminal.
- Figure 17-9 shows the layout for a 7506D BRI voice terminal.
- Figure 17-10 shows the layout for a 7507D BRI voice terminal.

Each of these voice terminals contains two appearance buttons for convenient answering of incoming calls. The blank buttons on the larger sets can be assigned as desired by the customer.

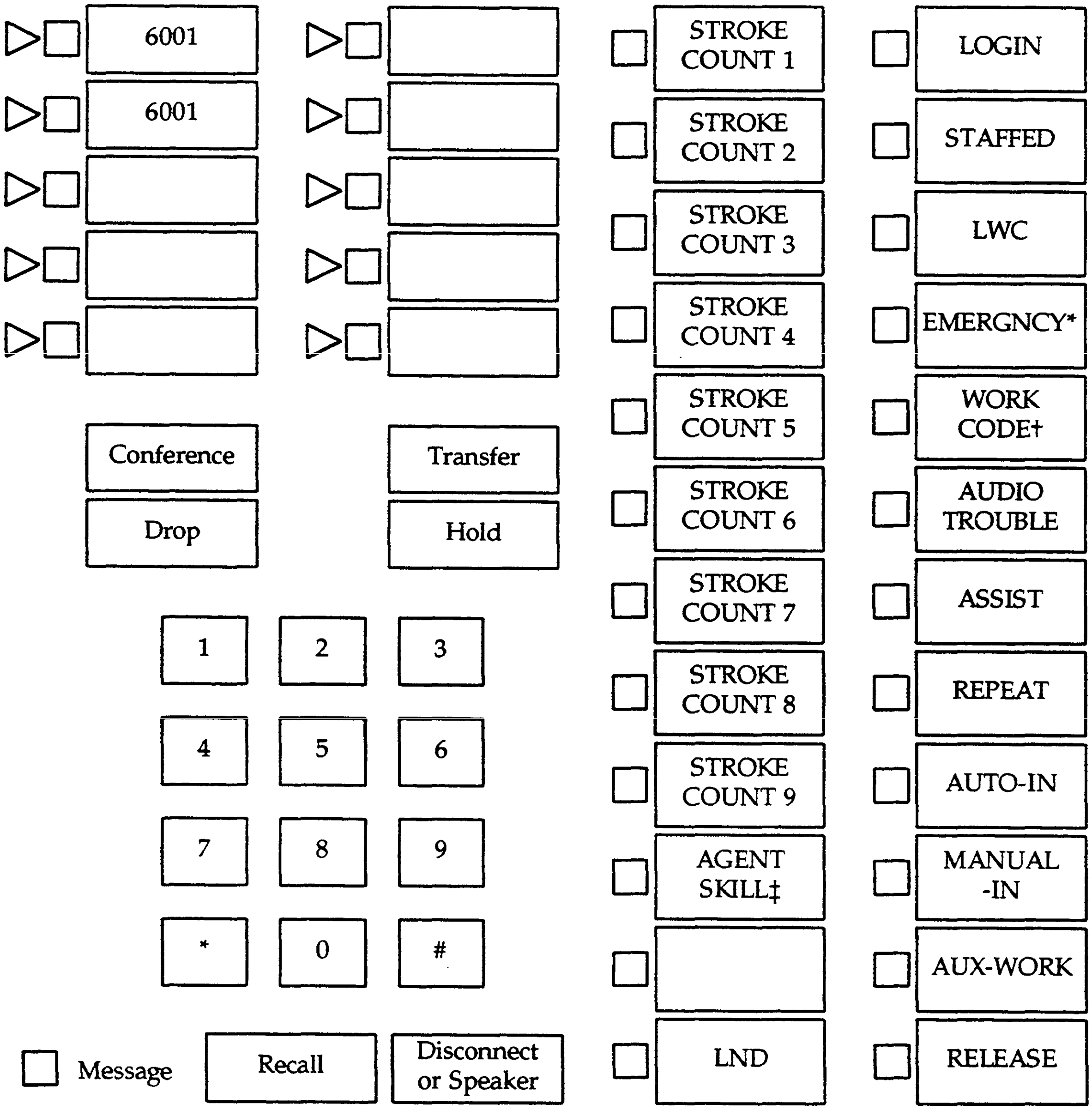


* Terminating and originating capabilities are recommended for the first appearance of an agent's voice terminal.

† Terminating originating capabilities are recommended for the second appearance of an agent's voice terminal.

‡ Whenever the RELEASE button is assigned to a voice terminal, automatic answering should also be assigned to the voice terminal.

Figure 17-4. Button Configuration for ACD Agent (16-Button Set)

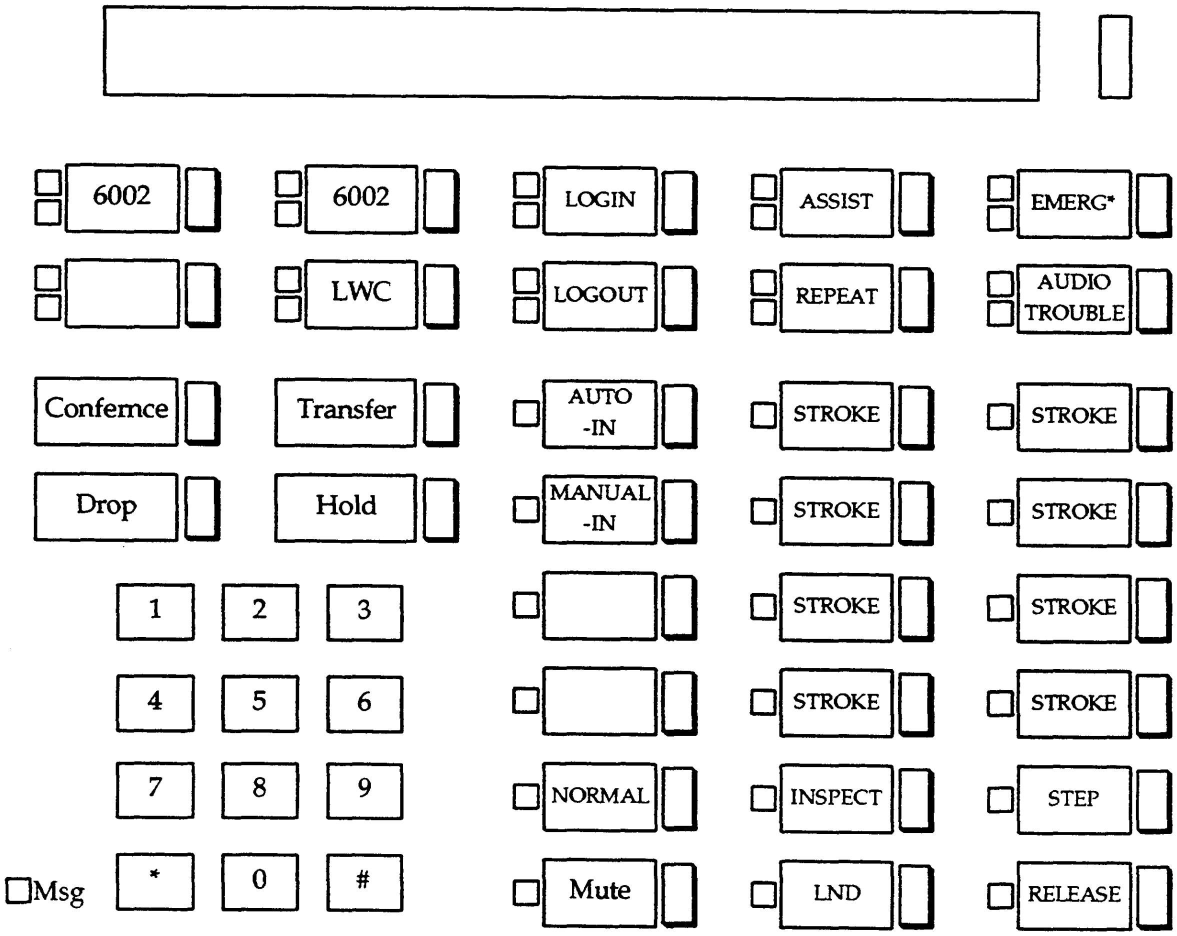


* The EMERGENCY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

† The WORK CODE button can only be assigned to DCP terminals beginning with DEFINITY Generic 2.2.

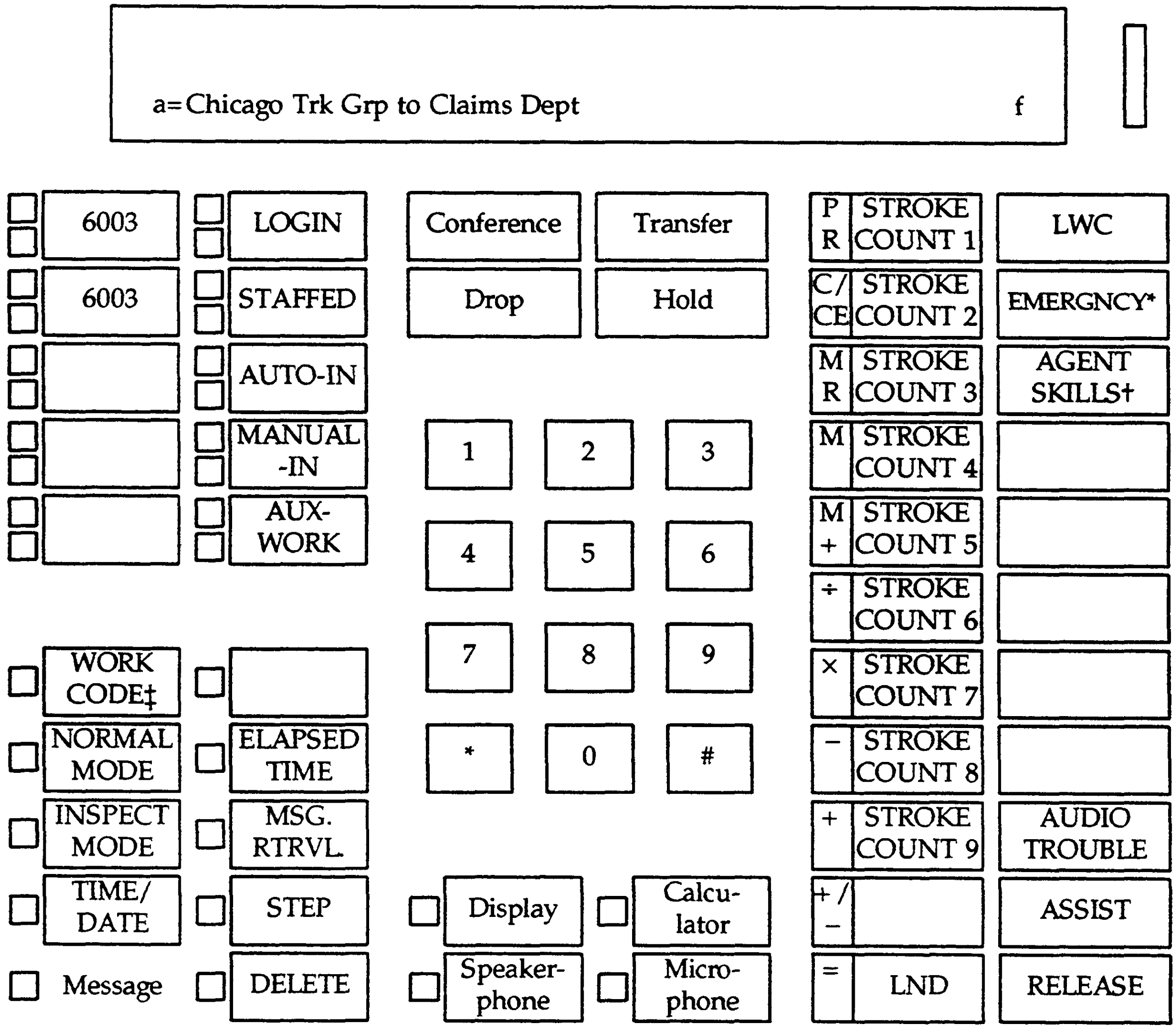
‡ The AGENT SKILLS button can only be assigned to agents' terminals beginning with DEFINITY Generic 2.2

Figure 17-5. Button Configuration for ACD Agent (40-Button Set)



* The EMERGENCY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

Figure 17-6. Button Configuration for ACD Agent (CALLMASTER Voice Terminal)

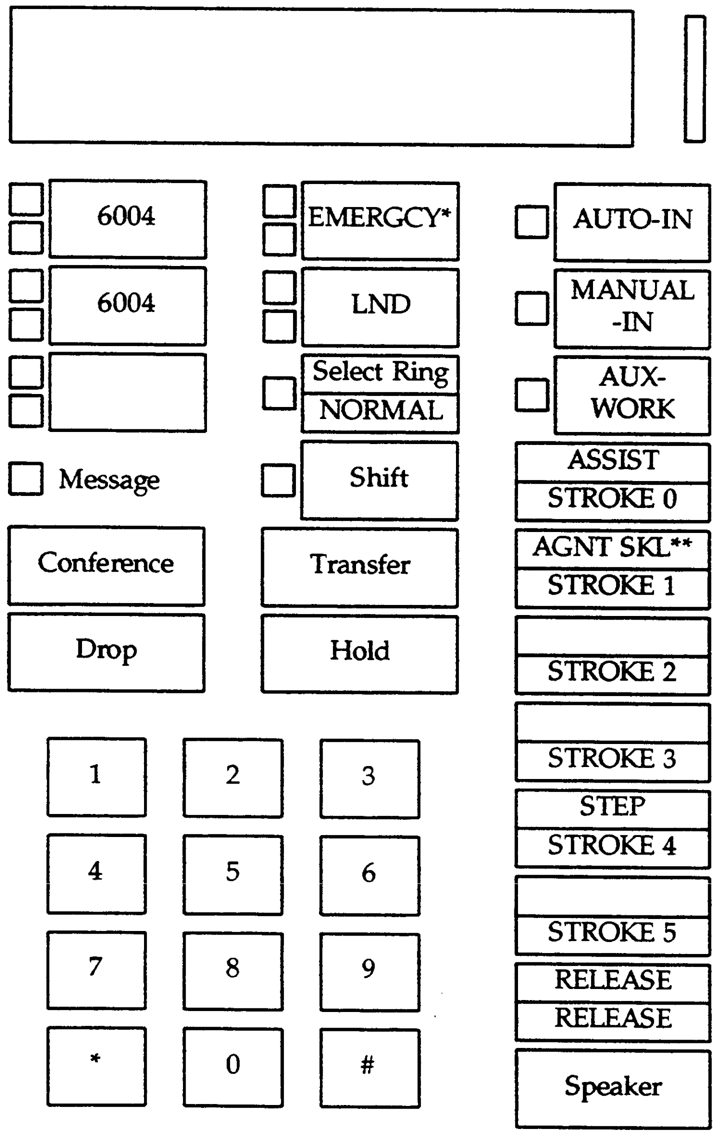


* The EMERGECY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

† The AGENT SKILLS button can only be assigned to agents' terminals beginning with DEFINITY Generic 2.2.

‡ The WORK CODE button can only be assigned to DCP terminals beginning with DEFINITY Generic 2.2.

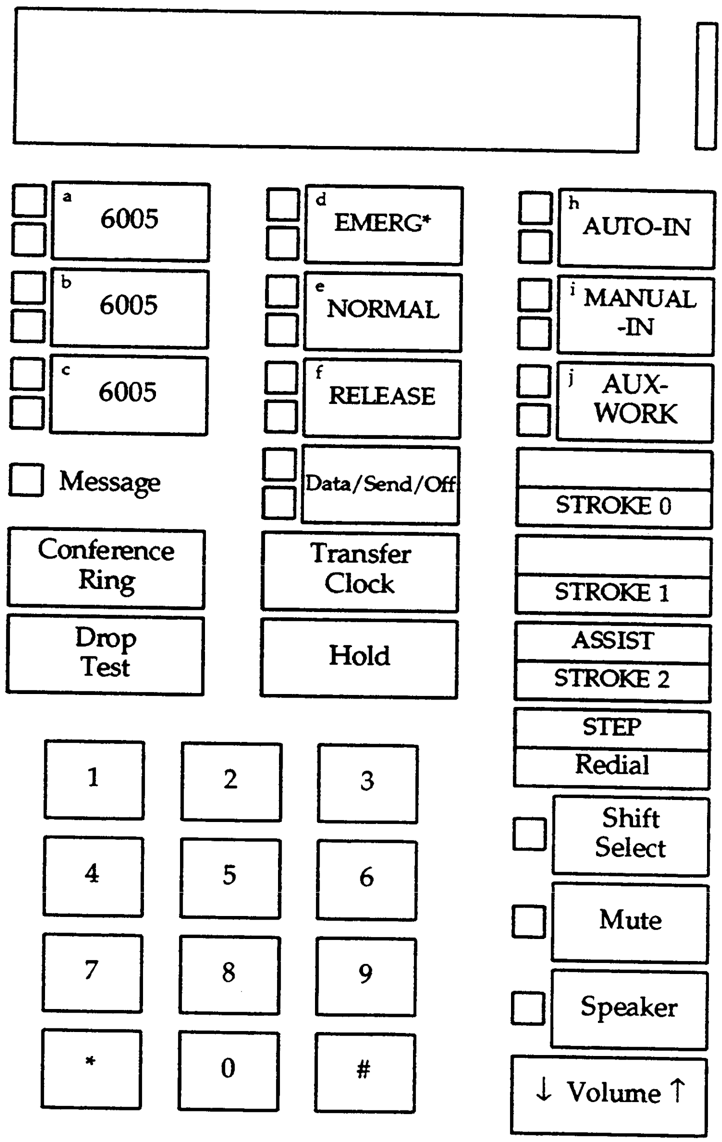
Figure 17-7. Button Configuration for ACD Agent (7407D Integrated Display Terminal)



* The EMERGENCY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

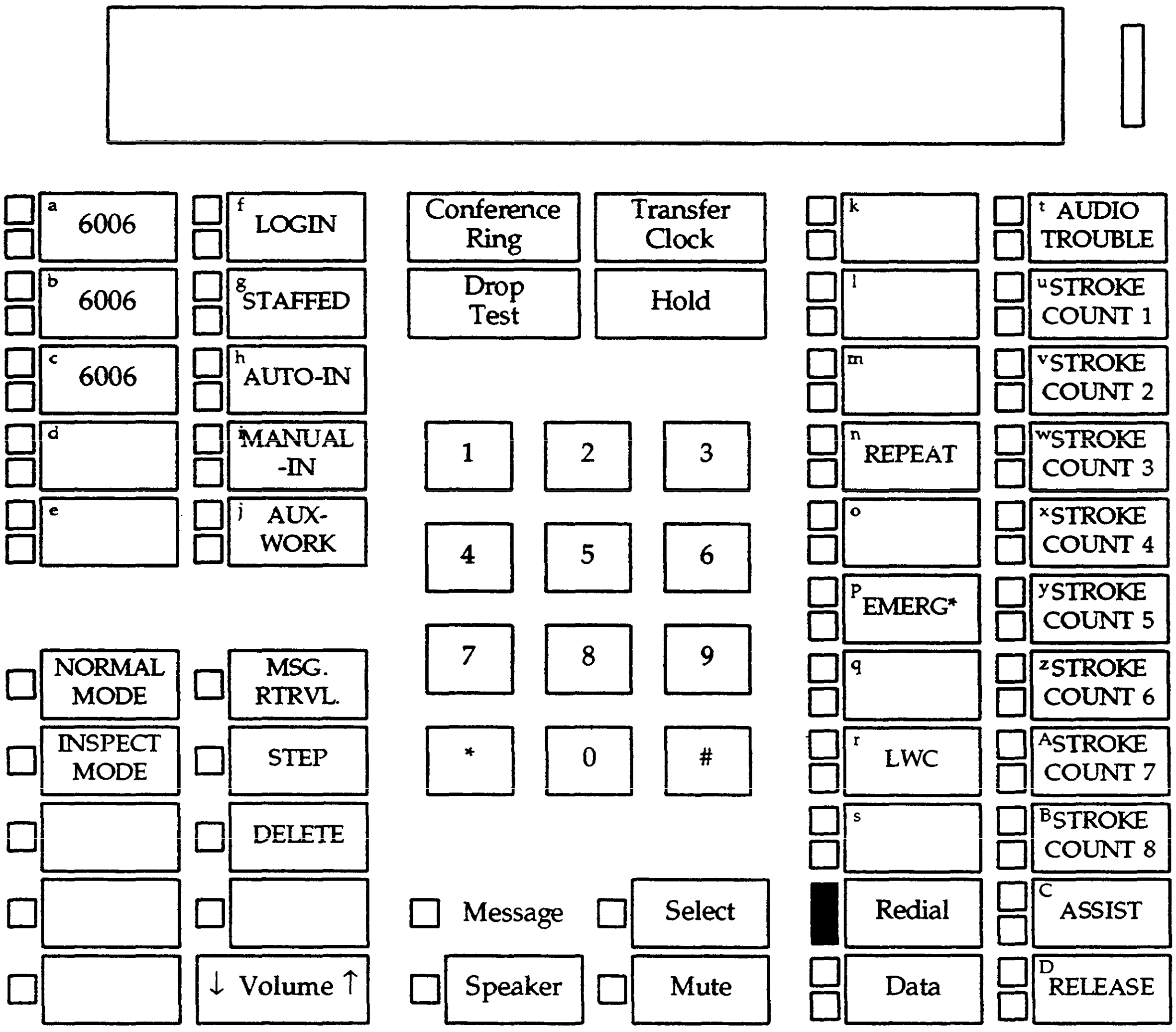
** The AGENT SKILLS button can only be assigned to agents' terminals beginning with DEFINITY Generic 2.2.

Figure 17-8. Button Configuration for ACD Agent (7406D With Display)



* The EMERGENCY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

Figure 17-9. Button Configuration for ACD Agent (7506D BRI Voice Terminal)

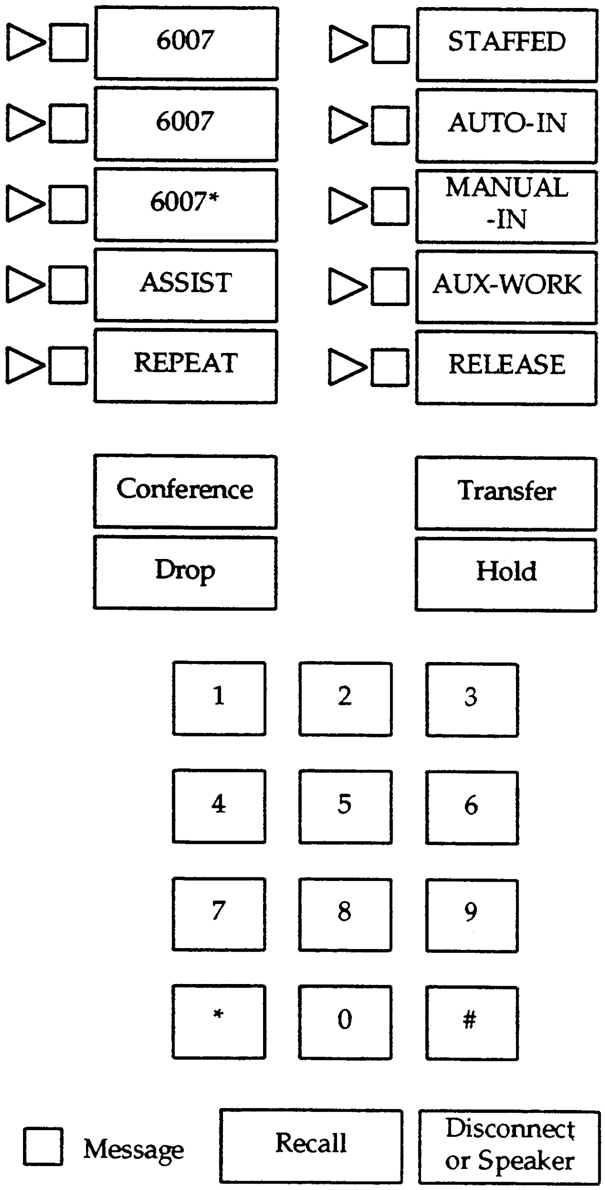


* The EMERGENCY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

Figure 17-10. Button Configuration for ACD Agent (7507D BRI Voice Terminal)

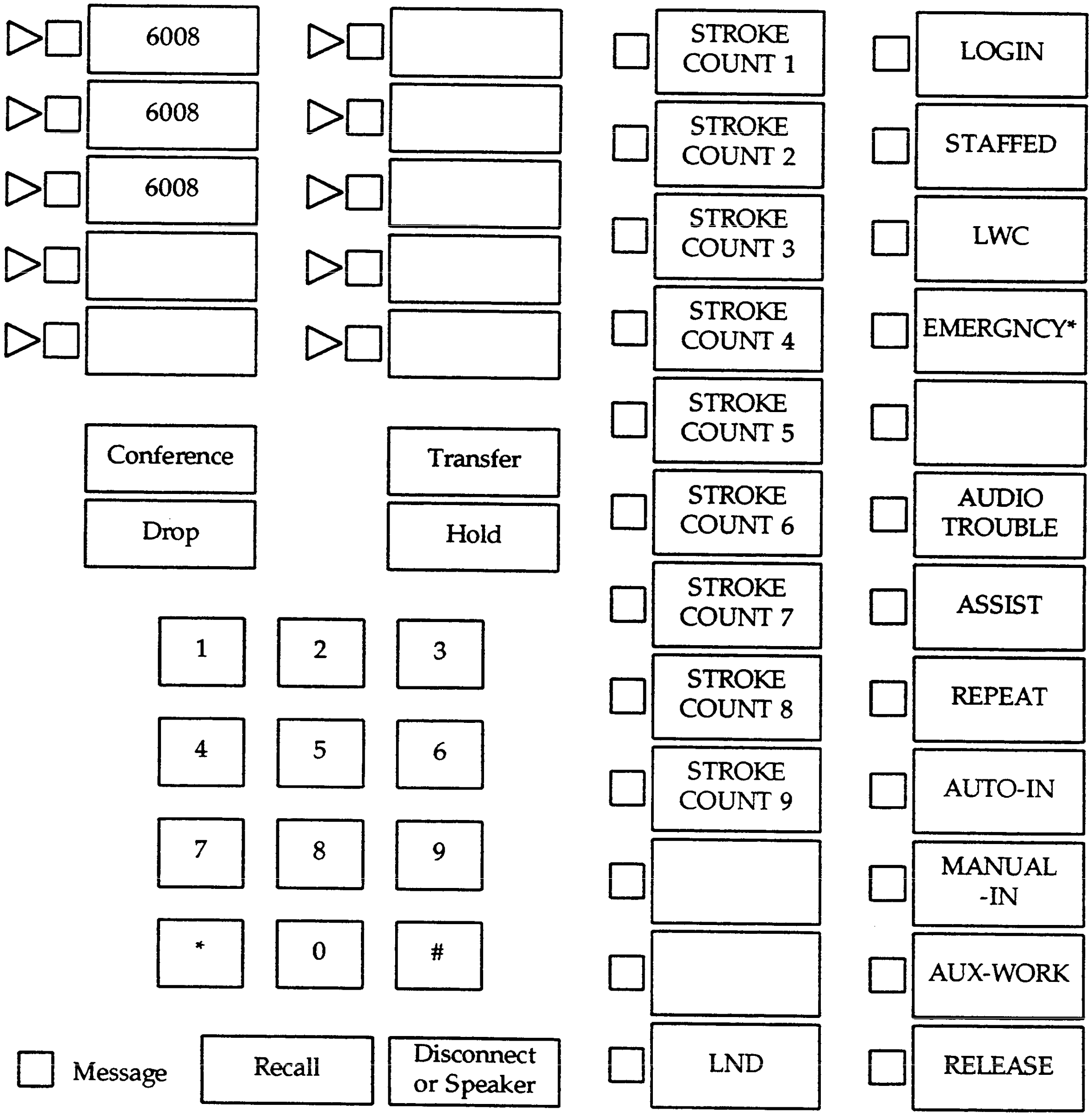
Figures 17–11 and 17–12 show the recommended button configurations for an agent who needs to receive terminal-to-terminal calls. Figure 17-11 shows the recommended layout for a 16-button voice terminal, and Figure 17-12 shows the layout for a 40-button voice terminal. Both of these terminals contain three appearance buttons. The first and second appearances provide basic ACD functionality, while the third appearance allows direct call completion to the agent's individual extension number by allowing the agent to hold calls.

The blank buttons on the 40-button set can be assigned as desired by the customer.



* Originating capabilities (only) are recommended for the third appearance of an agent's voice terminal.

Figure 17-11. Agent Who Also Receives Direct Calls (16-Button Set)

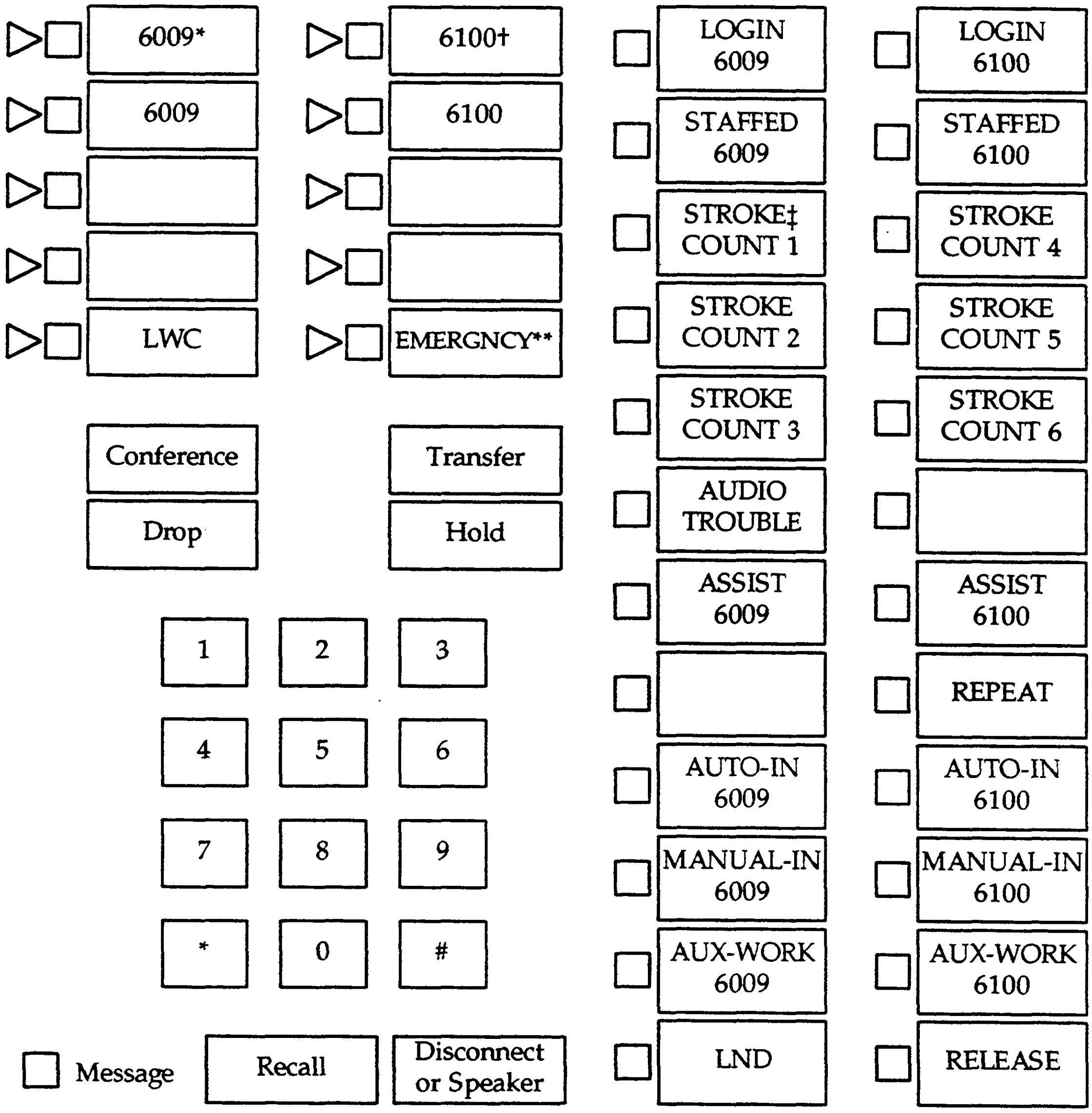


* The EMERGENCY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

Figure 17-12. Agent Who Also Receives Direct Calls (40-Button Set)

Figure 17-13 shows the recommended button configuration for an agent who handles ACD calls for two splits. Since many of the buttons are duplicated, a larger set is required for this purpose.

The blank buttons can be assigned as desired by the customer.



* Automatic answering can only be assigned to an appearance (or set of appearances) ***of one extension number*** on an agent's voice terminal.

† A System 85 ACD agent normally works in one split at a time. The two splits operate independently. If an agent is simultaneously staffed in both splits, the switch will deliver ACD calls from both queues at the same time.

‡ To assign STROKE COUNT buttons for R2 V3 ACDs, an extension must be specified in Field 8 of Procedure 054, Word 1.

** The EMERGENCY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

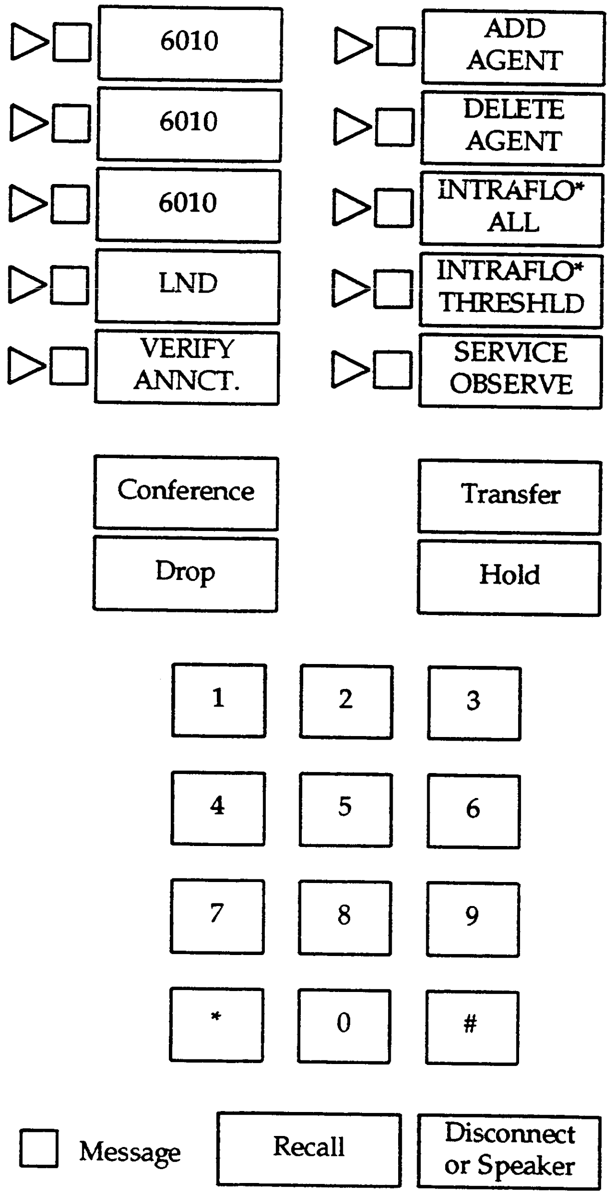
Figure 17-13. Agent Handling ACD Calls for Two Splits (40-Button Set)

Figures 17-14, 17–15, and 17–16 show the recommended button configurations for a split supervisor without agent responsibilities.

- Figure 17–14 shows the recommended layout for a 16-button voice terminal.
- Figure 17–15 shows the layout for a 40-button voice terminal.
- Figure 17–16 shows the layout for a CALLMASTER voice terminal.

All of these voice terminals contain three appearance buttons, but the third is optional.

The blank buttons on the 40-button sets can be assigned as desired by the customer.



* When Call Vectoring is assigned to the switch these buttons are not used.

Figure 17-14. Split Supervisor — Without Agent Responsibilities (16 Button)

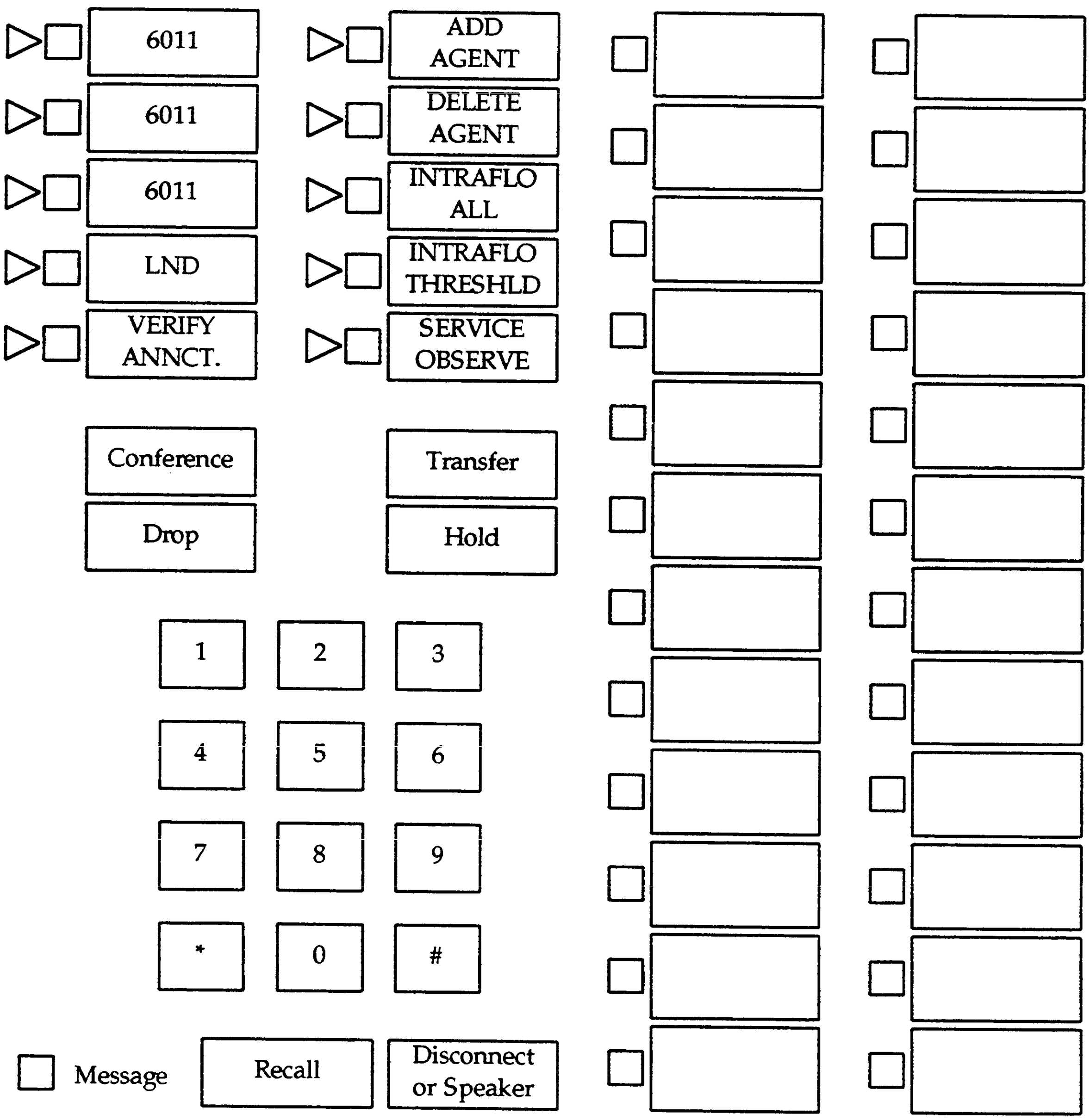


Figure 17-15. Split Supervisor — Without Agent Responsibilities (40-Button Set)

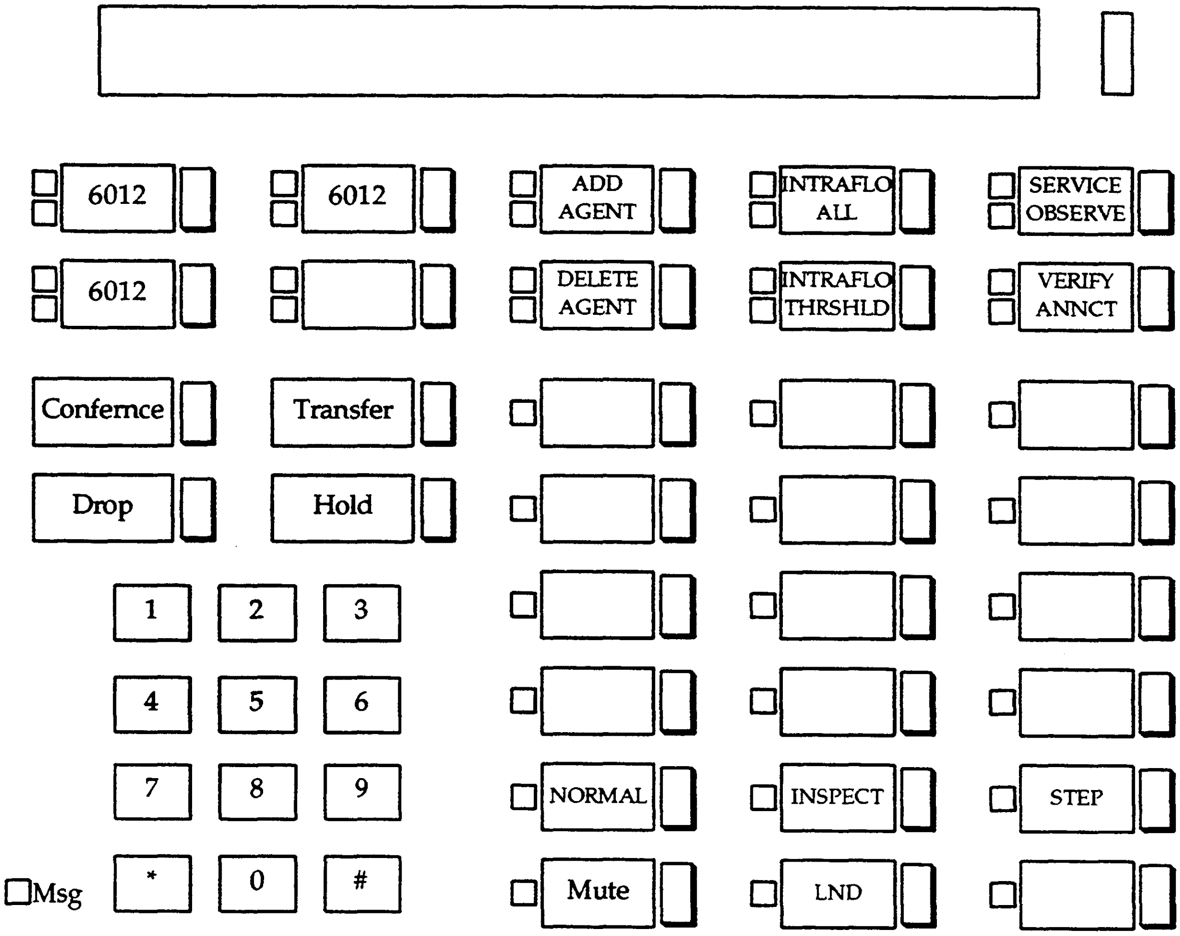
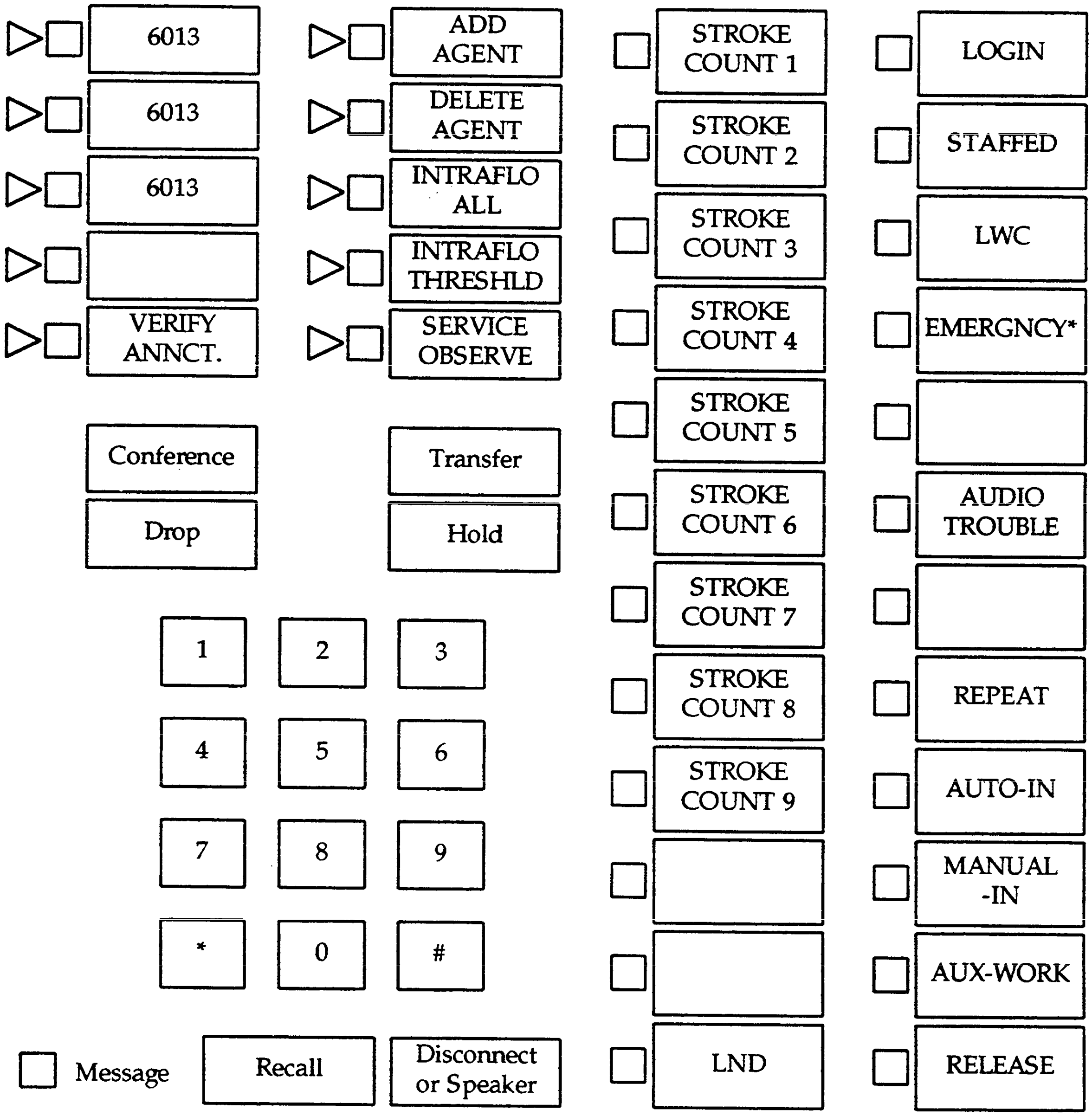


Figure 17-16. Split Supervisor — Without Agent Responsibilities (CALLMASTER Voice Terminal)

Figure 17–17 shows the recommended button cofiguration for a split supervisor with agent responsibilities. Since this set contains the buttons needed by an agent and the supervisor, a larger set is required for this purpose.

The blank buttons can be assigned as desired by the customer.



* The EMERGENCY button can only be assigned to agents' terminals beginning with R2 V4 System 85.

Figure 17-17. Split Supervisor — With Agent Responsibilities (40-Button Set)

User Operations

The following are the user operating procedures for this feature.

To Log Into CMS

A measured agent should:

1. Go off-hook on an idle appearance. [Dial tone]
2. Dial the Agent Log-In access code, or press the **[LOGIN]** button. [Dial tone]
3. Dial the 4-digit Agent Log-In ID twice (e.g., 11101110). [Confirmation tone]

   NOTE: Agent Log-In IDs are not allowed to have leading zeros. Prior to DEFINITY Generic 2.2, Log-In IDs 0 -999 are reserved for AUDIX and VRU (Voice Response Unit) applications, human agents must use Log-In IDs 1000-9999. Beginning with DEFINITY Generic 2.2, Log-In IDs 0 - 1099 are reserved for AUDIX and VRU applications, human agents must use Log-In IDs 1100-9999.

4. Press the **[RELEASE]** button or go on-hook. (The agent is now STAFFED and automatically placed in the AUX-WORK mode.)

To Log Out of CMS

A measured agent can:

1. Verify that the voice terminal is in the STAFFED mode.
2. Press the **[STAFFED]** button. (The agent is placed in the UNSTAFFED mode.)

A measured agent can also:

1. Verify that the voice terminal is in the STAFFED mode.
2. Go off-hook on an idle appearance. [Dial tone]
3. Dial the Agent Log-Out access code, or press the LOGOUT button. [Confirmation tone]
4. Press the **[RELEASE]** button or go on-hook. (The agent is placed in the UNSTAFFED mode.)

To Receive an On-Demand Update of a Queue-Status Display During an ACD Call

An ACD agent with a display and the class-of-service assignment should:

Press the **[NORMAL MODE]** button. [Display shows the number of queued calls and the amount of time the oldest call has waited.]

To Receive a New ACD Call After Placing a Call On Hold

An agent with Multiple Call Handling assigned (and a MANUAL-IN button) should:

1. Press the **[MANUAL-IN]** button. [A new ACD call terminates to a terminating/originating appearance of the agent's extension (if a call is in queue).]
2. Go off-hook on the ringing appearance. [Agent is connected to and can now attend to the new ACD call.]*
3. Press the **[HOLD]** button. [New ACD call is put on hold. (There are now two calls on hold.)]

   or

   Press the **[RELEASE]** button. [Agent is disconnected from the call.]

   or

   Transfer the new ACD call. [Agent is disconnected from the transferred call.]

   Handling choice for next call.

   Press the **[MANUAL-IN]** button again. [Another ACD call can terminate to a terminating/originating appearance (if the appearance is available).]

   or

   Select a held appearance. [The agent can now attend to a held call.]

An agent with Multiple Call Handling assigned (and an AUTO-IN button) should:

1. Press the **[AUTO-IN]** button. [A new ACD call terminates to a terminating/originating appearance of the agent's extension (if a call is in queue).]
2. Go off-hook on the ringing appearance. [Agent is connected to and can now attend to the new ACD call.]†
3. Press the **[HOLD]** button. [New ACD call is put on hold. (There are now two calls on hold.)]

   or

   Press the **[RELEASE]** button. [Agent is disconnected from the call.]

   or

   Transfer the new ACD call. [Agent is disconnected from the transferred call.]

---

* During the call, the agent can press the AUTO IN button to change to the following mode (AUTO-IN) of Multiple Call Handling.

† During the call, the agent can press the MANUAL-IN button to change to the previous mode (MANUAL-IN) of Multiple Call Handling.

Handling choice for next call.

4. If the previous call was put on hold, press the **[AUTO-IN]** button again. [Another ACD call can terminate to a terminating/originating appearance (if the appearance is available).]

   or

   If the previous call was released or transferred, simply wait for another ACD call to arrive. [Another ACD call can terminate to a terminating/originating appearance (if the appearance is available).]

   or

   Select a held appearance. [The agent can now attend to a held call.]

To Activate Call Forwarding for ACD Calls

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Call Forwarding—Activate access code (either unconditional or overflow), or press the INTERFLO ALL (if administered as Call Forwarding), INTERFLO ALL, or INTERFLO THRESHLD button. [Dial tone]
3. Dial the number of the forwarding destination. [Confirmation tone]
4. Press the **[RELEASE]** button or go on-hook. (This activation can be performed as many as three times. The order of activation sets the order of priority for forwarding. A distant destination for unconditional forwarding should be, for better functionality, the final destination activated. Also, unconditional forwarding to a distant destination cannot be paired with Overload Balancing.)

The system supervisor (using the designated attendant console) should:

1. Press an idle loop button.
2. Press the **[START]** button. [Dial tone]
3. Dial the Call Forwarding—Activate access code (either unconditional or overflow). Dial tone]
4. Dial the split supervisor's extension number (to identify the split). [Dial tone]
5. Dial the number of the forwading destination. [Confirmation tone]
6. Press the **[RELEASE]** button. (This activation can be performed as many as three times. The order of activation sets the order of priority for forwarding. A distant destination for unconditional forwarding should be, for better functionality, the final destination activated. Also, unconditional forwarding to a distant destination cannot be paid with Overload Balancing.)

To Establish a Default Destination for Overload Balancing

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Overload Balancing—Default access code. [Dial tone]
3. Dial the AAR/ARS/WCR access code. [Dial tone]
4. Dial the default 7-digit number of the private-network destination or the default 7- to 16-digit DDD number of the public-network destination. [Silence]
5. Dial **[#]** . [Confirmation tone]
6. Press the **[RELEASE]** button or go on-hook.

To Activate Overload Balancing to the Default Destination

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Overload Balancing—Activate access code (either unconditional or overflow, or press the **[INTERFLO ALL]** (if administered as Overload Balancing) or **[INTERFLO THRESHLD]** button. [Dial tone]
3. Dial **[#]** . [Confirmation tone]
4. Press the **[RELEASE]** button or go on-hook. (This destination is automatically ***inferred*** as the last priority destination for redirection of ACD calls.)

To Activate Overload Balancing to a Special Destination

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Overload Balancing—Activate access code (either unconditional or overflow), or press the **[INTERFLO ALL]** (if administered as Overload Balancing) or **[INTERFLO THRESHLD]** button. [Dial tone]
3. Dial the AAR/ARS/WCR access code. [Dial tone]
4. Dial the 7-digit number of the private-network destination or the 7- to 16-digit DDD number of the public-network destination. [Confirmation tone]
5. Press the **[RELEASE]** button or go on-hook. (This destination is ***automatically inferred*** as the last priority destination for redirection of ACD calls.)

To Add an Agent to a Split

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Member Add access code, or press the **[ADD AGENT]** button. [Dial tone]
3. Dial the extension number of the agent to be added. [Confirmation tone]
4. Press the **[RELEASE]** button or go on-hook. (The added agent is in the unstaffed mode.)

To Remove an Agent From a Split

The split supervisor should:

1. Be sure that the agent is in the unstaffed mode.
2. Press an idle appearance button. [Dial tone]
3. Dial the Member Delete access code, or press the **[DELETE AGENT]** button. [Dial tone]
4. Dial the extension number of the agent to be removed. [Confirmation tone]
5. Press the **[RELEASE]** button or go on-hook.

To Activate Agent Override With Warning Tone

An observer should:

1. Be sure the agent is active on a call.
2. Press an idle appearance button. [Dial tone]
3. Dial the agent override (warning tone) access code, or press the **[AGENT OVERRIDE]** (with warning tone) button. [Dial tone]
4. Dial the extension number of the agent to be observed.
5. After observation, press the **[RELEASE]** button or go on-hook.

To Activate Agent Override Without Warning Tone

An observer should:

1. Be sure the agent is active on a call.
2. Press an idle appearance button. [Dial tone]
3. Dial the agent override (no tone) access code, or press the **[AGENT OVERRIDE]** (without warning tone) button. [Dial tone]
4. Dial the extension number of the agent to be observed.
5. After observation, press the **[RELEASE]** button or go on-hook.

To Activate Service Observing

An observer should:

1. Press an idle appearance button. [Dial tone]
2. Press the **[SERVICE OBSERVE]** button. [Dial tone]
3. Dial the extension number of the agent to be observed [The switch returns confirmation tone and drops the regular line appearance. The Service Observe lamp either lights (without warning tone) or flashes (with warning tone. On a voice terminal with a display, the display goes blank.]
4. After observation, press the **[DISCONNECT]** button, press the **[RELEASE]** * button, or go on-hook.

To Activate 2-Way Observing During Observation

An observer should:

Press the **[SERVICE OBSERVE]** button. [Audible 2-way connection and Service Observe lamp flashes.]

To Restore Muting During Observation

An observer should:

Press the **[SERVICE OBSERVE]** button. [Silent 1-way connection, and Service Observe lamp lights steady.]

To Verify a Split's First Recorded Announcement

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Announcement Verify access code, or press the **[VERIFY ANNCT]** button.
3. Listen to the announcement.
4. Press the **[RELEASE]** button or go on-hook.

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* An observer can use the RELEASE button beginning with R2 V3, Issue 1.4 and R2 V4, Issue 1.1.

To Extend an Attendant Call to an ACD Split

A local attendant should:

1. Press the **[ANSWER]** button. [Talking connection between the attendant and the calling party]
2. Press the **[START]** button. [The switch returns dial tone and places the calling party on hold.]
3. Dial an associated extension number for the desired split. [Ringback tone]
4. Press the **[RELEASE]** button ***within 4 seconds.*** The switch places the calling party in the split's queue.]

   NOTE: If the attendant does not release within 4 seconds, the call is treated as a direct attendant call. Attendant calls to an ACD split do not enter the split's queue. Instead, the switch scans the split for an available agent, and if found, completes the call to that agent. If an available agent is not found, there are several possible switch responses. First, if the supervisor has a single-appearance voice terminal and if Attendant Call Waiting is assigned to the switch, the call waits on the split supervisor's single appearance voice terminal. Second, with a single-appearance voice terminal and without Attendant Call Waiting, the switch returns busy tone to the attendant. Third, with a multiappearance voice terminal, the call terminates to an idle appearance.

To Extend an Incoming RLT Call to an ACD Split at a Branch Location

A CAS attendant should:

1. Press the **[ANSWER]** button. [Talking connection between the attendant and the calling party]
2. Press the **[START]** button. [The switch returns dial tone and places the calling party on hold.]
3. Dial an associated extension number for the desired split at the branch. [Ringback tone]
4. Press the **[RLT RELEASE]** button ***within 4 seconds.*** [The switch places the calling party in the split's queue.]

   NOTE: If the CAS attendant does not release within 4 seconds, the call is treated as a direct attendant call. Attendant calls to an ACD split do not enter the split's queue. Instead, the switch scans the split for an available agent, and if found, completes the call to that agents. If an available agent is not found, there are several possible switch responses. First, if the supervisor has a single-appearance voice terminal and if Attendant Call Waiting is assigned to the branch location, the call waits on the split supervisor's single appearance voice terminal. Second, with a single-appearance voice terminal and without Attendant Call Waiting, the switch returns busy tone to the CAS attendant. Third, with a multiappearance voice terminal, the call terminates to an idle appearance.

An ACD agent (or other voice terminal user) should:

1. After receiving the calling party's instructions, press **[TRANSFER]** or **[CONFERENCE]** .† [Dial tone] (Calling party is placed on hold.)
2. Dial an associated extension number for the desired split at the branch. [Ringback tone]
3. When ringback is heard, press **[DISCONNECT]** ***within 4 seconds.*** [The call enters the split's queue.]

   NOTE: If an ACD agent (or other answering position) at the CAS main does not disconnect within 4 seconds, the call is treated as a direct attendant call (because the call came in on an RLT). An attendant call to an ACD or EUCD split does not enter the split's queue. Instead, System 85 or Generic 2 scans the split for an available agent, and if found, completes the call to that agent. If an available agent is not found, there are two possible switch responses. First, if Attendant Call Waiting is assigned at the branch location, the call waits on the split supervisors voice terminal. Second, without Attendant Call Waiting, the switch returns busy tone to the agent at the main.

To Extend an Incoming RLT Call to a Branch Location*

An ACD agent (or other voice terminal user) should:

1. After receiving the calling party's instructions, press **[TRANSFER]** or [CONFERENCE] .† [Dial tone] (Calling party is placed on hold.)
2. Dial the requested number.
3. If ringback (or Call Waiting ringback) is heard, press **[DISCONNECT]** , **[RELEASE]** , or Go on-hook. (The RLT is released.)

   NOTE: If the called party does not answer before the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main.

   or

   If busy tone is heard, press **[TRANSFER]** or **[CONFERENCE]** (Voice terminal user is reconnected to the calling party.) and inform the calling party that the extension is busy.

---

* For RLTs that terminate to ACD splits or VDNs, any voice terminal with a Conference or Transfer button can be used to transfer incoming RLT calls from the CAS main to a branch location.

† For incoming RLT calls, the Conference button works the same as the Transfer button That is, pressing the Conference button transfers an incoming RLT call back to the originally called branch location. Pressing the Conference button does not set up a 3-party conference.

If the calling party does not want to wait for the called party to answer, press **[DISCONNECT]** , **[RELEASE]** , or Go on-hook. (The RLT is released.)

or

If the calling party wants to wait for the called party to answer, press **[TRANSFER]** or **[CONFERENCE]** . [Dial tone] (Calling party is placed on hold at the main.)

Dial the Remote Hold dial access code. [Confirmation tone] (Calling party is placed on hold at the branch.)

Press **[DISCONNECT]** , **[RELEASE]** , or Go on-hook. [Dial tone] (The RLT is released.)

NOTE: If the calling party is a station at a branch location, Remote Hold will be denied.

NOTE: When the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main. The terminal user can attempt to complete the call again.

To Turn Off the Reload Warning Lamp

The system supervisor (using the designated attendant console) should:

1. Press an idle loop button.
2. Press the **[START]** button. [Dial tone]
3. Dial the Reload Warning Lamp access code. [The switch returns confirmation tone, and the lamp turns off.]
4. Press the **[RELEASE]** button.

A split supervisor should:

1. Go off-hook on an idle appearance. [Dial tone]
2. Dial the Reload Warning Lamp access code. [The switch returns confirmation tone, and the lamp turns off.]
3. Press the **[RELEASE}** button or go on-hook.

Considerations

Switch Capacities

Prior to DEFINITY Generic 2.2, the maximum number of ACD agents is 1024, arranged in as many as 60 splits (30 splits in R2 V3). The splits' sizes are always built in blocks of 16 positions. Therefore, a split may contain 16, 32, 48, etc., potential answering positions. As an example, an acceptable configuration using all 1024 positions spread across 20 splits would be 4 splits of 128 positions, 4 splits of 64 positions, 4 splits of 32 positions, and 8 splits of 16 positions. A processor occupancy evaluation can be performed to determine the impact of the desired configuration.

Beginning with DEFINITY Generic 2.2, the maximum number of ACD agents per system is 2048 and as many as 1023 extension numbers can be measured by CMS. Furthermore, split size restrictions are removed beginning with DEFINITY Generic 2.2. Split sizes do not have to be specified in multiples of 16. An ACD split may contain any number of agents from 0 to 1024 (1023 if the split is measured by CMS).

ACD splits are used to distribute calls to Message Center, AUDIX and ISDN Gateway. This decreases the number of ACD splits and agent positions available for traditional agent operations (for example, order taking).

An ACD split serves as the gateway to each AUDIX system. An AUDIX adjunct can have a maximum of 32 voice ports. For System 85 and DEFINITY communications systems before Generic 2.2, AUDIX gateway splits should contain 16 or 32 (usually, 32) members. Beginning with DEFINITY Generic 2.2, split size restriction have been removed and, consequently, AUDIX gateway splits may contain any number of members from 1 to 32.

System 85 and DEFINITY Generic 2 software allows as many as 64 observers (using service observing) to observe agents at the same time. When a 65th observer tries to activate service observing, the switch returns reorder tone.

As many as 60* queue status lamps (i.e., eight 30A8 SSI panels) can be provided with the switch to display the status of ACD queues.

Each 106B display unit can monitor the calling activity of up to 20 agents. As many 106Bs as needed can be provided. For example, if 1024 agents were assigned to the system, 52 display units could be provided to monitor their calling activity.

NOTE: An agent can only be monitored on one 106B display unit at a time.

* R2 V3 switches have a limit of 30.

Delay Recorded Announcements

The System 85 or DEFINITY Generic 2 only provides a delay announcement for incoming ACD calls when there is at least one staffed agent in the associated ACD split. The calling party continues to hear ringback. Otherwise, the delay announcement would encourage calling parties to wait when no agents are available to answer their calls.

Recorded Announcement Limit

Based on time-slot and TMS-blockage limitations, as many as 255 callers per module can listen to the same recorded announcement at the same time. In practice, the limit is considerably lower.

The ACD software contains a 2-second task that adds an announcement to the time slot of every calling party waiting to hear the announcement.

Legal Considerations

Laws and union contractual agreements governing the use of service observing, agent override, and CMS differ in different locations, and are subject to change. It is the responsibility of the customer's switch administrator to understand and comply with the applicable regulations.

Log-In IDs

Prior to DEFINITY Generic 2.2, Log-In IDs 0 -999 are reserved for AUDIX and VRU (Voice Response Unit) applications, human agents must use Log-In IDs 1000-9999. Beginning with DEFINITY Generic 2.2, Log-In IDs 0-1099 are reserved for AUDIX and VRU applications, human agents must use Log-In IDs 1100-9999.

CMS (Call Management System)

As many as 237 trunk groups (numbered from 18 to 255) can be measured by the CMS system. The remaining trunk groups (numbered from 256 to 999) can complete calls to ACD splits but are not measured by the CMS system.

The number of measured ACD splits is designated from the CMS system. This group of measured splits is presumed to be comprised of the lowest consecutively numbered splits assigned to the switch. As an example, CMS measurement is desired for six splits. Also, four splits without CMS measurement are desired. The six measured splits must be assigned as splits ***one*** through ***six*** on the switch. The four unmeasured splits can be assigned to any remaining split number [from 7 to 60 (30, using the AP 16)]. However, to facilitate growth, a gap between the split numbers of measured and unmeasured splits is recommended. For example, the unmeasured splits could be assigned as split numbers 15, 16, 17, and 18. In this way, the next measured split can be assigned as split number seven (as required) without "moving" an unmeasured split to make space. Beginning with DEFINITY Generic 2.2, measured splits do not have to be the lowest consecutively numbered splits assigned to the switch.

Prior to DEFINITY Generic 2.2, measured agents (extensions) in unmeasured splits are allowed. Individual extension numbers or a range of extension numbers is assigned as measured by CMS in Procedure 028, Word 1. Beginning with DEFINITY

Generic 2.2, splits (rather than extension numbers) are assigned as measured or not measured in Procedure 026, Word 2 and agents take on the attribute of the split they are assigned to. For all software versions, unmeasured agents in measured splits are not allowed.

Pegging Agent Calling Activity for the CMS

Prior to R2 V3, Issue 1.4 or R2 V4, Issue 1.1, the switch notifies the CMS whenever an agent goes off-hook on an idle appearance. This method of pegging the calling activity of ACD agents can distort the corresponding CMS statistics by reporting agent state changes, feature requests, and many invalid or misdialed numbers as completed calls.

Whenever a measured agent dials an access code (or presses an Abbreviated Dialing button with an access code as the stored number), this dialing activity is tabulated as an outgoing call by the CMS system Therefore, whenever possible, feature buttons (not Abbreviated Dialing buttons) should be assigned to the agents' multiappearance voice terminals. In this way, an agent can change states without distorting the accumulated data that describes the agents' call-origination activity.

Beginning with R2 V3, Issue 1.4 or R2 V4, Issue 1.1, the switch pegs ACD agent calls more accurately. Before pegging a call placed by an ACD agent, the switch reviews the dialed digits and, for internal calls, monitors the call's progress. In this way, the CMS will not be notified when feature access codes, many invalid numbers, and many calls resulting in busy tone or reorder tone are dialed by ACD agents.

The switch determines as closely as possible when an agent has completed a personal call. For internal and DCS calls, the agent is pegged for a personal call when the called party's voice terminal rings. For external calls (other than DCS calls), the agent is pegged for a personal call when the switch seizes an outgoing trunk.

Intraflow and Interflow

Regular Operation of Intraflow and Interflow

The call forwarding function of the ACD feature (i.e., intraflow) is used to redirect ACD calls to local destinations. The overload balancing function (i.e., interflow) is used to redirect ACD calls to distant network nodes.

There can be three local destinations for redirection of ACD calls. These destinations are arranged in a priority scheme. If the first destination is unavailable, the switch checks the second and third destinations. Assign the destinations by repeatedly activating either the Call Forwarding—Follow Me (unconditional forwarding) or the Call Forwarding—Busy and Don't Answer (threshold forwarding) feature. Cancel the destinations in reverse order by repeatedly deactivating Call Forwarding.

Additionally, there can be one internode destination for redirection of ACD calls. As mentioned, this destination is provided via overload balancing. The internode destination is always automatically implemented as the last priority. The internode destination can be used alone or after the list of local destinations. As such, there can be as many as four possible destinations.

Unconditional Internode Forwarding

The Call Forwarding—Follow Me feature can be used to redirect all ACD calls to a remote destination. However, only ***an AUDIX or an MCS split is*** recommended as the ***remote forwarding destination*** for ACD calls. If any other type of remote destination is desired (i.e., a distant voice terminal, attendant, or ACD split), the overload balancing function should be used to redirect the calls.

When Call Forwarding—Follow Me is used as the mechanism for distant forwarding the distant destination should be (for better functionality) the final destination activated.

Multiple Call Handling and CMS (Call Management System)

When multiple call handling is assigned to a split, the CMS system provides three new measurements to track the status of these agents. These measurements include:

- The number of held ACD calls
- The number of ACD calls that were abandoned while being held
- The length of time that ACD calls are held.

The CMS tracking of ACD agents is modified whenever an agent activates multiple call handling. Since the CMS system tracks agents (not calls), an agent's average call time is reduced while the agent is using multiple call handling. An agent could have more than one call on hold, but the total call-handling time would equal the time that actually elapsed.

For similar reasons, while an agent is using multiple call handling, the average talk time and the average after-call-work time are also less accurate on the real-time reports.

Multiple call handling does not reduce the accuracy of the CMS historical reports.

Published Numbers and Malicious Call Trace

For ACD applications of Malicious Call Trace, malicious calls tend to be placed by individuals outside the organizations. These people are likely to place malicious calls to numbers found in the public telephone directory. When this is the case, ACD agents answering calls directed to published numbers should be equipped with a feature button for convenient activation of Malicious Call Trace.

Power Interruptions

After a power interruption, an automatic reloading occurs of switch translations from the tape into memory. The system's reload warning lamp lights. To turn off the reload warning lamp, the system supervisor should press an idle loop button, press the START button, dial the Reload Warning Lamp access code, and press the RELEASE button.

After a switch reload, all split members are unavailable for split calls. Unmeasured agents with plugged in headsets are placed in the AUX-WORK mode. Unmeasured agents without plugged in headsets are placed in the UNSTAFFED mode and must press the STAFFED button to return to the AUX-WORK mode. Measured agents are placed in the UNSTAFFED mode and must log into CMS. From the AUX-WORK mode, an agent may press the AUTO-IN or MANUAL-IN button to receive split calls.

Music-on-Hold

For ACDs using R2 V3 (beginning with Issue 1.4) or R2 V4, it is strongly recommended that each module in the switch where Music-on-Hold is desired be equipped with a music source. A music source per module reduces the use of time slots and links between modules.

> **NOTE:** Multiple music sources are not available in R2 V3 prior to Issue 1.4.

Agent Override

Agent override allows a voice terminal user to enter an ACD agent's calls. However, the agent must have an established call in progress.

To observe successive calls using agent override, the observer must reenter the connection after each call is established.

While an observer (using agent override) is connected to an agent's active call, features such as Conference—Three Party, Transfer, Call Waiting, and Hold are denied for use by the agent.

System 85 and DEFINITY Generic 2 do not allow agent override to be activated toward a split supervisors voice terminal.

Agent override is administered on a line-class-of-service basis. An extension that is not a member of any ACD split but has Agent Override in its class of service can be used to observe an agent in any split. An extension that is a member of an ACD split and has Agent override in its class of service can only be used to observe agents in the same split.

Service Observing

To implement the service observing function, multiappearance voice terminals are required for both the agents and the observers

Service observing allows an observer to monitor an agent's incoming and outgoing calls. For some outgoing calls, however, the observer may not be added to the connection for up to 10 seconds after the agent completes dialing. For these calls, the observer is protected from listening to unattenuated touch-tones.

For some other outgoing calls involving a "cut-through" connection between the local switch and a distant switch, an observer (without a headset) may hear unattenuated touch-tones. The recommended headsets are equipped with an automatic volume control that limits these touch-tones to an acceptable volume.

An agent's calls with an attendant or a centralized attendant cannot be observed. Also, if an agent goes on-hook, presses the HOLD or RECALL button, answers a call on hold, or is put on hold by the other party, the switch removes the observer from the connection. The observer need only remain off-hook to be reconnected to the same call, if allowed, or to the next call where service observing is allowed.

Call Distribution to "Unavailable" Agents

All of the System 85 and DEFINITY Generic 2 call-distribution algorithms distribute calls to agents who are perceived as "available" by the switch. Therefore, ACD agents should ***always*** announce their unavailability to the switch by routinely entering the After-Call-Work, Aux-Work, or Unstaffed mode when unavailable. Otherwise, the switch will distribute calls to unoccupied agent positions, and these calls will not be answered.

DNIS

The DNIS function attaches names to ACD calls that use ***dial-repeating*** type routing to reach an ACD split. The switch receives the dialed digits and routes the calls to dummy extensions. In turn, these dummy extensions redirect (forward or cover) every call to the split's queue.

For DNIS, dummy splits ***should not*** be used instead of dummy extensions. Dummy splits are not needed for DNIS. When automatic muting is used to direct calls to an ACD split, these trunk groups can be directly assigned to the "real" split and given unique names. In fact, using a dummy split causes an unexpected problem. Since calls are redirected from the dummy split to the real split, the real split cannot use the Intraflow-Threshold function. (Every call arriving to the real split has already redirected once.)

CALLMASTER Voice Terminals and ACD

This is the recommended voice terminal for ACD agents. This digital multiappearance voice terminal is primarily designed for use in the ACD environment. The special attributes of this terminal for ACD agents include:

- Built-in alphanumeric display
- Two direct Starset-headset connection (no adapters needed)
- Raised feature buttons for improved tactile response
- Horizontal button layout for easier button access
- Status lamps for every button give complete visual feedback
- Fixed mute button
- Moderate price.

White Noise With CALLMASTER Voice Terminal

Prior to DEFINITY Generic 2, the CALLMASTER voice terminal, with a directly connected headset, produces white noise at a low volume. This noise is most noticeable when the ACD queue is empty and an agent is waiting for another call.

DEFINITY Generic 2 provides special DCP (digital communications protocol) S-channel messages that silence the audio connection to the switch between calls and serve to minimize this white noise.

Single-Appearance Terminals and ACD

Although ACD agents are allowed to use single-appearance voice terminals, there are some persuasive reasons why they shouldn't. The following is a summary of the shortcomings confronted when single-appearance voice terminals are used in the ACD environment.

- No status lamps to indicate the current agent state
- No method to to change the current agent state (without disrupting the active call)
- No automatic answering capability
- No RELEASE button to quickly disconnect a finished call
- No ASSIST button to quickly obtain assistance from the split supervisor
- No STROKE COUNT buttons to augment the CMS measurements
- No LND (Last Number Dialed) button to quickly redial outgoing calls
- No display modules or built-in display units (for DNIS, queue-status, city-of-origin, queue-of-origin, or LND displays)
- No service observing allowed (either ***to*** or ***from*** a single-appearance voice terminal)
- No multiple call handling allowed.

Straight Line Sets and Automatic Answering

An automatic line appearance cannot be assigned (Procedure 052, Word 1, field 10) to an SLS (Straight Line Set).

7401D Terminals and ACD

Although ACD agents are allowed to use 7401 D voice terminals, there are some persuasive reasons why they shouldn't. The following is a summary of the shortcomings confronted when 7401D voice terminals are used in the ACD environment.

- No automatic answering for a 7401D voice terminal
- No status lamps to indicate the current agent state
- Only two administrable line appearances
- Only seven administrable feature buttons
- Two button presses (instead of one) to activate a feature or to change the agent state
- No headset adapter accepted [a Starmate headset (or equivalent) must be used]
- No display modules or built-in display units (for DNIS, queue-status, city-of-origin, queue-of-origin, or LND displays)
- Limited appearances for multiple call handling
- Inconvenient multiple call handling operation.

BRI Terminals and Service Observing

When a DCP terminal is used for service observing, the line appearance (on the observer's voice terminal) that is used to establish the connection (with the agent being observed) is released once the connection is established. When a BRI terminal is used for service observing, the appearance that is used to establish the connection is not available to terminate calls until the observing session is completed. Therefore, if BRI terminals are used for service observing, an extra line appearance may be required.

Forced Entry of Account Codes

On System 85 and DEFINITY Generic 2.1, for interflowed calls, FEAC (Forced Entry of Account Codes) should not be assigned to trunk groups in an AAR/ARS pattern or to the system class of service (for ARS calls).

Beginning with DEFINITY Generic 2.2, for interflowed calls, FEAC (Forced Entry of Account Codes) should not be assigned to trunk groups in a WCR pattern or to a WCR network.

Hard and Soft Processor Swaps

Stable ACD calls will endure a hard processor swap.

If an ACD call is being distributed to an idle agent when a hard swap occurs, the call will fail.

ACD queues are stored in a status portion of switch memory. Therefore, if an ACD call is queued to a split when a hard swap occurs, the call is never routed to an idle agent, and the queue is cleared.

If an observer is using service observing to monitor an agent when a hard swap occurs, the connection for the current call will endure a hard swap. However, service observing activations are stored in a status portion of switch memory. Therefore, after the current call is finished, the observer would have to reactivate service observing to continue monitoring the agent.

If an observer is using agent override to monitor an agent when a hard swap occurs, this connection will endure a hard swap.

If an ACD agent with multiple call handling has an ACD call on hold when a hard swap occurs, the held call is lost.

An ACD call that is transferred or conference after a hard swap will not be tracked by CMS.

Call Forwarding and Overload Balancing relationships are stored in a translation portion of switch memory. Therefore, if a split supervisor activates intraflow or interflow and then a hard swap occurs, these relationships will endure the hard swap.

The ACD feature operates normally during a soft processor swap.

TABLE 17-D. ACD Function Access

Function Name	Application	Access Code-Encode	Feature Button Name	Accessed By (Note 1)
Call Forwarding—Follow Me	Unconditional local forwarding of split's calls (Intraflow—All).	1	INTRAFLO ALL	1,2
Call Forwarding—BY/DA	Overflow local forwarding of a split's calls (Intraflow—Threshold).	2	INTRAFLO THRESHLD	1,2
Call Forwarding—Cancel	Deactivates split forwarding.	3		1,2
Hold	Places call in progress on hold.	4	HOLD	2,3,4
Auto-In Mode	To receive calls in the automatic mode.	70	AUTO-IN	2,3
Aux-Work Mode	To get out of receive ACD calls mode.	71	AUX-WORK	2,3
Manual-In Mode	Enables agent position to receive a single ACD call.	72	MANUAL-IN	2,3
Staffed Mode	Indicates to switch that agent's Position is occupied.	73	STAFFED	2,3
Member Add	Adds members to a split.	74		2,5
Member Delete	Removes split members.	75		2,5
Announcement Verify	Allows verification of split's recorded announcement.	76		2
Agent Override (No Tone)	Allows entry into an agent's call in progress without tone alert	77		*2,4*
Agent override (Warning Tone)	Allows entry into an agent's call in progress with tone alert.	78		2,4
Reload Warning Lamp	Turns off reload warning lamp after a tape load on the switch.	79		1,2
Overload Balancing—All	Unconditional distant forwarding of a split's calls.	84		1,2
Overload Balancing—overflow	Overflow distant forwarding of split's calls.	85		1,2
Overload Balancing—Default	Establishes default destination for overload balancing.	86		2

Notes provided at end of table.

TABLE 17-D. ACD Function Access (Contd)

Function Name	Application	Access Code—Encode	Feature Button Name	Accessed By (Note 1)
Overload Balancing —Cancel	Deactivates either overload balancing condition.	87		1,2
Agent Log In	Logs an agent onto the CMS system.	88		2,3
Agent Log Out	Logs an agent off of the CMS system.	89		2, 3
MCT Activate	Activates a trace to identify the calling party of a malicious call.	100	EMERGENCY	2, 3
Release Call	Releases any type of call in progress.	—	RELEASE	2,3,4
Repeat announcement	Repeats city-of-origin message to agents.	—	REPEAT	2,3
Service Observing	Allows extended observation of an ACD agent's line.	—	SERVICE OBSERVE	2,3,4
Split Supervisor Assistance	Allows an agent to request help from the split supervisor.	—	ASSIST (NOTE 2)	3
Audio Difficulty	Tabulates occurrences of poor transmission quality on calls.	—	AUDIO TROUBLE	2,3
Stroke Count	Allows the CMS system to count events of current interest.	—	STROKE COUNT	2,3
Work Code	Beginning with DEFINITY Generic 2.2, if the Call Work Codes feature is active, this function allows an agent to associate a code with an ACD call.	—	WORK CODE	3
Agent Skill	Beginning with DEFINITY Generic 2.2, if the Expert Agent Selection feature is active, this function allows an agent to specify up to four call-handling skills.	107	AGENT SKILL	3

NOTE 1: 1 = System Supervisor (using the designated attendant console)
2 = Split Supervisor
3 = ACD Agent
4 = Observer
5 = Switch Administrator (using an SMT)

NOTE 2: The ASSIST button is an Abbreviated Dialing button with the split supervisor's individual extension number as the stored number.

NOTE 3: Where an encode is shown in the third column, Abbreviated Dialing buttons (using the access code as the stored number) can *always* be assigned to multi-appearance terminals.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature

ASAI Gateway Interface

Multiple call handling works normally when used with the ASAI Gateway Interface feature. However, if the hold operation is initiated from the agent's voice terminal, information about the held call is not sent to the call-center software and, therefore, the call-center software cannot coordinate the delivery of caller information for subsequent voice calls.

A City-of-Origin announcement cannot be provided for incoming trunk calls controlled by the call-center software because the incoming trunks (to the switch) do not terminate directly to an ACD split. However, the customer's call-center software could provide similar information.

Attendant Call Waiting

Attendant calls to a local ACD split are not queued. The switch attempts to complete the call to an idle agent either by scanning the agent queue or by scanning the split. If no idle agents are found in the split, the call waits on the split supervisor's single-appearance terminal. When an attendant places a call to an individual ACD agent's single-appearance terminal and that terminal is busy, the call waits on the busy terminal when Attendant Call Waiting is provided.

Attendant calls cannot wait on an ACD agent's single-appearance voice terminal while an observer (using agent override) is connected to the agent's call.

Attendant Direct Extension Selection With Busy Lamp Field

In a switch with 3- or 4-digit extensions, an attendant can use the appropriate DXS (Direct Extension Selection) buttons to place or extend calls to the associated extension number of an ACD split. However, since a split's queue is never really "busy," the BLF (Busy Lamp Field) lamps adjacent to these DXS buttons are never lit.

Attendant Display

When standard ACD interflow diverts a coverage call to the attendant queue by activating Call Forwarding (with the Attendant DAC as the destination) at the split supervisor's voice terminal], the attendant who answers the redirected call receives the usual display information on the alphanumeric display (i.e., the same display information that would have been provided for a direct call to the attendant). The called principal's identification is not provided.

Attendant Release Loop Operation

The Attendant Release Loop Operation feature does not apply to calls that an attendant extends to an ACD split. Once an attendant-extended call enters the ACD queue, this call is not timed and no reminder will be given to the attendant.

Automatic Callback

Automatic Callback can be activated toward the individual extension number of an ACD agent. However, when this is done toward an agent in a busy split (i.e., calls are waiting in queue), a race condition occurs. The switch simultaneously begins the Automatic Callback sequence and the distribution of the next queued call when the agent goes on-hook. Since the ACD call-distribution algorithms are considerably faster, the agent will usually (if not always) receive the ACD call first.

Automatic Callback can also be activated toward an associated extension number of an ACD split. When this is done, the callback sequence is performed against the split supervisor's voice terminal. Again, if the split is busy and if the split supervisor is answering ACD calls, the same race condition occurs with the same results.

> **NOTE:** Automatic Callback is usually activated toward an agent to facilitate calling from within the organization, and is always more effective when activated toward single-appearance terminals. If ACD agents (using single-appearance terminals) need to receive intracompany calls, the race condition can be avoided by using the Priority Calling feature. However, the better solution would be to provide agents with multiappearance voice terminals equipped with either two or three appearances. In this way, an extra appearance is available for routine communication with other departments.

Bearer Capability

The Bearer Capability feature has no direct impact on the ACD feature. An ACD Split does not have a BCCOS (Bearer Capability Class of Service); therefore, Bearer Capability is not checked when muting calls to an ACD Split. Agent extension numbers, however, do have BCCOSs, and Bearer Capability is checked for calls routed directly to an agent extension number or originating from an agent extension.

Busy Verification of Lines

An attendant can use the Busy Verification of Lines feature to check the busy/idle condition of the terminals in an ACD split. The agent's terminals can be checked whether or not the terminals are available for split calls. If Busy Verification of Lines is activated toward an ACD associated extension number, only the split supervisors line is verified.

Call Coverage

An ACD split (including a Message Center split or an AUDIX system) can be assigned as the final point in a coverage path. The split number, however, rather than an associated extension number, is used to designate the coverage point. Therefore, all coverage calls which are redirected to an ACD split are placed in the nonpriority queue. When Call

Vectoring is assigned to the system, an ACD split number cannot be assigned as the final point in a coverage path. However, a VDN associated with a vector that routes calls to an ACD split can be assigned as the final point in a coverage path.

On an attendant-extended call to a principal with coverage assigned to an ACD split (including a Message Center split), the attendant receives coverage tone before the call is redirected to the ACD split. During the 4-second period of silence following the tone (the caller response interval), the attendant should release the call to allow the call to be queued. Also, the attendant can release the call prior to the tone.

CDR (Call Detail Recording)

The CDR feature records the trunk-group dial access code as the calling number on an incoming ACD call. The extension number of the answering agent, rather than the split's published number, is recorded as the dialed number.

Call Forwarding—Busy and Don't Answer

The Call Forwarding—Busy and Don't Answer feature cannot be used to forward calls to an associated extension number of an ACD (including AUDIX) split or to the split supervisor's personal extension. When this is attempted, the switch returns intercept tone.

When the split supervisor activates this feature, ACD calls are forwarded to a local destination when an overflow condition occurs. The Don't Answer portion of this feature does not apply.

The split supervisor cannot use this feature to forward calls for the supervisor's individual extension number. Activation of this feature only forwards calls which are directed to the split's queue.

There can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destinations were an attendant or an ACD split ***(without*** the inflow level specified for the split), the remaining priority destinations would not be checked. Instead, a forwarded call would unconditionally enter the attendant's or the split's queue.

The forwarding destinations are assigned by repeated activation of the Call Forwarding—Busy and Don't Answer feature. Priority is determined by the order of activation. The destinations are canceled in reverse order by repeated deactivation of the feature.

When an ACD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split.

Call Forwarding—Don't Answer

An ACD split supervisor cannot activate Call Forwarding—Don't Answer to forward the supervisor's calls. When an ACD split supervisor activates Call Forwarding—Don't Answer, the split's calls are forwarded to a local destination in an overflow condition (as if

Call Forwarding—Busy and Don't Answer were instead assigned to the supervisor's class of service).

For ACD calls, there can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destination is an attendant or an ACD split (without the inflow level specified for the split), the remaining priority destinations would not be checked. Instead, a forwarded call would unconditionally enter the attendant's or the split's queue.

When an ACD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split.

Call Forwarding— Follow Me

The Call Forwarding—Follow Me feature can be used to forward all calls to an associated extension number of an ACD (including AUDIX) split. When this is done, the forwarded calls enter the split's queue.

The split supervisor cannot use this feature to forward calls for the supervisor's individual extension number. Activation of this feature only forwards calls which are directed to the split's queue.

There can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destination is an attendant or an ACD split, the remaining priority destinations are never checked. Instead, a forwarded call would unconditionally enter the attendant's or split's queue. Also, if the first or second priority destination is a DCS destination outside the local switch, the remaining priority destinations are never checked. Instead, the selected trunk group is rechecked until the call is able to forward.

The destinations are assigned by repeated activation of the Call Forwarding—Follow Me feature. Priority is determined by the order of activation. The destinations are canceled in reverse order by repeated deactivation of the feature.

When an ACD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split.

Call Forwarding—Off Net

An ACD split supervisor or system supervisor is not allowed to activate Call Forwarding—Off Net for the split. When this is attempted, the switch returns intercept tone.

Call Pickup and CMS (Call Management System)

Before DEFINITY Generic 2.2 and R2 V4 System 85, if agent A used the Call Pickup feature to answer an ACD call ringing at agent B's voice terminal (both agents are measured by CMS), the call is reported to CMS as a personal call to agent A. Beginning with DEFINITY Generic 2.2 and R2 V4 System 85, the call is reported to CMS as a call forwarded from agent B to agent A.

An agent can be active on an ACD call and elect to pickup a call directed to the agent's Call Pickup group. Using a single-appearance voice terminal, this can be done by going on-hook (to finish the picked up call) and allowing the held ACD call to ringback the agent. However, when this method is used, the callback by the held ACD call is again tabulated by CMS as an answered ACD call.

Call Vectoring

When Call Vectoring is enabled, the abandon call search function of ACD operates normally. During any step of the vector processing for incoming calls, the check for abandoned calls is performed just before ringing an idle agent The switch only distributes the call to an agent when the trunk is found active at the CO (Central Office).

When Call Vectoring is enabled, the multiple call handling function of ACD operates normally. When an agent receives a distributed call that was processed by a vector, the agent can place the call on hold and remain available to receive another call that is distributed from the split's queue.

When Call Vectoring is enabled, ACD queue warning lamps can still be provided. (The warning lamp threshold for each split is still assigned in Procedure 026, Word 1, Field 5.) However, the meaning of lamp activity on the 30A8 is slightly modified The appropriate lamp still lights when the number of calls in the split's queue is greater than or equal to the assigned threshol. With call Vectoring assigned, however, the assigned queue warning threshold ***can be a different value*** than the parameters) within the vector itself that actually divert calls to alternate destinations.

Call Waiting

Call Waiting is denied to an ACD agent while an observer (using agent override) is connected to the agent's call.

Centralized Attendant Service

Beginning with Issue 3.0 of DEFINITY Generic 2.1, an RLT (Release Link Trunk) can terminate to an ACD split, a VDN (Vector Directory Number), or an attendant console. The following CAS interactions apply to Issue 3.0 or later of DEFINITY Generic 2.1.

A centralized attendant, ACD agent, or other answering position at the CAS main can transfer (extend) an RLT call to an ACD split at a branch location. When ringback tone is heard, the answering position should release the call within 4 seconds. This allows the call to enter the split's queue.

An answering position (other than a CAS attendant) at the CAS main can receive either Call Identification Tones or zip tones,but not both. The same is true for recorded announcements. If recorded announcements (VDN-of-Origin, Queue-of-Origin, or City-of-Origin) are provided at the main, Call Identification Tones cannot be provided.

However, an answering position that is equipped with a display voice terminal can be given information about the source of an incoming call. The combined information an answering position receives from zip tones, recorded announcements, and a display voice terminal is often a suitable substitute for Call Identification Tones. For more information, refer to *ACD From the Agent's Perspective* in this feature description.

For RLTs that terminate to ACD splits, the answering positions should be equipped with display voice terminals so that the user can distinguish between RLT and non-RLT calls. The reason is that the user operation for the Conference—Three Party and Transfer features is different for RLT calls. The Conference—Three Party and Transfer features work normally for non-RLT calls. For more information, refer to the User Operations section of this feature description.

For RLTs that terminate to ACD splits, incoming calls should not be routed to a destination outside of the CAS arrangement by way of Call Forwarding. This type of muting disables the dropout and reuse capabilities that make RLTs desirable.

While an observer, using Agent Override, is connected to an agent's call (at the CAS main), the agent cannot use the Conference—Three Party, Transfer, Call Waiting, or Hold features except to transfer a call to a branch location.

Conference—Three Party

When an agent adds another agent to an ACD call, an outgoing work-related call, or a personal call, the resulting conference is not considered a work-related activity for the second agent unless the second agent was reached by dialing a QDN. If the second agent was not reached by dialing a QDN, the second agent is not removed from the agent queue.

A voice terminal user on a 2-party call is allowed to add the recipient of an ACD call to a 3-party conference. However, the second button press (of the switchhook, RECALL button, or CONFERENCE button) in the conferencing operation is ignored until the receiving agent has actually answered the call. (That is, the second button press is ignored while the call is queued.)

Beginning with Issue 3.0 of DEFINITY Generic 2.1, the operation of the Conference—Three Party feature has been changed for an ACD agent (or other voice terminal user) who transfers an incoming RLT call to a branch location in a CAS arrangement. Refer to the Centralized Attendant Service interaction and to the User Operations section of this feature description for more information.

Beginning with DEFINITY Generic 2.2, the switch notifies CMS if a measured ACD agent initiates a 3-party conference while handling an ACD call.

Display—Voice Terminal

When queue-status displays are assigned in the ACD environment, these displays use 8 characters (6 digits and 2 spaces) of the available 40 characters. Further, these 8 characters overlap with the source and destination fields on the 40-character display. Therefore, unless the source and destination names are fairly brief, these names are more likely to be truncated when queue-status displays are enabled.

DCS (Distributed Communications Systems)

In a DCS environment, direct attendant calls and attendant-extended calls to an ACD split in another node are queued. However, for attendant-extended calls, the attendant does not receive confirmation tone to indicate that the queue has been entered.

ETN (Electronic Tandem Networking)

When Overload Balancing is used to interflow ACD calls to an ACD split within a different ETN node and these calls reach the receiving switch over an ETN trunk group, the receiving switch applies distinctive 3-burst zip tone to these calls as they are distributed to an idle ACD agent. Beginning with Issue 3.0 of R2 V4 System 85 and DEFINITY Generic 2.1, 3-burst zip tone is no longer given for interflow calls (except Look-Ahead Interflow calls) that route over ETN trunk groups.

Extension Number Portability

An extension number must be removed from an ACD split, if assigned, before the number can be ported to another node.

FRL (Facilities Restriction Level)

When interflowed ACD calls use the AAR, ARS, or WCR feature to route these calls outside the local switch, the FRL of the split's supervisor is used to determine whether these calls can use the available network facilities.

Hold

If an ACD agent (including Message Center agents) with queue-status display puts an ACD call on hold, then the queue-status display is automatically updated as the agent retums to the held call for subsequent handling.

Hunting

Any individual extension number of an ACD agent may be included in a Hunting sequence. Hunting functions normally in this situation. However, an ACD split associated extension number cannot be assigned to a Hunting sequence. When this is attempted, an administration error will occur.

Interpartition Access

To protect the privacy of ACD agents, service observing is only allowed from within the same extension partition or Extension Partition 0. An observerin a different extension partition (but the same partition group) as an ACD agent cannot observe the agent using service observing.

To protect the privacy of ACD agents, agent override is only allowed from within the same extension partition or Extension Partition 0. An observer in a different extension partition (but the same partition group) as an ACD agent cannot observe the agent using agent override.

ISDN—BRI (Basic Rate Interface)

On switches prior to DEFINITY Generic 2.1, Issue 2.0, a BRI terminal cannot be used to perform the service observing function.

Beginninhg with Issue 3.0 of DEFINITY Generic 2.1, any line appearance on a BRI set can be assigned as a data line appearance. For ACD agents using BRI sets, the first line appearance should not be assigned as a data line appearance (the data line appearance is considered unavailable to receive voice calls and the agent is skipped by the call distribution algorithm).

Last Number Dialed

The LND (Last Number Dialed) feature stores and redials calls to an associated extension number of an ACD split.

ACD agents can use the LND feature to minimize redialing calls to other ACD agents, to other departments, or to other locations outside the switch.

Since the ASSIST button is an Automatic Dialing button, the LND feature does not redial assistance calls to the split supervisor. Instead, the LND feature will redial the last digits that were manually dialed before the assistance call.

Look-Ahead Interflow

The Look-Ahead Interflow feature can divert ACD calls from and interflow ACD calls to ACD splits that use any distribution algorithm: direct, circular, or most idle agent. While a call resides in a particular ACD split's queue, that queue's distribution algorithm distributes calls to an available agent. When the call is diverted to another queue, that queue's algorithm assumes control of the call's distribution.

At a receiving switch, the Look-Ahead Inter-flow feature and the Abandon Call Search function of the ACD feature may not be compatible. Look-Ahead Interflow calls arrive at the receiving switch over ISDN—PRI facilities (that provide Positive Immediate Disconnect), and the receiving switch does not verify whether the calling party has abandoned before distributing a queued interflow call to an available agent.

However, for Look-Ahead Interflow calls, the incoming trunk facility to the sending switch ***may or may not*** be an ISDN facility with Positive Immediate Disconnect. If the incoming trunk group at the sending switch is not ISDN, the receiving switch could distribute a "ghost call" to the answering agent.

If an ACD agent becomes available at a sending (or tandeming) switch during ISDN messaging with the receiving switch, the agent at the sending switch is given precedence to answer the call. However, ISDN messaging is gracefully exited before the ACD agent at the sending switch receives the call.

Since a Look-Ahead Interflow call always contains the Look-Ahead IE as part of the Call Setup message, the receiving switch can always identify the incoming call as an "interflowed" call. Therefore, for Look-Ahead Interflow calls that are queued to an ACD split at the receiving switch, the receiving switch always provides 3-burst zip tone to the answering agent.

The receiving switch for a Look-Ahead Interflow call can deliver the equivalent of a city- or queue-of-origin announcement to the answering ACD agent. At the receiving switch, multiple VDNs can access the same receiving vector. Therefore, separate VDNs with different corresponding announcements (administered in Procedure 033, Word 1) can be assigned for each sending switch or sending vector that interflows calls to the receiving vector.

At a receiving switch, the Look-Ahead Interflow feature and the Queue-Status Display function of the ACD feature are compatible. An ACD agent (with a display set and Queue-Status Display assigned to the class of service) at the receiving switch will normally receive the queue-status information for the local ACD split with incoming Look-Ahead Interflow calls. Table 17-E is a sample display with queue-status information.

TABLE 17-E. Look-Ahead Interflow Queue-Status Display Information

Type of Call	Display
ISDN call	a=212-281-7733 to DETROIT CLAIMS 17 045

However, within this Queue-Status Display, the length of time that the oldest call has waited (in this example, 45 seconds) does ***not*** include the amount of time that an interflowed call may have already waited at the sending switch.

At a receiving switch, the Look-Ahead Interflow feature and the Queue-Warning Lamp Control option of the ACD feature are compatible. When a Look-Ahead Interflow call enters an ACD queue at a receiving switch, the Queue-Warning Lamp software recognizes the interflowed call and includes this call in the threshold total for lighting the corresponding lamp on the 30A8 System Status Indicator.

At a receiving switch, the Look-Ahead Interflow feature and the Service Observing function of the ACD feature are compatible. A local voice terminal user (with a SERVICE OBSERVE button) at the receiving switch can normally monitor Look-Ahead Interflow calls that are answered by ACD agents (with multiappearance voice terminals) at the receiving switch.

At a receiving switch, the Look-Ahead Interflow feature and the Agent Override function of the ACD feature are compatible. A voice terminal user (with Agent Override in the class of service) at the receiving switch can normally monitor Look-Ahead Interflow calls that are answered by ACD agents (except split supervisors) at the receiving switch.

At a receiving switch, the Look-Ahead Interflow feature and the Multiple Call Handling function of the ACD feature are compatible. An ACD agent (with a HOLD button and multiple call handling assigned to the agent's split) can put a Look-Ahead Interflow call on hold and become available to receive another ACD call from the split's queue. Conversely, the ACD agent can put a regular ACD call on hold and become available to receive a Look-Ahead Interflow call.

At a sending (or tandeming) switch, an ACD split that is assigned as Automatic Available (Procedure 026, Word 2) can outflow Look-Ahead Interflow calls to/from the other switch. Likewise, at a receiving switch, an Automatic Available split can inflow Look-Ahead Interflow calls from a sending switch.

Malicious Call Trace

If an ACD agent receives a malicious call, the Malicious Call Trace feature can be activated by the ACD agent. It is preferable, when malicious calls are more than a rare occurrence, to assign the EMERGENCY button to every agent's multiappearance voice terminal.

If an ACD agent receives a malicious call and activates the Malicious Call Trace feature while the agent is being observed using Service Observing (as opposed to Agent Override), the observer may continue to monitor the malicious call and subsequent calls normally. Furthermore, Service Observing allows the observer to ***begin observation*** after Malicious Call Trace has been activated.

City/Queue-of-Origin Announcement. An ACD agent cannot activate Malicious Call Trace during a city-of-origin or a queue-of-origin announcement. When this 1.5-second announcement finishes, a trace can be activated.

CMS (Call Management System). With CMS, messages are sent to the CMS processor and interpreted as EXCEPTION messages by the CMS software. For Release 2 CMS, an exception message will appear on any real-time report displaying data for that split. For Release 3 CMS, exception messages appear in the real-time exceptions log.

Agent Override. If an Agent Override call is attempted toward a line involved in a Malicious Call Trace, the activating party receives intercept tone.

Message Center Service

One Message Center split can use intraflow to divert excess calls to another Message Center split. However, when this is done, it is recommended that both splits be assigned to the same adjunct processor. If the MCS splits are assigned to different adjunct processors, the diverted calls will still enter the backup split's queue, and these calls will be distributed to available MCS agents. But, the Coverage Screen will not be displayed for the answering agent in the backup split.

Music-on-Hold Access

For ACDs using R2 V3 (beginning with Issue 1.4) or R2 V4, it is strongly recommended that each module in the switch where Music-on-Hold is desired be equipped with a music source. A music source per module reduces the use of time slots and links between modules.

NOTE: Multiple music sources are not available in R2 V3 prior to Issue 1.4.

Override

When Override is activated toward the individual extension number of a busy ACD agent, the override call intrudes into the agent's active call. When Override is activated toward the individual extension number of an idle ACD agent, the override call alerts the agent with 3-burst ringing.

When Override is activated toward an associated extension number of an ACD split and the split supervisor is busy, the call intrudes into the split supervisor's active call. When the split supervisor is idle, the call alerts the supervisor with 3-burst ringing. (The Override call does not enter the split's queue.)

NOTE: If the called agent (or split supervisor) is using a multiappearance terminal, an override call (in preference to intruding into the active call) will terminate to an idle appearance (if available) with 3-burst ringing. When no idle appearances are available, the override call will intrude into the active call.

Precedence Calling

The precedence level of AUTOVON calls that are directed to or forwarded to an ACD split are checked. If the precedence level is higher than "Routine," the call does not enter the split's queue. Instead, the call is redirected to the attendant queue.

Priority Calling

When Priority Calling is activated toward an individual single-appearance terminal in an ACD split, the call waits on that terminal. When Priority Calling is activated toward an associated extension number of an ACD split, the call always waits on the split supervisor's single-appearance terminal. (The priority call does not enter the split's queue.)

NOTE: If the called agent (or split supervisor) is using a multiappearance terminal, a priority call will terminate to an idle appearance (if available) with 3-burst ringing. If no idle appearances are available, the switch returns busy tone to the calling party.

Priority calls cannot wait on an ACD agent's single-appearance voice terminal while an observer (using agent override) is connected to the agent's call.

Tenant Services

Automatic-in type routing to ACD splits is not partitioned. There are no checks in Procedure 115 to ensure that the partition of an automatic-in type trunk group matches the partition of the assigned split. It is the responsibility of the system manager to ensure that these partition numbers match.

Dial-repeating type muting to ACD splits is partitioned. When the digits of an associated extension number are either passed to System 85/DEFINITY Generic 2 from the serving switch or dialed from inside the switch, the Tenant Services feature makes the necessary partitioning checks. Associated extension numbers are assigned to partitions using Procedure 000, Word 4.

There are no checks made in switch administration to ensure partitioning of ACD splits. When administered from the Manager II, MAAP or SMT, a single split may contain agents from several extension partitions. Once an ACD call enters the split's queue, the call will terminate to the selected available agent regardless of the agent's extension partition assignment.

In general practice, each ACD split provides a functional division of call answering responsibilities. For most (if not all) ACD applications, it is recommended that each split contain only agents assigned to the same extension partition. The System Manager should be careful to follow this guideline.

The ACD feature also allows a split supervisor (in a partition other than Extension Partition 0) to add and remove agents to/from the supervisor's ACD split. When this method for adding agents is used, the Tenant Services feature provides a check to ensure that the added agent belongs either to the supervisors extension partition or to Extension Partition 0. If not, the switch returns intercept treatment to the split supervisor.

A split supervisor in Extension Partition 0 is allowed to add an agent to or remove an agent from their ACD split regardless of the agent's extension partition.

An observer (in a partition other than Extension Partition 0) is allowed to use Agent Override to monitor the call-handling activity of an agent in the same extension partition or in Extension Partition 0. When the observer attempts to activate Agent Override toward an agent in any other partition, the switch returns intercept treatment to the observer.

An observer in Extension Partition 0 is allowed to use Agent Override to observe any agent in the switch.

An observer (in a partition other than Extension Partition 0) is allowed to use Service Observing to monitor the call-handling activity of an agent in the same extension partition or in Extension Partition 0. When the observer attempts to activate Service Observing toward an agent in any other partition, the switch returns intercept treatment to the observer.

An observer in Extension Partition 0 is allowed to use Service Observing to observe any agent in the switch.

Timed Reminder

The Timed Reminder feature does not apply to calls that an attendant extends to an ACD split. Once an attendant-extended call enters the ACD queue, this call is not timed and no reminder will be given to the attendant.

Transfer

When an agent adds another agent to an ACD call, an outgoing work-related call, or a personal call, the transferred call is not considered a work-related activity for the second agent unless the second agent was reached by dialing a QDN (or VDN if the system has Call Vectoring). If the second agent was not reached by dialing a QDN (or VDN), the second agent is not removed from the agent queue.

Beginning with Issue 3.0 of DEFINITY Generic 2.1, the operation of the Transfer feature has been changed for an ACD agent (or other voice terminal user) who transfers an incoming RLT call to a branch location in a CAS arrangement. Refer to the Centralized Attendant Service interaction and to the User Operations section of this feature description for more information.

Beginning with DEFINITY Generic 2.2, the switch notifies CMS if a measured ACD agent transfers an ACD call.

Unattended Console Service—Preselected Call Routing

Beginning with Issue 1.6 of R2 V3 and Issue 1.2 of R2 V4, an ACD split can indirectly perform night service for the attendant queue. The designated terminal, the common night terminal, or the default night terminal for preselected call routing can activate Call Forwarding—Follow Me to forward all calls to an associated extension number of an ACD split. When this is done, every call to the night terminal enters the ACD split's queue and receives normal ACD processing.

Restricting Feature Use

The Manager II, MAAP, and SMT can apply Termination Restriction to the individual extension number of an ACD agent. This has the effect of preventing direct calls from terminating at the agent's terminal. However, calls to the ACD split terminate normally at the agent's position.

An attendant can apply Controlled Termination Restriction to the individual extension number of an ACD agent. This has the effect of preventing direct calls from terminating at the agent's terminal. However, calls to the ACD split terminate normally at the agent's position.

Hardware Requirements

The ACD feature requires the following additional or special hardware.

To provide the CMS option:

- A local 3B2 or 3B5 with CMS software to manage ACD activity (An Al? 16 can also be used with R2 V3 switches and ***nonvectoring*** R2 V4 switches.)
- A dedicated DCIU link to the local 3B2, 3B5, or AP 16

To provide the recorded announcements:

- 13A announcements system(s) (eight channels per 13A), A 36A voice coupler, with a 2012D power transformer, is required for each 13A announcement trunk.

  or

- KS-65270, L12 Single Channel Digital Announcer(s) to provide the recorded announcements. One line circuit of an SN228B, SN229, or TN742 (eight circuits per pack), or one circuit of a TN746 (16 circuits per pack) must be provided for each KS-65270 to support remote announcement recording.

  or

- KS-65272 4-Channel Digital Announcer to provide recorded announcements. One line circuit of an SN228B, SN229, or TN742 (eight circuits per pack), or one circuit of a TN746 (16 circuits per pack) must be provided for each KS-65272 to support remote announcement recording.
- Space in an auxiliary cabinet to house the announcement set(s)
- An auxiliary trunk circuit of an SN231 or TN763C circuit pack for each announcement trunk (four circuits per SN231 or TN763C)

To provide the queue warning lamp control option:

- 30A8 system status indicator panel to display queue warning status for eight ACD splits
- SN241 (eight circuits per circuit pack) contact interface

To provide lamp monitoring of ACD agents:

- 106B display unit to monitor the status of 20 ACD agents. Two circuits of an SN224 or TN735 circuit pack (four circuits per SN224 or TN735) are required for each 106B display unit.

To provide convenience for agents, supervisors, and observers:

- Headsets
  - 3122 StarSet II PLANTRONICS* headsets. These headsets are recommended for ACD agents. Several models are available.
  - 3122 Starset Supra headsets. These headsets are recommended for split supervisors and observers Several models are available.

    NOTE: The recommended Starset headsets are equipped with an automatic volume control to limit unattenuated touch-tones to an acceptable volume.

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* Trademark of Plantronics, Inc.

NOTE: The Starset headsets are not compatible with the 7407D voice terminal. (However, the Starmate headsets can be used with this voice terminal.)

- 31712 headset adapters to connect headsets to voice terminals

  NOTE: The headset adapters are not needed to connect the Starset headsets to the CALLMASTER voice terminal.

To provide display capabilities for agents:

- CALLMASTER, 7406D With Display, 7407D, and 7405D (with D401A display module) voice terminals, or the ISDN Advantage.

To combine displays with ISDA—BRI capabilities for agents:

- 7506 or 7507 BRI voice terminals

Feature Administration

Assignment of the ACD feature is on a per-system, per-trunk group, and a per-terminal class-of-service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
AUTOMATIC CALL DISTRIBUTION

PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the voice terminal class of service to the agents' extension numbers	Yes
001	—	Assigns an associated extension number to a split supervisor's individual extension number.	Yes
010	1	Assigns ACD membership to an agent's voice terminal class of service. Assigns agent override to an observer's class of service. Also, use this procedure to assign ACD Queue-Status Display to an ACD agent's class of service.	Yes
011	1	Assigns an ACD split number as the final point in a coverage path. **NOTE:** When the Call Vectoring feature is assigned to the system, an ACD split number cannot be assigned as the final point in a coverage path.	Yes
026	1	Administers the split size, the ICI message number, the queuing trunk group, the outflow and inflow levels, the lamp control circuit for the queue warning indicators, the hunting (call distribution) type, and the split type. Each split is identified by a split number [a number between 1 and 60 (30 in R2 V3)]. Beginning with DEFINITY Generic 2.2, the split size field has been removed from this procedure. The size of an ACD split does not have to be specified in multiples of 16 (16, 32, 48, etc.).	Yes
026	2	Assigns the split supervisor's extension number, the queue directory number (first associated extension number), priority extension number, auto available", and Multiple Call Handling (beginning with R2 V4) to a split. Beginning with DEFINITY Generic 2.2, this procedure is used to specify whether an ACD split is measured by CMS or not and to administer the Forced Entry option. (For information about the Forced Entry option, see the Call Work Codes feature description.)	Yes

* "Auto available" ***should not*** be assigned to an AUDIX split or to splits with human agents.

ADMINISTRATION PROCEDURES AUTOMATIC CALL DISTRIBUTION (Contd)			
PROCEDURE	WORD	PURPOSE	SMT
026	3	Assigns nonsupervisory members to a split in the order in which direct or circular hunting is to take place. Beginning with DEFINITY Generic 2.2, this procedure is used to assign CMS measurement to nonACD extensions by assigning extensions to split zero.	Yes
026	4	Assigns the ACD system supervisor and warning tone for service observing.	Yes
026	5	Beginning with DEFINITY Generic 2.2, this display-only procedure shows whether the specified split is measured by CMS or not, the number of members (agents) assigned to the split, the number of members that can be added to the split, the highest member number assigned to the split, and the total number of members assigned to all splits.	Yes
027	1	Assigns the first delay recorded announcement and the timing intervals preceding the first and second delay recorded announcements to an ACD split.	Yes
027	2	Assigns the second delay recorded announcement to the switch.	Yes
027	3	Assigns the city-of-origin and queue-of-origin announcements to an ACD split.	Yes
028	1	Assigns an extension or a range of extensions to be measured by CMS. Beginning with DEFINITY Generic 2-2, this procedure has been removed. CMS measurement is assigned on a per-split basis in Procedure 026, Word 2. CMS measurement of nonACD extensions is assigned on a per-extension basis in Procedure 026, Word 3.	Yes
028	2	Busies out the CMS system. Beginning with DEFINITY Generic 2.2, this function moves from Procedure 028, Word 2 to Procedure 028, Word 1.	Yes

ADMINISTRATION PROCEDURES AUTOMATIC CALL DISTRIBUTION (Contd)			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
051	1	Assigns multiappearance voice terminal and data module translations including origination and termination preferences. Before an automatic line appearance can be assigned to a voice terminal (in Procedure 052, Wodr 1), the origination and termination preferences (fields 10 and 11) must be set to zero using this procedure. **NOTE:** For a CALLMASTER voice terminal, the origination preference (field 10) must always be set to zero, no matter what type of line appearances are assigned.	Yes
052	1	Assigns automatic answering (line type) to an appearance (or a set of appearances) of a multiappearance terminal. **NOTE:** An automatic line appearance cannot be assigned to an SLS (Straight Line Set).	Yes
054	1	Administers the feature buttons, the stroke count buttons, and the RELEASE button for the automatic answering mode. Refer to Table 17–D for a cross--reference of applicable feature buttons and dial access encodes.	Yes
054	2	Assigns the SERVICE OBSERVE button to an observer's multiappearance voice terminal.	Yes
060	1	Assigns a split member to a position on the 106B display unit. Beginning with DEFINITY Generic 2.2, 106B assignments are based on extension number rather than split and member numbers.	Yes
060	2	Associates the halves of the 106B display unit	Yes
100	1	Assigns a trunk type to a trunk group. The applicable encodes include: 6 Special queue 65 SN241 contact interface 66 CAS release link trunk 1-way incoming at main 90 ACD first announcement 91 ACD second announcement 92 ACD origin announcement.	No

ADMINISTRATION PROCEDURES
AUTOMATIC CALL DISTRIBUTION (Contd)

PROCEDURE	WORD	PURPOSE	SMT
115	—	Assigns termination to an ACD split, priority queuing, and CMS measurement to a trunk group. Beginning with Issue 3.0 of DEFINITY Generic 2.1, an RLT trunk group can terminate to an ACD split.	No
150	—	Assigns the equipment locations of auxiliary trunks and incoming ACD trunks to their trunk-group numbers.	No
155	—	Administers the contact interface circuit for the queue warning indicators.	No
204	1	Designates the desired alphanumeric display for incoming calls to a split's special queue trunk group that reach an attendant.	No
256	1	Assigns the DCIU data link characteristics for CMS.	No
256	2	Administers the level 2 timers and counters for the DCIU link.	No
256	3	Administers the level 3 timers and counters for the DCIU link.	No
257	1	Administers the components and priority status of the DCIU channel.	No
257	2	Administers the DCIU port for the DCIU channel.	No
257	5	Reserves a DCIU port for use by the CMS system (beginning with R2 V4).	No
258	1	Copies the translation changes made using Procedures 256 and 257 to working tables.	No
258	2	Refreshes the DCIU temporary translation tables before using Procedures 256 and 257.	No
275	4	Assigns abandon call search, answer supervision, and CMS for ACD to the system class of service.	Yes
350	1	Assigns the first digit of the feature dial access codes (if required).	No
350	2	Assigns the ACD feature dial access codes. Refer to Table 17-D for a cross-reference of the applicable dial access codes and feature buttons.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — AUTOMATIC CALL DISTRIBUTION	
PATH NAME	**PURPOSE**
terminal-change terminal display unit	Displays or prints the split members being monitored by a 106B display unit
terminal-change class-of-service attributes	Assigns ACD membership to a voice terminal class of service and assigns agent override to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change group call-distribution attributes	Administers the characteristics of the ACD split.
terminal-change group call-distribution members	Adds or removes agents to/from an ACD split. Also, use this screen to assign automatic answering to the agent's extension. This screen is also used to assign the split's priority associated extension number.
terminal-change group call-distribution trunk-groups	Assigns trunk group termination to an ACD split and priority queuing to the trunk group.
terminal-change system parameters (select the Call-Distribution option)	Declares the attendant-console number of the system supervisor and assigns abandon call search and answer supervision to the system class of service.
terminal-change terminal buttons	Assigns the feature buttons for the ACD feature. Refer to Table 17–D for a cross-reference of applicable feature buttons and dial access encodes.

The following is the applicable FM path name used with the AP 16.

FM SCREEN — AUTOMATIC CALL DISTRIBUTION	
PATH NAME	**PURPOSE**
facilities-mgmt dciu link-assignments	Generates a report of the current DCIU link assignments.

DCIU Administration

For a CMS system on an adjunct computer, refer to Appendix H of this manual for detailed information about DCIU administration. This information includes some specific instructions for assigning a DCIU link to a CMS system.

Automatic Callback

Description

The Automatic Callback feature allows voice terminal users to call a busy extension number, go on-hook, and have their calls automatically placed by the switch when the called extension becomes idle.

With Automatic Callback, the calling party doesn't waste time and effort by repeatedly calling a busy terminal. Communication is increased by reaching the busy terminal as soon as both terminals are idle and before the called party leaves the busy terminal.

The switch monitor the busy/idle status of both the calling and called extensions. When the called extension is idle, and the calling terminal has an idle appearance, the switch sends a priority call (3-burst ringing) to the calling terminal. When the calling terminal goes off-hook, the switch rings the called terminal.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Activate Automatic Callback

With an AUTO CALLBACK button:

1. Go off-hook on an idle appearance. [Dial tone]
2. Dial the extension number of the called terminal. [If the called voice terminal is busy, busy tone is heard].
3. Press **[AUTO CALLBACK].** [Confirmation tone is heard, and the AUTO CALLBACK lamp lights].
4. Go on-hook. [If the activating voice terminal is off-hook when the called voice terminal goes idle, the AUTO CALLBACK lamp flutters on the activating voice terminal.

   When both the calling and called voice terminals become idle the Automatic Callback sequence is initiated by the switch. (3-burst distinctive ringing is heard, and the AUTO CALLBACK lamp flutters on the activating terminal)].

5. Go off-hook on an idle appearance. [Ringback tone is heard, and the called terminal rings].
6. A 2-party connection is established when the called party goes off-hook.

Without an AUTO CALLBACK button:

1. Go off-hook. [Dial tone]
2. Dial the Automatic Callback access code. [Second dial tone]
3. Dial the extension number of the called terminal.
4. If the called voice terminal is busy, confirmation tone is heard.
5. If the called voice terminal is idle, ringback tone is returned to the calling voice terminal, and the called voice terminal rings. (Automatic Callback is not needed).
6. To activate automatic callback, go on-hook within 6 seconds.
7. The called voice terminal goes on-hook. [The calling and called voice terminals are idle. Distinctive 3-burst ringing is heard at the activating terminal].
8. Go off-hook. [Ringback tone is heard, and the called voice terminal rings].
9. A 2-party connection is established when the called party goes off-hook.

To Cancel Automatic Callback

Without an Assigned Feature Button:

1. Go off-hook. [Dial tone]
2. Dial the Cancel Automatic Callback access code. [Confirmation tone]
3. Go on-hook.

With an Assigned Feature Button:

1. Go off-hook. [Dial tone]
2. Press **[AUTO CALLBACK]** . [Switch returns confirmation tone, and the AUTO CALLBACK lamp goes out].
3. Go on-hook.

Considerations

Switch Capacities

Any number of voice terminals can activate Automatic Callback toward other voice terminals in the local switch (or the DCS network).

Moreover, any number of voice terminals could activate Automatic Callback toward the ***same*** voice terminal.

However, ***each*** voice terminal can only have Automatic Callback activated toward one other voice terminal at a time. When Automatic Callback is already activated toward one voice terminal, a user can cancel this callback request and then activate Automatic Callback toward another voice terminal.

Intercept Tone

Intercept tone is returned when the calling party does not go on-hook within 6 seconds of receiving confirmation tone, the Automatic Callback feature is already in use and the access code is dialed again, or the activating voice terminal tries to activate Automatic Callback with a call on hold.

Calls Outside the DCS

A caller cannot initiate Automatic Callback toward any voice terminal that is outside the DCS network.

Callback Time Out

The callback must complete within 20 to 40 minutes or the feature request is canceled. The user must then reinitiate the callback.

Automatic Callback to Multiappearance Voice Terminals

Automatic Callback can only be activated toward a busy extension. However, multiappearance terminals provide multiple appearances of the same extension. As such, the called multiappearance voice terminal user must be using every incoming (not originating only) appearance of the extension before the switch will return busy tone to the calling party. As a result, multiappearance terminals are not usually busy or are busy for relatively brief intervals. As a result, the Automatic Callback feature is seldom if ever needed when the called terminal is a multiappearance voice terminal.

Hard and Soft Processor Swaps

Automatic Callback activations are stored in a status portion of switch memory. Therefore, if a voice terminal user activates Automatic Callback and then a hard processor swap occurs, the switch will be unable to initiate the callback sequence.

If a hard swap occurs during a callback sequence, the callback fails. The Automatic Callback feature operates normally during a soft processor swap.

Interactions with Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

Activation of Automatic Callback is denied if Attendant Call Waiting is activated toward the calling terminal line. Activation of Attendant Call Waiting is denied if a called single-appearance terminal has Automatic Callback active. Attendant Call Waiting is denied if all the lines of a multiappearance voice terminal are busy.

ACD (Automatic Call Distribution)

Automatic Callback can be activated toward the individual extension number of an ACD agent. However, when this is done toward an agent in a busy split (that is, calls are waiting in queue), a race condition occurs. The switch simultaneously begins the Automatic Callback sequence and the distribution of the next queued call when the agent goes on-hook. Since the ACD call-distribution algorithms are considerably faster, the agent will usually (if not always) receive the ACD call first.

Automatic Callback can also be activated toward an associated extension number of an ACD split. When this is done, the callback sequence is performed against the split supervisor's voice terminal. Again, if the split is busy and if the split supervisor is answering ACD calls, the same race condition occurs with the same results.

NOTE: Automatic Callback is usually activated toward an agent to facilitate calling from within the organization, and is always more effective when activated toward single-appearance terminals. If ACD agents (using single-appearance terminals) need to receive intracompany calls, the race condition can be avoided by using the Priority Calling feature. However, the better solution would be to provide agents with multiappearance voice terminals equipped with either two or three appearances. In this way, an extra appearance is available for routine communication with other departments.

Bearer Capability

The Bearer Capability feature functions normally with the Automatic Callback feature in most instances. When the callback call is placed by the switch, the Bearer Capability associated with the originally called station is used to place the call. In most instances, this will pose no problems.

However, when both the called and calling terminals are ISDN—BRI Voice / Data stations using a single extension number (BCCOS 2), a complication can arise. In this instance, there is no way of telling if the original call was voice or data (the original bearer capability IE is not preserved). The callback call could be either a voice or a data call, without regard to what type of call was originally placed.

Busy Verification of Lines

While terminal B is on-hook waiting for terminal A to become idle after activating Automatic Callback toward terminal A, both terminals can be busy verified. But, when terminal A goes on-hook and the automatic callback process begins, neither terminal can be busy verified until the talking connection between terminals A and B has been established.

Call Coverage

After a voice terminal user with coverage active activates Automatic Callback toward another voice terminal, the 3-burst callback call is treated as a priority call and does not route to coverage.

Automatic Callback cannot be activated toward a voice terminal whenever a regular nonpriority call would have redirected to coverage. Therefore, Automatic Callback cannot be activated toward a terminal that has Cover Active, Cover Busy, or Cover All assigned as the internal coverage criterion. Automatic Callback cannot be activated toward a terminal that has Send All Calls active. Also, Automatic Callback cannot be activated during the caller response interval of a coverage call. However, Automatic Callback ***can*** be activated toward a principal that is off-hook with either Cover Don't Answer or no criterion assigned as the internal coverage criterion.

Call Forwarding—Busy and Don't Answer

Call Forwarding—Busy and Don't Answer has no effect on the calling party's use of Automatic Callback. Callback always directs to the originating terminal, not to the forwarded-to terminal.

Call Forwarding— Don't Answer

Call Forwarding—Don't Answer has no effect on an Automatic Callback call origination. Callback calls are always directed to the originally called terminal, not to the forwarded-to terminal.

Call Forwarding— Follow Me

Call Forwarding—Follow Me has no effect on the calling party's use of Automatic Callback. The callback always directs to the calling terminal, not to the forwarded-to terminal.

If this feature is active at the called terminal, the forwarded-to terminal is treated as the called terminal.

If this feature is activated at the called terminal after the Automatic Callback call is placed, the originally called terminal remains the called terminal for the Automatic Callback call.

Call Pickup

A Call Pickup group member cannot use the Call Pickup feature to answer an Automatic Callback call to the originating terminal.

Call Vectoring

The switch denies activation of Automatic Callback toward a vector directory number. When this is attempted, the switch returns intercept tone to the activating party.

Whenever a "forced busy" step in a vector returns busy tone to a calling party, the switch denies activation of Automatic Callback in response to the busy tone. When this activation is attempted, the switch returns intercept tone to the calling party.

Call Waiting

If a busy party has a call waiting and another party tries to call the busy party, the switch returns busy tone to the calling party. The calling party can now activate Automatic Callback toward the busy party. However, the callback sequence is delayed until there are no calls waiting.

If a calling party activates Automatic Callback toward a busy terminal and then becomes busy with another call, use of the Call Waiting feature by another party toward the calling party is still allowed.

If a party is waiting on a busy line, the busy party with the waiting call cannot put the active call on hold and then activate Automatic Callback toward another line.

DCS (Distributed Communications System)

Automatic callback works within a DCS in essentially the same way that it works between two extensions on the same switch. Automatic Callback is considered a transparent feature in a DCS.

DDC (Direct Department Calling)

Any individual DDC group member can use the Automatic Callback feature.

If Automatic Callback is activated toward a DDC group number, callback occurs only when the calling terminal and the DDC controlling terminal become idle.

Hold

Activation of Automatic Callback is allowed if the called terminal line has a party on hold. However, callback is not completed until after the held call is serviced.

Hunting

If Automatic Callback is activated toward a busy terminal line in a hunt group, the call does not complete until that specific terminal line is idle. The call does not hunt.

IPA (Interpartition Access)

A caller (in a partition other than Extension Partition 0) is allowed to activate Automatic Callback toward terminals in the same partition or in Extension Partition 0.

If the caller tries to activate Automatic Callback toward a terminal in any other partition group, the switch returns intercept treatment.

A voice terminal user in Extension Partition 0 is allowed to activate Automatic Callback toward any voice terminal in the switch.

LXD (Last Extension Dialed)

Automatic Callback can be used in conjunction with Last Extension Dialed. This is useful when activation of the Automatic Callback feature toward the last extension dialed is desired. (Dial the Automatic Callback DAC or press the feature button before pressing the LXD feature button).

LND (Last Number Dialed)

Automatic Callback can be used in conjunction with Last Number Dialed. This is useful when activation of the Automatic Callback feature toward the last number dialed is desired. (Dial the Automatic Callback DAC or press the feature button before pressing the LND feature button).

Line Lockout

Automatic Callback can be activated toward a voice terminal in the lockout condition. When the voice terminal in the lockout condition goes on-hook, the callback process will begin.

Priority Calling

If a busy party has a priority call waiting and another party tries to call the busy party, the switch returns busy tone to the calling party. The calling party can then activate Automatic Callback toward the busy party. However, the callback sequence is delayed until there are no calls waiting.

If a calling party activates Automatic Callback toward a busy terminal and then becomes busy with another call, the switch does not allow application of the Priority Calling feature *by another party* toward the original calling party.

When the switch initiates the Automatic Callback sequence, if the activating party (the original calling party) is using a multiappearance voice terminal, the callback call to the original calling party is a priority call (3-burst ringing). This means that if other appearances are busy, the call will ring on an originate only appearance if one is available. However, if the activating party is using an analog (single appearance) voice terminal Priority Calling is not used. Rather, the switch waits until the calling party goes on-hook.

Queuing

The Automatic Callback feature and ringback queuing, at the local switch, cannot be activated at the same time by a terminal.

Activation of the Automatic Callback feature towards a terminal line that has Queuing activated is allowed, but deferred, until either the Queuing process is resolved or the administered Queuing interval times out.

Remote Access

A remote access user cannot use the Automatic Callback feature. The switch has no way of placing a callback call to an off-net number.

Restriction— Attendant Control of Voice Terminals

If any of the Attendant Control of Voice Terminals restrictions are activated on either the calling or called lines after the Automatic Callback call origination (but prior to the start of the call completion sequence), the restrictions are ignored for the call.

Tenant Services

A voice terminal user (in a partition other than Extension Partition 0) is allowed to activate Automatic Callback toward voice terminals in the same partition or in Extension Partition 0. When the user tries to activate Automatic Callback toward a voice terminal in any other partition, the switch returns intercept treatment.

A voice terminal user in Extension Partition 0 is allowed to activate Automatic Callback toward any voice terminal in the switch.

UCD (Uniform Call Distribution)

Any individual UCD group member can use the Automatic Callback feature. If Automatic Callback is activated toward a UCD group number, callback occurs only when the calling terminal and the UCD controlling terminal become idle.

WCR (World Class Routing)

The Automatic Callback feature is compatible with the WCR feature within a DCS. Automatic Callback uses the WCR feature on DEFINITY Generic 2.2 switches when a callback call is made within a DCS.

Hardware Requirements

None.

Feature Administration

Assignment of the Automatic Callback feature is on a per-extension class of service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

Administration Procedure Automatic Callback Feature			
Procedure	**Word**	**Purpose**	**SMT**
000	1	Assigns the class of service to an extension number.	Yes
010	1	Assigns the Automatic Callback feature to an extension class of service.	Yes
054	2	Assigns the Automatic Callback feature button to a multiappearance voice terminal. The applicable encode is: 6 Automatic Callback.	Yes
200	1	Specifies the don't answer timing interval (one to eight cycles) for Automatic Callback.	No
350	1	Assigns the first digit of the dial access codes for the Automatic Callback feature.	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 9 Cancel Automatic Callback 19 Activate Automatic Callback.	No

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Automatic Callback	
Path Name	**Purpose**
terminal-change class-of-service attributes	Assigns Automatic Callback to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change terminal buttons	Assigns the Automatic Callback feature button to a multiappearance voice terminal.

Notes:

Automatic Circuit Assurance

Description

This feature provides a means of early detection of possible trunk problems. Early detection of problems can reduce out-of-service time and irritation caused by trunks not performing up to standards.

Detectable Fault Conditions

Frequently, callers connected on a trunk with poor transmission characteristics will hang up and call again. This results in many ***short hold time*** calls for these trunks. Another common trunk failure is an inability to disconnect normally after the call (the trunk circuit remains open after both ends of the call have disconneted). This results in ***long hold times*** for these trunks.

The ACA (Automatic Circuit Assurance) feature can detect these problems by comparing trunk hold times to standards established by the user (long- and short-hold times). When a long duration call or a specified number of short duration calls is detected on a trunk, a designated attendant is automatically alerted by a switch-generated **referral call.** The attendant can then either check the condition of suspect trunks by using the Trunk Verification—Attendant feature or take other action that may be appropriate.

Threshold Limits

Different users will have different norms or average hold times depending on their type of business or calling activity. For example, an order taking operation will have a relatively short average call duration, say 2- to 3-minutes, while data calls (data terminals linked to central computers, etc.) will normally have very long average call times (several hours). For this reason, both the short- and long-hold time thresholds can be set separately for each monitored trunk group.

- The short call threshold can be set to any even value between 0 and 160 seconds. This is the value that determines if a short call has occured.
- The short call referral level can be set to any even value between 0 and 30. This is the number of short calls that will cause the switch to initiate a short call referral.
- The long call limit can be set to any value between 0 and 24 hours.

A typical short call threshold value for voice calls would be from 10 to 20 seconds; while for a data call, a duration of 160 seconds could be considered a short call. A typical long call threshold value for voice calls would be 1 to 2 hours; while for data calls, the duration might be 4 hours or more.

As an example, consider a trunk group that is usually used for voice traffic. The short call threshold for this trunk group is set to 10 seconds, and the short call referral level is set to 8. Given this arrangement, when a trunk within that trunk group experiences eight consecutive calls that are hung up within 10 seconds, the switch generates an ACA referral to the designated attendant console.

The ACA referral from the switch to the attendant appears like an incoming call with an identifying attendant display (such as, ACAL).

Release 2, Version 4 Enhancement

Prior to R2 V4 System 85, ACA referral calls were sent either to a local attendant of CAS (Centralized Attendant Service) attendant. Beginning withR2 V4 System 85, another option is available. ACA referral calls can now be sent to a ***central referral point*** associated with a maintenance or service center, such as Trouble Tracker rather than a local or CAS attendant.

When an ACA referral is sent to the central referral point, specific trunk failure data is not included in the referral message. The referral call shows only that a possible trunk failure has occurred. To obtain trunk failure data, the central referral point polls the switch and accesses the ACA ***audit trail record.*** This record contains trunk data for the 32 most recent suspected failures. The audit trail record is stored at the switch where the possible failure was detected.

Feature History and Development

This feature was first available on System 85 in Release 1. Beginning with R2 V4, ACA referral calls can be sent to a central referral point. Otherwise, there are no other enhancements to the ACA feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Activate ACA:

1. From an attendant console, press an idle loop button. [PA lamp goes out, and loop lamp lights].
2. Press **[START] .** [Dial tone]
3. Dial the ACA Start access code. [Confirmation tone]
4. Press **[RELEASE]** . [PA lamp lights].

To Deactivate ACA:

1. From an attendant console, press an idle loop button. [PA lamp goes out, and loop lamp lights].
2. Press **[START]** . [Dial tone]
3. Dial the ACA Stop access code. [Confirmation tone]
4. Press **[RELEASE]** . [PA lamp lights].

To Identify a Possibly Defective Trunk:

1. From the designated attendant console, press **[ANSWER]**.
2. Press **[TRK ID]**. [Alphanumeric display shows the trunk-group access code].
3. Press **[TRK ID]**. [Alphanumeric display changes to show individual trunk number].
4. Press **[TRK ID]**. [Alphanumeric display changes to show original referral message].
5. Press **[RELEASE]**.
6. Test the trunk and/or take other actions that may be appropriate.

Considerations

Designated Console

Only one attendant console can be designated to receive system-generated ACA referral calls, regardless of the number of attendant consoles that may be assigned to the switch.

Statistical Probability of Error

Due to the statistical nature of the operation of the ACA feature, the switch will occasionally alert the attendant to check a good circuit. It must be understood that an occasional long call or series of short calls will occur as a matter of chance and trigger an ACA referral. An ACA referral call does not confirm a problem, but merely suggests the possibility of a problem.

Data Calls

Data calls normally have longer holding times than voice calls. For the ACA feature to be most effective on a switch that carries a lot of data traffic, it is desirable to provide separate trunk groups for data calls and for voice calls. A different long- and short-hold time threshold can be set for each trunk group (Procedure 120, Word 1) based on the type of traffic it is expected to carry.

Maintenance Alarms and ACA Referrals

When ACA referrals are directed to a local or CAS attendant, maintenance alarms can be directed to a ***separate*** destination, such as RMATS. However, when a central ACA referral point (such as Manager IV) is used, maintenance alarms must be directed to the ***same*** destination.

Hard and Soft Processor Swaps

The short call threshold, the short call referral level, and the long call limit are stored in a translation portion of memory. Therefore, these parameters will endure a hard processor swap.

Attendant activations of ACA toward trunk groups are stored in a translation portion of switch memory. Therefore, these activations will endure a hard processor swap.

The contents of ACA referral messages are stored in a status portion of switch memory. Therefore, if a hard swap occurs while the designated attendant is reviewing a referral message, the unread part of the referral messages is lost.

If a hard swap occurs while the switch is placing a referral call to the designated attendant, the referral call fails.

The ACA feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Dedicated Switch Connections

The ACA feature ignores a long holding time threshold exception when a Dedicated Switch Connection is involved in the call.

DCS (Distributed Communications System)

In a DCS environment with as many as 40 nodes, attendant consoles can be concentrated at one node. When an attendant at this node activates or deactivates the ACA feature, the feature is activated or deactivated for every switch in the DCS. In a DCS environment, the operation of ACA is similar to the operation in a non-DCS environment. However, the initial alphanumeric display to identify a faulty trunk is different. In addition to the usual information, the initial display shows the dial access code of the tie trunk involved in the ACA referral call. In a DCS, trunk groups that require the use of the ACA feature across DCS nodes cannot have 4-digit dial access codes.

Tenant Services

An attendant in Attendant Partition 0 is allowed to activate or deactivate the Automatic Circuit Assurance feature. However, the switch will only direct ACA referral calls to the designated attendant console. The designated console must belong to Attendant Partition 0.

Attendants (in partitions other than Attendant Partition 0) are not allowed to activate ACA, deactivate ACA, or receive ACA referral calls.

Trunk Verification—Attendant

The Trunk Verification—Attendant feature is the principal tool used to follow up an ACA referral call. A distant trunk which has an ACA record associated will be updated.

Unattended Console Service—Preselected Call Routing

When the switch is in the Preselected Call Routing mode, partial ACA operation allows ACA trunk referrals to be recorded in the audit trail without sending referral calls to the attendant.

Hardware Requirements

None.

Feature Administration

Assignment of the ACA feature is on a per-system and on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the FM (Facilities Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

The feature can also be administered using Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES AUTOMATIC CIRCUIT ASSURANCE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
120	1	Administers the short call threshold, the long call limit, and the short call referral level for each trunk group.	Yes
203	1	Assigns the TRK ID button to the attendant console. The applicable encode is as follows: 28 TRK-ID Button.	No
204	1	Designates the desired alphanumeric displays for referrals to the designated attendant. R2 V1 to R2 V3: 298 Short call threshold 299 Long call limit R2 V4 and later: 2298 Short call threshold 2299 Long call limit.	No
285	—	Assigns the ACA feature to the system class of service.	Yes
286	1	Administers the system-wide characteristics of ACA.	Yes
350	1	Assigns the first digit of dial access codes for the ACA feature (if required).	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 62 ACA - Start 63 ACA - Stop.	No

The following are the applicable FM path names used with the AP 16.

FM SCREENS — AUTOMATIC CIRCUIT ASSURANCE	
PATH NAME	**PURPOSE**
facilities-mgmt system-parameters aca-activation	Assigns the ACA feature to the system class of service and designates the attendant console to receive ACA referral calls.
facilities-mgmt trunk-groups aca-thresholds	Assigns the short call threshold, the long call limit, and the short call referral level to each trunk group.
facilities-mgmt trunk-groups maintenance-busy	Displays or prints a report of those trunks listed as "maintenance-busy".
facilities-mgmt traffic reporting report	Displays or prints the audit trail report (select the ACA referral option) of ACA referrals.

Automatic Identification of Outward Dialing

Description

The AIOD (Automatic Identification of Outward Dialing) feature is used by both the telephone company and the customer to provide valuable information about outgoing calls. The AIOD feature identifies, translates, and transmits the calling party's extension number and trunk-group access code to the serving CO (Central Office), or either the CCSA (Common Control Switching Arrangement) or EPSCS (Enhanced Private Switched Communications Service) switching office. The CO, CCSA, or EPSCS switching office contains AMA (Automatic Message Accounting) equipment for recording this information for later billing.

The telephone company uses the AIOD feature to provide the customer with reports containing detailed toll-call information. This information helps the customer with cost allocation, traffic analysis, and the policing of toll calls. Call screening is possible because each toll-call record includes the calling extension number and the called (destination) telephone number. The duration of a call is part of the call record. This record assists in identifying calls that are made frequently and of excessively long duration. The time of day is recorded in each call record to assist in identifying any suspicious out-of-hours calls.

Other Call Record Features

Besides the AIOD feature, System 85 and DEFINITY Generic 2 offer a variety of other CDR formats: SMDR (Station Message Detail Recoding), CDRR (Call Detail Recoding and Reporting), and Variable Format Call Detail Recording. These alternative formats allow the switch to independently record information about outgoing calls. (The CO records aren't needed).

Feature History and Development

The AIOD feature was first available for System 85 in Release 1. During the Release 1 time frame, the AIOD circuitry was provided on two circuit packs. By the Release 2 time frame, this circuitry was compressed onto a single pack, the SN244. Otherwise, this feature has remained unchanged since its introduction.

Billing

The System 85 or DEFINITY Generic 2 may be arranged for five types of individual AIOD billing numbers:

- ***Individual Voice Terminal Billing —*** Once the voice terminal has finished dialing the dial access code for an outward AIOD call, the extension number and trunk-group access code are sent to the CO, CCSA, or EPSCS switching office. If a bill is prepared for this call, the extension number is used.
- ***Auxiliary Voice Terminal Billing —*** After an auxiliary ANI (Automatic Number Identification) billing number has been assigned (Manager II or MAAP adjustable) and the voice terminal(s) has finished dialing the trunk-group access code for an

outward AIOD call, the auxiliary extension number and trunk-group access code are sent. Toll calls placed using Remote Access trunks are treated identically (charged to the auxiliary voice terminal number). This function is useful for billing many voice terminals to a single extension number.

- ***Auxiliary Trunk Billing*** — After an auxiliary ANI billing number has been assigned and the voice terminal(s) has finished dialing the trunk-group access code for an outward AIOD call, the trunk-group access code and the auxiliary tie trunk number are sent. Toll call(s) placed by Remote Access users are charged to the auxiliary tie trunk number.
- ***Attendant Billing*** — An attendant with an assigned LDN can originate an outward AIOD call or complete an outward AIOD call that was originated by a voice terminal. After the attendant has finished dialing the dial access code for an outward AIOD call, the call is processed and billed to the LDN.
- ***Failure Billing*** — When an AIOD process is not completed due to a failure in the ANI function, the AIOD call is unidentified for billing purposes. For example, if an AIOD call is placed in queue and cannot be serviced, the identity of the call is unknown. The failure billing number associated with the trunk group over which the call is directed is recorded by the serving office, and the call is completed. The failure billing number is recorded on the AMA unit and is subsequently translated by the accounting center to the LDN of the local switch.

User Operations

The AIOD feature doesn't require special user actions. Once the feature is administered, its operation is completely automatic. When voice terminal users or attendants want to access a particular trunk group associated with AIOD, they just dial the trunk-group access code in the same manner as they would dial any other outgoing trunk group.

Considerations

Coordination With Serving Central Office

The serving CO must be equipped to receive ANI messages from the local switch.

Allowable Trunk Types for AIOD

The AIOD feature can record outgoing calling activity on the following trunk types:

- 12 — 15: 2-way APLT trunk types
- 17 — 20: 1-way outgoing or 2-way CO trunk types
- 22 — 25: 1-way outgoing or 2-way FX trunk types
- 27 and 28: Outgoing WATS trunk types
- 50: 2-way Remote Access trunk type.

Number of Recordable Trunks

The AIOD feature can be assigned to transmit call records for as many as 10,000 trunks in any number of trunk groups connected to the serving CO. However, in practice, the limit can be smaller. Several PBXs may need to share the AIOD equipment at the serving CO.

Five-Digit Dialing

If an AIOD billing number contains five digits, the leading digit is truncated because the CO will only receive the four least significant digits of the billing number. Therefore, this feature is not recommended for switches that use 5-digit extension number with more than one first digit.

Hard and Soft Processor Swaps

During a hard processor swap, the ANI buffer is cleared and ANI messages are not sent to the serving CO. Incomplete messages can also be transmitted when a hard swap occurs.

The AIOD feature operates normally during a soft processor swap.

Traditional Module Required

The AIOD feature is not available on DEFINITY Generic 2 switches with all universal modules. At least one traditional module is required for this feature.

Interactions With Other Features

Presently, AIOD has no interactions with other features.

Hardware Requirements

Traditional Module

The AIOD circuitry consists of a single circuit pack (SN244) on the System 85. The SN244 uses a single output channel to send the necessary information for billing by the CO. The SN244 can be used only with the traditional module.

Universal Module

There is no AIOD circuit for the universal module. In systems with all universal modules, AIOD is not available.

Feature Administration

This feature is assigned on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using Manager IV.

The applicable procedures are as follows.

ADMINISTRATION PROCEDURES
AUTOMATIC IDENTIFICATION OF OUTWARD DIALING

PROCEDURE	WORD	PURPOSE	SMT
000	2	Assigns auxiliary billing to an extension number.	Yes
100	1	Assigns the trunk type to the ANI trunk group. The applicable encode is as follows: 58 ANI interface.	No
101	1	Assigns AIOD billing numbers to trunk groups (tie trunks only).	No
116	1	Assigns the AIOD equipment number to a CO trunk, a ECSA/APLT trunk, a Foreign Exchange trunk, a WATS trunk, and a Remote Access trunk.	No
150	1	Assigns the equipment location of the ANI circuit to its trunk-group number. This Procedure also assigns up to 10,000 AIOD equipment numbers to CO trunks, ECSA/APLT trunks, Foreign Exchange trunks, WATS trunks, and Remote Access trunks.	No
204	1	Assigns a listed directory number to be handled by an attendant (R2 V3 and later).	No
275	1	Assigns AIOD ANI Timing Delay and Auxiliary ANI number to the system class of service.	Yes
352	1	Assigns listed directory number to be handled by an attendant (R2 V1 and V2).	No

Automatic Route Selection

Description

The ARS (Automatic Route Selection) feature provides alternate routing, also known as *least cost routing,* of public network calls. The ARS feature provides automatic selection from multiple routes for a call to reach its final destination.

Feature History and Development

The ARS feature was first available on System 85 in Release 1. In System 85, Release 2, Version 3, the following enhancements were added.

- There can be as many as 160 6-digit translators (up from 64 in System 85, R2 V1 and V2, and 32 in R1).
- Each 6-digit translator can have ten patterns (up from four).
- Each pattern can contain up to 16 preferences (up from 10).
- Pattern Queuing allows ARS queuing to more than one preference.
- ARS Subnetwork trunking was expanded to provide IXC Access.

  — Seven digits can be deleted from the front of the destination code (no change).

  — Twenty digits can be inserted in place of the deleted digits (up from four).

  — Inserted digits can be formed into four groups separated by pauses (up from two).

  — A maximum of 15 digits can be included in a single group.

In system 85, R2 V4, the following Generalized Route Selection enhancements were added.

- ARS call categories for partitioned switches
- Bearer Capability Classes (BCCs) for more effective routing.

In DEFINITY Generic 2, BCC routing by the ARS feature was expanded to allow routing according to as many as 256 Bearer Capability Classes of Service (BCCOSs). Also, for data calls, the preference-selection process for BC routing is enhanced. The ARS preference-selection software is more selective in its choices of trunk facilities and inserts Modem Pooling conversion resources (when necessary) to route data calls.

In System 85, R2 V4, Issue 2.0 and DEFINITY Generic 2.1, Issue 2.0, selective international (01X) call routing was added. This enhancement supports customers with international private line service by allowing the switch to examine "01X" plus the digits that follow in order to make routing decisions.

In Generic 2.2, the ARS feature is discontinued and replaced by the World Class Routing (WCR) feature.

Routing Structures

Plans

There can be from one to three ARS Plans. These plans are related to the time-of-day and day-of-week selection functions described later. The time-of-day and day-of-week functions are normally operated automatically as clocked functions; however, there is a manual override to this automatic clocking to allow for holidays and other special occurrences.

As a typical example, one plan would be used for normal business hours, another plan for off-hours on normal business days, and the third plan could be used for weekends and holidays.

Each plan is assigned a set of patterns and associated preferences in Procedure 309, Word 1.

Patterns

ARS routes are grouped into routing patterns. ***Routing patterns*** are ordered lists of the preferences that the switch can use to complete a particular call. Each ARS Plan can have from 1 to 64 patterns.

The term "Routing Designator" is synonymous with routing pattern.

Preferences

Each ARS routing pattern contains from 1 to 16 preferences. An ***ARS preference*** is a trunk group with associated characteristics such as Facility Restriction Level (FRL) and Bearer Capability Class of Service (BCCOS). Note that the associated characteristics are assigned to the pattern in Procedure 309 and may or may not be the same as the same characteristics assigned to the trunk group. For patterns with more than one preference, there is a first-choice preference and one or more alternate preferences arranged in order of desirability.

First-Choice Preference

Usually, the first-choice preference in an ARS pattern is the least expensive trunk facility for a given time-of-day and day-of-week. Each alternate route is increasingly more expensive. However, the first-choice route for a particular pattern can be selected on some criteria other than cost (for example, transmission quality).

Available Trunk Types

An ARS routing pattern may contain any or all of the following trunk types:

- Local Central Office (CO) trunk groups
- Foreign Exchange (FX) CO trunk groups
- Tie trunk groups
- Wide Area Telecommunications Service (WATS) Access trunk groups
- MEGACOM WATS Access trunk groups
- Software Defined Network (SDN) Access trunk groups
- Integrated Services Digital Network (ISDN) trunk groups.

Facilities Restriction Level (FRL)

General Operation

Each preference in an ARS routing pattern is assigned a Facility Restriction Level (FRL). Each originating facility (voice or data terminal, remote access trunk, etc.) is also assigned an FRL (the default FRL). When an ARS call is placed, the switch selects a routing pattern based on the first three or six digits (area and / or office code) of the dialed number. The switch then compares the FRL of the call (the default FRL) to the FRL of the first-choice preference in the selected pattern. A call can use a preference only if the FRL of the call is equal to or higher than the FRL of the preference. If the call has access to the first-choice preference, the switch checks that trunk group for an available trunk. If every trunk is busy, the switch checks the next preference (again assuming that the FRL of the call is high enough to gain access to that preference). The use of FRLs is discussed in more detail under the Facilities Restriction Level feature in this manual.

Multiple FRLs

Each trunk group is assigned its own FRL; however, this FRL is not used by the ARS feature. Each ARS preference is also assigned its own FRL which is not necessarily the same as the FRL of the associated trunk group. When a call is routed by the ARS feature, the FRL assigned to the preference in Procedure 309, Word 1 is used, not the FRL assigned to the trunk group. The same trunk group can reside in more than one pattern with a different FRL in each. In this way, the preference's FRL can more closely correspond to the cost of the call, as well as the cost of the trunk facility. Figure 21-1 shows an example of such an arrangement.

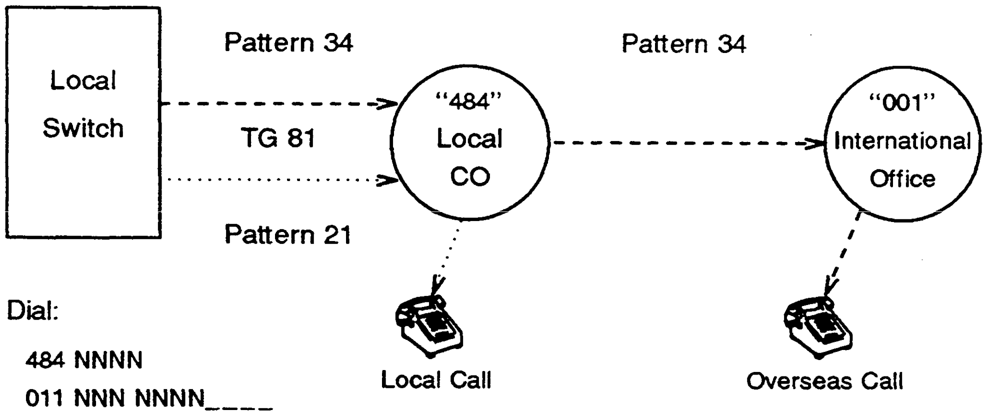


Figure 21-1. Multiple FRLs Assigned by Preference

In this example, office code "484" leads to a telephone in the local calling area. Office code "001" (corresponding to the dialed digits "011") leads to a telephone in Europe. The office code "484" points to Pattern 21, and office code "001" points to Pattern 34. Both Pattern 21 and Pattern 34 contain the trunk group 81 as a preference. Trunk group 81 leads to the local central office. For pattern 21, calls on this trunk group route to a local destination. For pattern 34, calls on trunk group 81 route through the same local CO to an international exchange for overseas destinations. This trunk group, serving in two patterns, has an FRL of "1" in Pattern 21 and an FRL of "7" in Pattern 34. Therefore, using the same trunk group, the calling party needs a much higher FRL to call Europe than to call home.

Authorization Codes

The Authorization Code feature can be used to provide an alternative to the default FRL. If the Authorization Code feature is assigned, and an ARS call fails to route because of an insufficient FRL, and if a higher FRL would provide access to additional preferences, the switch prompts the caller (with recall dial tone) for an Authorization Code. If the caller dials an authorization code that has an FRL that is higher than the default FRL, the switch replaces the default FRL with the authorization code's FRL. Using this new FRL the switch makes another attempt to find an available trunk. Authorization codes are discussed in more detail in the Authorization Codes chapter of this manual.

Queuing

If the new FRL is still too low (or if no authorization code is dialed) and the first-choice preference provides Queuing the switch tries to queue the call on that preference. The call's FRL (either the default FRL or the authorization code FRL) must allow access to the first-choice preference before the call can be queued.

The trunk group's queue length (number of calls allowed in queue) must also allow access to queuing. Once the call has entered the queue, the switch checks the first-choice preference every 2 seconds for an idle trunk. If no trunk becomes idle before the time-in-queue limit is exceeded, the switch makes a "last try", checking every accessible preference (based on the FRL) in the pattern.

Pattern Queuing

The Pattern Queuing option replaces queuing solely on the first preference on System 85 Release 2, Version 3 and later switches. The pattern queuing option allows any number of preferences in the pattern to be checked during the entire queuing process. The switch may check one preference (for example, the first-choice trunk group, ignoring Pattern Queuing), check the first two preferences, etc. If Pattern Queuing indicates that three of ten preferences are to be checked, the switch still checks all ten during the "last try".

When a call is placed in queue, it queues on the first-choice preference and is restricted by that trunk group's queuing parameters (queue length, time-in-queue limit, etc.).

CAUTION: *Care must be exercised when setting the number of preferences to be included in Pattern Queuing. An increase in the number of preferences to be checked means an increase in processing time. If this added processing time does not produce a significant increase in calls served, queues could begin to overflow.*

FRL Raising

Just before the "last try", when the time-in-queue limit elapses for an ARS call, the switch can raise the call's current FRL to help provide an accessible trunk facility for the call. FRL Raising is assigned on a per-system basis for both ARS and AAR in Procedure 330,

FRL Raising first compares the timed-out call's current FRL with the assigned Threshold FRL (field 3). If the call's current FRL is equal to or greater than the Threshold FRL, the call is qualified for FRL Raising. At this time, the switch ***considers*** substituting the assigned Raised FRL (Field 4) for the timed-out call's current FRL. This substitution is made if the Raised FRL is greater than the current FRL.

Allowing Toll Routes for Nontoll ARS Calls

Also, during the "last try", after the time-in-queue limit has elapsed for an ARS call, the switch automatically ignores ARS Toll Restriction (for those ARS calls where the ***nontoll*** access code was dialed). This operation helps provide an allowable trunk facility for the call.

During the "last try", nontoll ARS calls are allowed to access ***any*** ARS preference within the assigned pattern that is allowed by the current (or the raised) FRL. That is, during the "last try", the switch allows nontoll calls to access preferences that have warning tone assigned. However, when a nontoll call does access one of these preferences, warning tone is applied to the connection.

Destination Code

To place an ARS call, a terminal user dials the ARS access code (usually one digit but can be up to four digits) followed by a destination code (telephone number). The destination code has the form:

P- NPA - NXX - XXXX

P - Prefix digit (not always required)

NPA - Area Code [3-digit NPA (Numbering Plan Area)]

(The NPA is of the form NIX where N = 2 through 9, I = 0 or 1, and X = 0 through 9).

NXX - Office Code (3-digits), where N = 2 through 9 and X = 0 through 9

XXXX - Extension number (4 digits).

Calls to public network telephones within the local service area [also known as the HNPA (Home Numbering Plan Area)] do not require an area code. This results in a 7-digit destination code.

Prefix Digits

The switch examines the digit following the ARS access code to determine if a prefix digit is present. A prefix digit is either "0" or "1".

The Prefix "0"

- If the prefix digit"0" is dialed, and no other digits are dialed within 4 seconds, the call routes to a local central office for operator assistance. If more digits are dialed (within 4 seconds) the call is routed for special processing such as Calling Card Billing.
- If the digits "0" are dialed, and no other digits are dialed within 4 seconds, the call routes to the operator of the chosen IXC (Interexchange Carrier) for assistance.
- If the digits "011" are dialed followed by a destination code, the call is routed either to IDDD (International Direct Distance Dialing) or to selective international call routing.
- If the digits "010" are dialed followed by a destination code, the call is routed to international credit card dialing.

> **NOTE:** From the perspective of System 85, R2 V4, Issue 1.2 and earlier or DEFINITY Generic 2.1, Issue 1.0, "011" routing is the ***same*** as "010" routing. As the System 85 or DEFINITY Generic 2 receives the digits 01X (where X = 0 through 9), the switch treats the call as a generic international call and selects the pattern corresponding to office code "011".

For System 85, R2 V4, Issue 1.2 and earlier or DEFINITY Generic 2.1, Issue 1.0 switches, these prefix "0" calls are routed over two specific patterns that are assigned in Procedure 311, Word 1.

When the digits "0" or "00" are dialed, the switch selects the ARS pattern corresponding to office code "000" (Field 1), and then outpulses the "0" digit(s) and any subsequent digits to the serving CO for further routing.

When the digits "01X" are dialed, the switch selects the ARS pattern corresponding to office code "011" (Field 1), and then outpulses the digits "01X" and the subsequent digits to the serving CO (or the serving toll office) for further routing.

NOTE: To completely restrict access to IDDD calling, assign the office code "001" to the Intercept Pattern (usually Pattern 1) in Procedure 311, Word 1.

Selective International Call Routing

Processing for international "01X" calls changes with System 85, R2 V4, Issue 2.0 and DEFINITY Generic 2.1, Issue 2.0. With the selective routing enhancement, the switch can examine the digits following the IDDD designator (01X) to make routing decisions. Users can dial a number in the form "011+ XXXXXXXXXXXXXXX" and have the switch choose either private facilities or least cost (public) services as available. With proper administration, users can dial public (01X) numbers and the switch can capture on-net traffic for private facilities or off-net traffic that is closer to an international destination (tail-end hop-off). This concept is pictured in Figure 21-2.

End of Dialing Character

The pound sign signals *end of dialing* and avoids waiting for the interdigit timeout period at the receiving switch. The switch automatically sends the end of dialing character (#) after international and "dial 0" calls routed through ARS. International and dial 0 calls that use subnetwork trunking must be administered to allow for this extra digit in Procedure 309, Word 3.

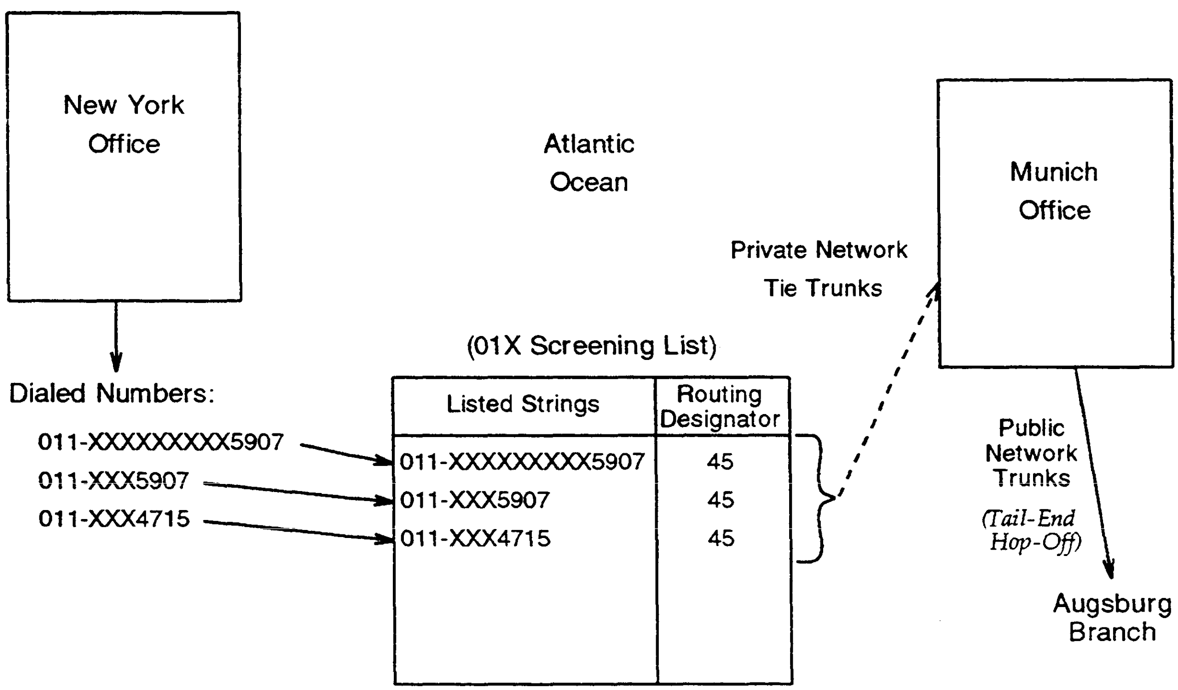


Figure 21-2. Selective International Call Routing

To use this enhancement, customer's set up a "01X exception list" for selected digit strings using Procedure 312, Word 3. When digit strings (beginning with 01X) are dialed, the switch checks the "01X exception list" for the dialed digits. If the dialed digits are on the exception list, they point immediately to a routing designator containing a list of facilities that can include both private and public network trunk groups. This allows customers to administer selected international dialing sequences to point to a pattern with private facilities first, while other international dialing sequences point to the default international routing pattern.

Administration for both public and private facilities is done through the ARS feature. This administration uses the same digit translation tables used by 10 to 7 digit conversion. The digit strings can be from 7 (minimum) to 18 digits in length and must begin with "01X". The corresponding routing pattern number (1 through 63) represent one of the existing ARS routing patterns defined in Procedure 309. This new administration is done using Procedure 312, Word 3, which is available only on the MAAP (Maintenance and Administration Panel) or the DEFINITY, Manager II.

The Prefix "1"

Dialing the prefix digit "1" can be optional. However, System 85 or DEFINITY Generic 2 callers must dial "1" in metropolitan areas where area codes and office codes can have the same three digits. Alternatively, the switch administrator can require that callers dial the prefix "1" so that they are always aware when a toll call is being placed. The following dial 1 options (assigned in Procedure 275, Word 3) are available:

- The prefix digit "1" does not have to be dialed. The System 85 or DEFINITY Generic 2 ignores the digit if it is dialed. As the switch outpulses the digits, the switch inserts the digit "1" if the central office at the other end of the selected trunk group needs it.
- The prefix digit "1" must be dialed for all 10-digit (area code + office code + extension number) destination codes. Seven-digit destination codes do not require the prefix digit "1". This option allows codes of the form NIX (where I = 0 or 1) to be used as office codes when the prefix "1" is not dialed. If NXX (including NIX) are the first three dialed digits, this is an office code in the HNPA, and the switch collects four more digits. If "1" plus NIX is dialed, this is an area code, and the switch routes the call according to the next three digits dialed (the office code NXX).
- The prefix digit "1" must be dialed for all domestic toll calls including 10-digit destination codes and 7-digit destination codes where the office code is included in the toll table.

Routing Codes

If the digit following the ARS access code is a number other than “0,” the switch examines the next three digits, excluding the prefix digit, to determine how many more digits to collect and how to route the call. These codes are one of the following:

- A service code
- An office code
- An area code

Service Code

A service code has the form N11, 1N11, or 11X (where N equals any of the digits 2 through 9, and X equals any of the digits 0 through 9). One example of a service code is 1411, the number for local directory assistance.

555 Routing

System 85 or DEFINITY Generic 2 users can dial the telephone numbers NPA-555-1212 for long-distance information. When this is done, these calls (destined for any NPA) normally route according to the assigned ARS pattern for routing 555 calls in Procedure 311, Word 1. [Enter "555" as the NXX (office code) in this procedure].

If the switch administrator attempts to assign separate 555 routing patterns for different NPAs in Procedure 311, Word 3, these assignments are not blocked by the administration software. However, these assignments are ignored by the System 85 or DEFINITY Generic 2 call-processing software.

The digit sequence for 800 Service Information (800-555-1212) is the exception to the previous routing information. The routing pattern for "800-555-1212" is either assigned as a 3-digit Foreign NPA (800) in Procedure 311, Word 2 or as a 6-digit NPA-NXX (800-555) in Procedure 311, Word 3.

Office Code or Area Code

To accommodate areas of the country where office codes and area codes have the same form, there are two ways to differentiate between these codes:

- NNX is always an office code and NIX (I stands for 0 or 1) is always an NPA

or

- NIX is an NPA only if the prefix "1" is dialed; otherwise, it is an office code of the form NXX.

The switch uses the NPA, service code, or office code as an index into software tables that contain information about the preferences in a routing pattern. If the leading dialed digits are a service code or office code in the range 200 to 999, the switch uses these digits as an index into the HNPA (Home Numbering Plan Area) table. If the leading dialed digits are an NPA or a service code in the range 200 to 919, the switch checks the associated 3-digit table. In turn, the selected entry in the 3-digit table either points to a routing pattern or to the 6-digit office code table associated with the dialed NPA. At this time, if the 6-digit table was pointed to, the ARS software indexes the 6-digit table according to the dialed NXX to select the correct pattern.

6-Digit Routing

The most significant difference between 3-digit and 6-digit routing is that 6-digit routing offers more than one (as many as ten) routes for a particular NPA. When an NPA's 3-digit table points to a pattern, calls to the NPA route via that pattern only. The contents (that is, the preferences) of the pattern can change by invoking a new ARS time-of-day plan. For 6-digit routing, the dialed office code determines what ARS pattern the switch uses to route the call.

After the switch selects a route, it examines the route data associated with that route (trunk group). This route data supplies the following information:

- Trunk-group number
- Minumum FRL to access trunks
- Warning tone (on or off)
- Terminating NPA of the trunk group (the area code of the destination public-network CO/private-network switch)
- Toll table provided (yes or no), and if yes, which office codes are toll and which are local (toll free)
- Dial-1 prefix sending for toll calls (as indicated by the route's assigned toll table)
- Whether the destination switch has extensions numbered 0XXX, where X equals any digit 0 through 9
- Digit modification and outpulsing requirements
- Bearer Capability Class.

Access Codes

Two ARS dial access codes are available.

Toll Code

The toll code allows access to every preference in a routing pattern. When routing would occur over more expensive trunk facilities, the switch can provide a 1-second warning tone to the calling party. After hearing the tone, the caller can either let the call proceed or hang up and try again later when a less expensive facility is available.

Nontoll Code

The nontoll code, except when an ARS call times-out in queue, only allows access to nontoll preferences that are also allowed by the FRL.

ARS Toll Restriction

Toll restriction for ARS is a class-of-service restriction that is used with the ARS toll code. When ARS toll restriction is assigned to a caller's class of service, the caller is not allowed to place calls using the ARS toll code. When the toll code is dialed by a restricted caller, the switch denies the call with intercept treatment. (ARS toll restriction is assigned to a class of service by entering a "1" in Procedure 010, Word 3, Field 22).

Time-of-Day Routing Plans

Long-distance toll charges vary from day-to-day and from hour-to-hour. To take advantage of these variations, the switch can automatically change the ARS routing plan as many as six times a day. Three ARS time-of-day routing plans are available, and each plan can have up to 64 routing patterns. The difference between routing plans is usually the order of the preferences in routing patterns. Typically, a routing plan will optimize call routing for a particular period of time. For example one plan will optimize call routing during normal business hours, another plan will optimize call routing after business hours, and the third plan will optimize call routing on weekends and holidays.

Clocked Manual Override

An override schedule (assigned in Procedure 287) can suspend the automatic plan-change times. The switch administrator specifies the parameters for the override capability, called Clocked Manual Override, including start time (Fields 1 to 3), stop time (Fields 5 to 7), and the routing plan to take effect (Field 4). This override capability lets the customer take advantage of low holiday toll rates. When a holiday falls on a weekday, for instance, the switch can be preprogrammed to use the weekend routing plan rather than the plan normally used during business hours.

Manual Override

Another override capability, Manual Override, allows an attendant to change the routing plan from the attendant console. The manually selected routing plan remains in effect until the attendant selects another plan or cancels Manual Override.

ARS Private-Network Routing

Tail-End Hop Off

A call-routing attribute called tail-end hop off, allows ARS calls to route partially over private-network trunks instead of initially routing over public-network trunks. Using tail-end hop off can significantly reduce long-distance bills.

ETN Arrangement.

As a tail-end hop off call begins in an ETN network, the ARS patterns at a main switch contain access tie trunks to the higher-level tandem switch and ARS patterns at the tandem contain intertandem tie trunks to other tandems. At either an ETN main or tandem, the 10-digit NPA-NXX-XXXX (including the home NPA, when necessary) is outpulsed over the tie trunk. [Tandems also attach the TCM(s) to the 10-digit number]. The receiving tandem recognizes these digits as a public-network number. Then, based on its own routing pattern, this tandem either routes the call to another tandem or allows the call to "hop-off" the private network to public-network facilities.

Tandem Tie-Trunk Arrangement:

Tail-end hop off can also be used in more traditional Tandem Tie-Trunk Networks. Instead of using ETN tie trunks (40-series), this type of network is interconnected with standard (30-series) tie trunks (usually Trunk Type 36). Most, if not all, of the switches in this arrangement ***do not*** have AAR assigned. In order for tail-end hop off to work in this arrangement, all of the switches (except the last switch in the sequence, the "hop-off" switch) ***must have*** the ARS feature and the Tandem Tie Trunk field (Procedure 275, Word 1, Field 15) assigned.

Like the ETN arrangement, each originating and tandeming switch selects the desired tie-trunk group based on its own routing patterns. Unlike the ETN arrangement, ARS subnetwork trunking at each originating and tandeming switch (except, perhaps, the last tandeming switch) needs to insert an ARS dial access code in front of the originally dialed digits. If the hop-off switch ***does not*** have ARS assigned, ARS subnetwork trunking at the last tandeming switch should insert the dialed digits with the correct public-network trunk-group dial access code, followed by a pause for the hop-off switch. Otherwise, the last tandeming switch can insert an ARS dial access code as usual.

International Call Routing.

For System 85 switches prior to R2 V3, Issue 1.4, the ARS feature cannot route IDDD (International Direct Distance Dialing) numbers over private-network tie trunks to another System 85 or DEFINITY Generic 2 switch. As long as TCM(s) are not included in the digit stream, ARS can route IDDD numbers to a System 75, DEFINITY Generic 1, or the SDN (Software Defined Network).

For System 85 switches prior to R2 V3, Issue 1.4, a frequently called private-network switch outside the United States can be assigned its own RNX to provide private-network routing that includes the FRL and / or the Conditional Routing TCM. At an appropriate tandem switch within U.S. boundaries, the tandem switch can either route international calls over a special tie-trunk group (perhaps a satellite link) to the international PBX or allow these international calls to Tail-End Hop Off to an International Toll Office.

Using either scenario, the tandem must perform the necessary AAR subnetwork trunking to change the RNX to the appropriate digits. If a special tie-trunk route is selected, the subnetwork trunking may involve deleting the dialed RNX and sending just the extension number. If Tail-End Hop Off to an international toll office is used, the subnetwork trunking would usually involve deleting the RNX, inserting "011", the Country Code, the City Code, and the local CO's exchange, and then sending the originally dialed extension (with the necessary pauses).

For System 85, R2 V4, Issue 2.0 and DEFINITY Generic 2.1, Issue 2.0 switches, selective international (011) call routing is available. With this enhancement, specific digit strings (up to 256 strings) contained in a "01X exception list" point immediately to a routing pattern that can contain both private and public trunking facilities.

CAUTION: *Laws governing the use of "special" international tie trunks vary from nation to nation and are subject to change. Also, laws governing the use of Tail-End Hop Off at receiving PBXs outside national boundaries vary from nation to nation and are subject to change. It is the responsibility of the customer's switch administrator to understand and comply with the applicable regulations.*

Two reasons the receiving tandem can elect to route the call to another tandem include:

- The next tandem is closer to the public-network destination.
- The next tandem is located in a time zone where lower long-distance rates are currently in effect.

10- to 7-Digit Conversion

An attribute of ARS, called 10- to 7-Digit Conversion, forces ARS calls destined for private-network locations to route over private-network trunks. When a user dials a 10-digit public-network telephone number that is marked for conversion, the switch converts the number to the 7-digit private-network telephone number and passes the call to the AAR software for routing. The call's final destination is ***primarily intended*** to be in the private network but can also be in the public network. If the call's destination is in the public network, subnetwork trunking (refer to the AAR feature description) converts the private-network number (RNX-XXXX) back to a public-network number [(NPA)-NXX XXXX]. Using the private network in this way can significantly reduce long-distance telephone bills.

NOTE: The switch does not perform ARS subnetwork trunking on an outgoing trunk group unless the trunk group's "Network Trunk" field is assigned to "1" and the "Main/Tandem" field is assigned to "0" in Procedure 103, Fields 3 and 4.

Unauthorized Call Control

The switch can also block certain telephone numbers on a more specific basis [for example, 900 (DIAL-IT) numbers or 212-674-1XXX]. Unauthorized Call Control assignments (in Procedure 313, Word 1) let the switch administrator prohibit calls based on the first seven, eight, nine, or ten digits of these destination codes. As a group, these controlled numbers have an FRL (assigned in Procedure 275, Word 3), and therefore a caller using an originating facility (or authorization code) with a high enough FRL can still call numbers marked for call control.

IXC (Interexchange Carrier) Access

ARS can provide equal access to any IXC (for example, a vendor providing long-distance facilities). The customer can specify by route the particular IXC used for extending calls to given locations. Subnetwork trunking is used to modify the destination code as needed. Up to 20 digits can be inserted in front of the destination code to accommodate an IXC access code and IXC security code.

Generalized Route Selection

Call Categories for Partitioned Switches

Figure 21–3 is a simplified picture of partitioned ARS routing. Partitioned ARS provides more flexible access to the 64 ARS patterns or Routing Designators. Several attributes of this figure call for discussion.

- More than one extension partition can route ARS calls via the same call category.
- Each call category points to 64 "routing designators" that correspond to the dialed digits (the way ARS patterns do in an unpartitioned System 85 or DEFINITY Generic 2).

- In turn, each routing designator points to a "pattern" that has not been previously assigned to that call category's routing designator.

  NOTE: The routing designators for Call Category 0 all point to patterns with the ***same*** numbers as the routing designators themselves. This is acceptable. In fact, this relationship is the default for unpartitioned switches. However, if desired, the relationship can be assigned differently.

- Each pattern can have as many as 16 "preferences" (as shown for Pattern 2 in Figure 21-3). These preferences are actually trunk groups arranged in descending order of desirability for that specific pattern.

- Patterns 10, 20, and 64 each contain three preferences: Trunk Groups 18, 36, and 40. But, these three preferences are arranged in a different order for each pattern. This type of arrangement is where some of the increased ARS flexibility resides. In this way, when these trunk groups are shared, different extension partitions can access the same trunk groups in different preferential orders.

  However, a more basic flexibility is derived from another attribute of this routing scheme. As shown in the drawing, different extension partitions can access different ARS patterns even though the same public-network digits are dialed.

- An extension partition does not need to be allowed use of every trunk group in every pattern that the partition can use. A partitioned System 85 or DEFINITY Generic 2 checks the trunk group / partition assignments individually within each pattern.

- Whenever a partition's voice terminal cannot use a trunk group in these patterns, the trunk group is skipped by the ARS / Tenant Services software. If the voice terminal cannot access any of the trunk groups, the switch denies the call with Intercept Treatment.

- The ARS routing in Figure 21–3 is based on 3-digit NXXs. This is not a limitation of partitioned ARS routing. Rather, the drawing shows 3-digit NXXs for simplicity. The partitioned ARS software can also screen 3-digit NPAs and 6-digit NPA-NXXs.

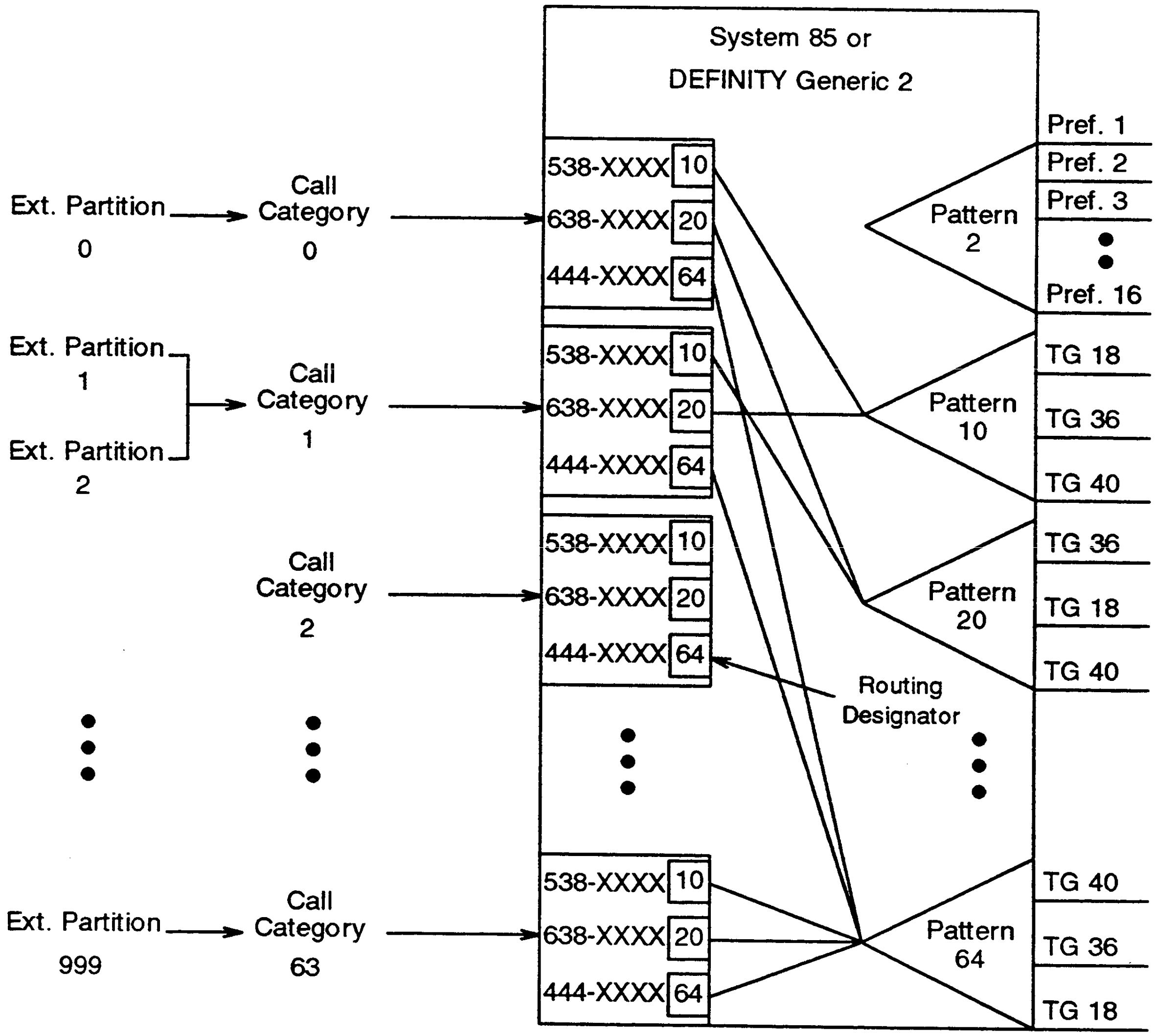


Figure 21-3. ARS Routing in the Tenant Services Environment

NOTE: Patterns can also have trunk groups that are not assigned to other patterns. This arrangement is not shown in this figure.

Bearer Capability

Bearer capability applies to all calls and call support facilities but is of primary significance to data calls. Bearer capability was first used on the System 85 with the Release 2, Version 4 ISDN—PRI feature. In this version it is assigned to ISDN facilities as part of the COS and consists of five bearer capability codes as follows:

Bearer Capability Code	**Type of Traffic Supported**
0	Voice and Voice Grade Data
1	Mode 1 data, 56 Kbps allowed
2	Mode 2 data, 64 Kbps allowed
3	Mode 3 data
4	Mode 0 data.

In ***DEFINITY Generic 2,*** the power and versatility of ISDN messaging is used to determine bearer capability needs for call routing, both externally (over trunks) and internally (for example, Modem Pooling). Identification of call type and resource requirements is based on the best available information, obtained as follows:

- ***Call Setup Messages***

  Call control information contained in the ISDN call setup message associated with each specific call is the primary source of information on protocol and call handling facility requirements. Call control setup messages (originating from ISDN facilities) contain IEs that indicate the type of call (such as, voice, Mode 0 data, etc.), protocol used, data rate, and other information needed to identify required resources.

- ***Optional Query***

  Data modules (both ISDN and DCP) have the ability to respond to requests for additional information from the switch. For information that is needed but not available in a specific call setup message, this optional query ability is used.

- ***Default Values***

  The last resort for determining resources needed for a specific call is the customer-administered ***BCCOS*** (Bearer Capability Class of Service) assigned to the origination points, carrier and support facilities, and termination points. This BCCOS provides default requirements and characteristics for specific ISDN facilities. The default BCCOSs are associated with the facility (trunk, etc.) and not a specific call.

If call processing must, for any reason, use only the BCCOS default values (unchanged), the resulting DEFINITY Generic 2 call handling will be consistent with call handling provided by a System 85, Release 2, Version 4 switch for the same call. In effect, the default processing for DEFINITY Generic 2 equates to the basic bearer capability handling provided by the System 85, Release 2, Version 4 switch.

Switch actions based on BCCOS are specified in administration (Procedure 014, Word 1, Fields 4 through 13). These switch actions determine how a call with a specific BCCOS will be handled by each preference in each ARS pattern. Three specific switch actions are used:

- Circuit switch the call
- Insert a Modem Pooling conversion resource
- Block the call.

With Bearer Capability, the ARS search algorithm operates essentially as follows:

1. ***Preferred Option***

   First the search looks for a preference that calls for the switch action ***circuit switch the call.*** If a preference is found that provides this action, and the FRL allows that preference to be used, and a trunk is available, then that preference is selected for routing the call.

2. ***Acceptable Option***

   While looking for a preference that calls for circuit switching the call, the search algorithm also checks for a preference that calls for ***insert a Modem Pooling conversion resource.*** If a preference is found that calls for the action ***insert a Modem Pooling conversion resource,*** that preference is recorded for future reference if needed.

3. ***Exercising the Alternative***

   If the search for the ***preferred option*** is not successful (no available trunk with a usable FRL is found), the algorithm tries to connect the call to an ***acceptable option*** trunk if one is available.

4. ***Unacceptable Option***

   Blocking the call is an unacceptable option. All other alternatives (Authorization Code, FRL raising, Queuing, etc.) must first be exhausted. No attempt will be made to connect a call to a trunk that will block that call.

More details on specific BCCOSs, their characteristics and applications are provided in the Bearer Capability feature description.

Flow Diagram

To help conceptualize this complex feature, a flow diagram describing the ARS feature is provided in Figure 21-4. This diagram does not show all of the decisions made in the ARS software. The diagram does, however, unify many of the different ARS functions.

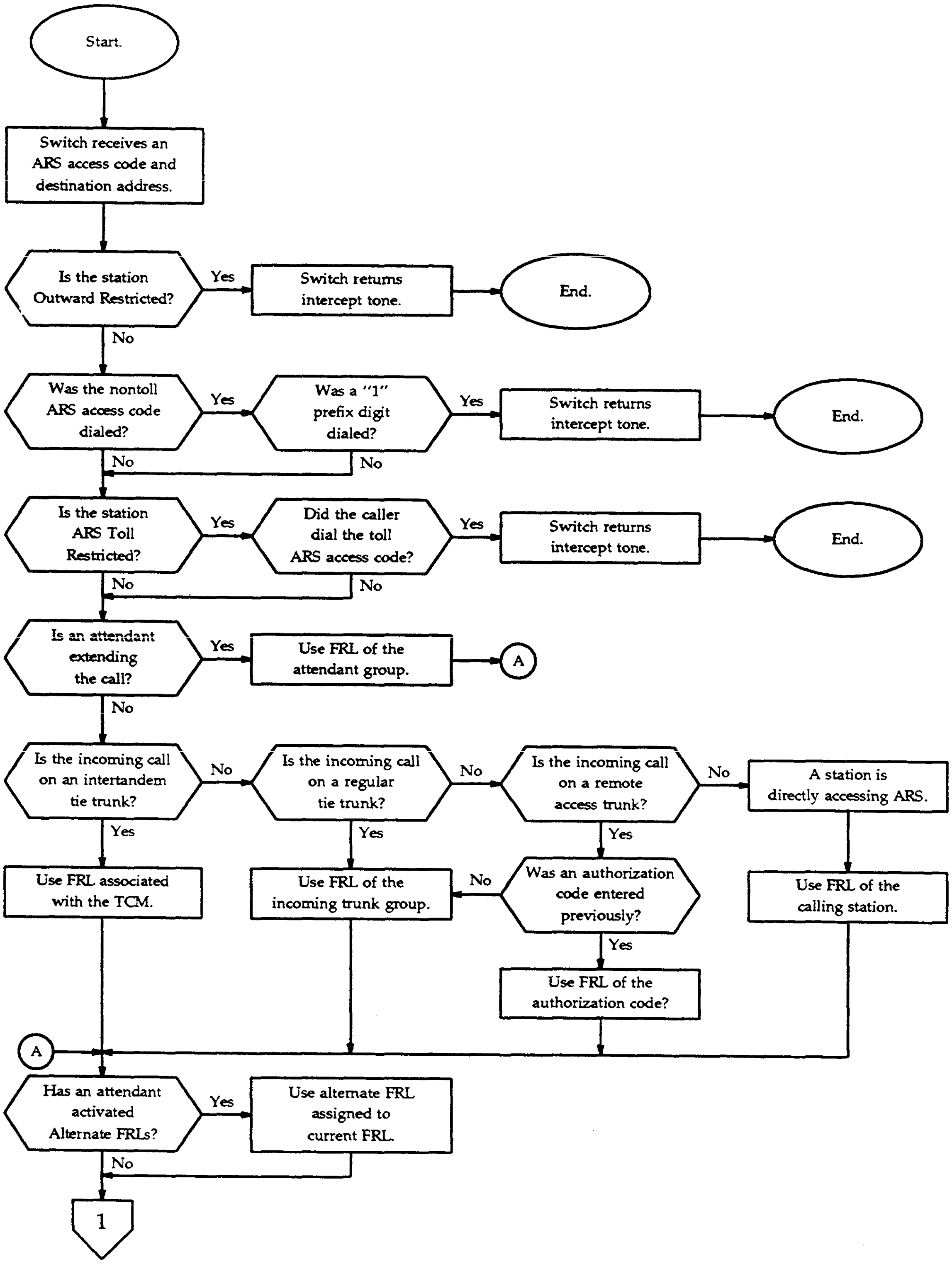


Figure 21-4. ARS Feature Flow

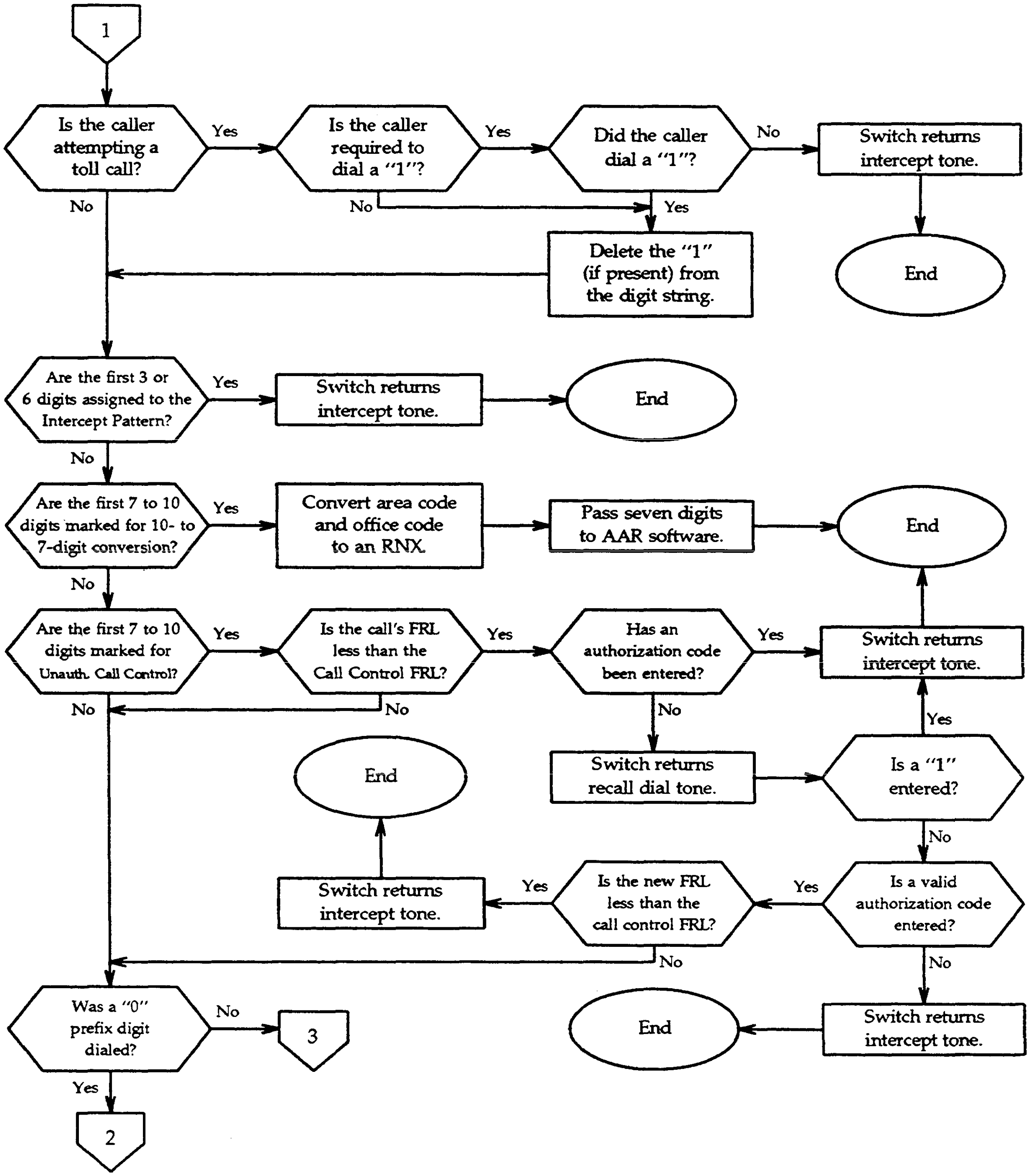


Figure 21-4. ARS Feature Flow (Sheet 2 of 8)

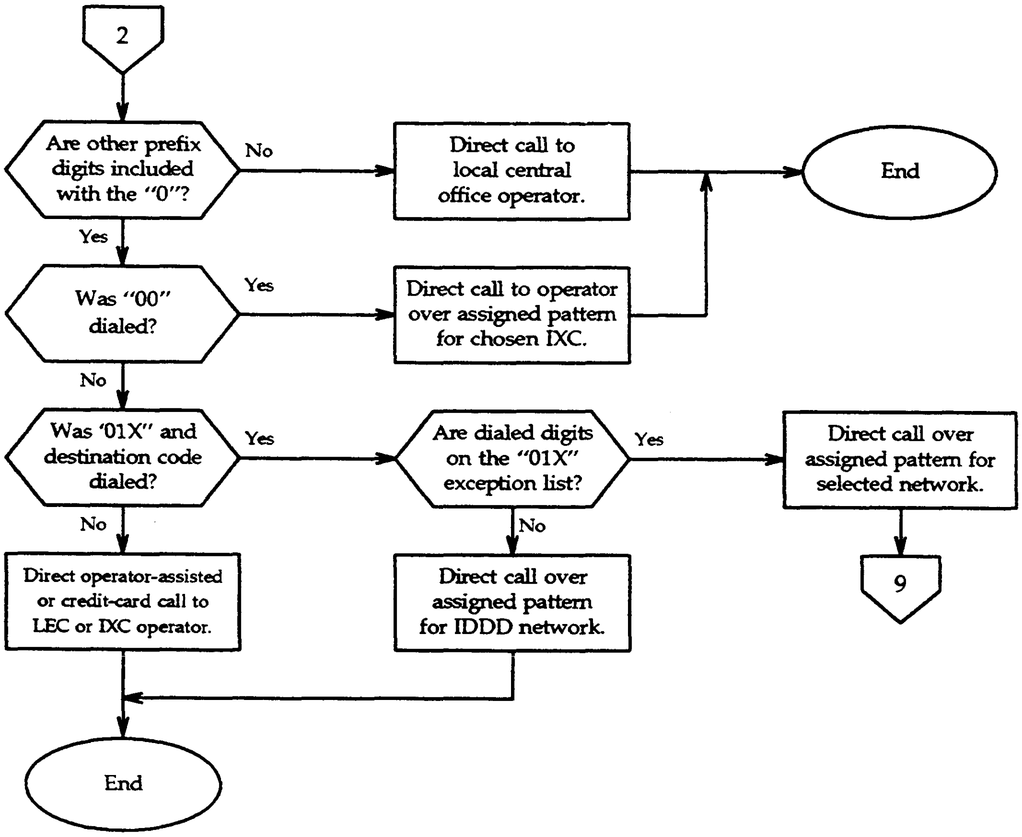


Figure 21-4. ARS Feature Flow (Sheet 3 of 8)

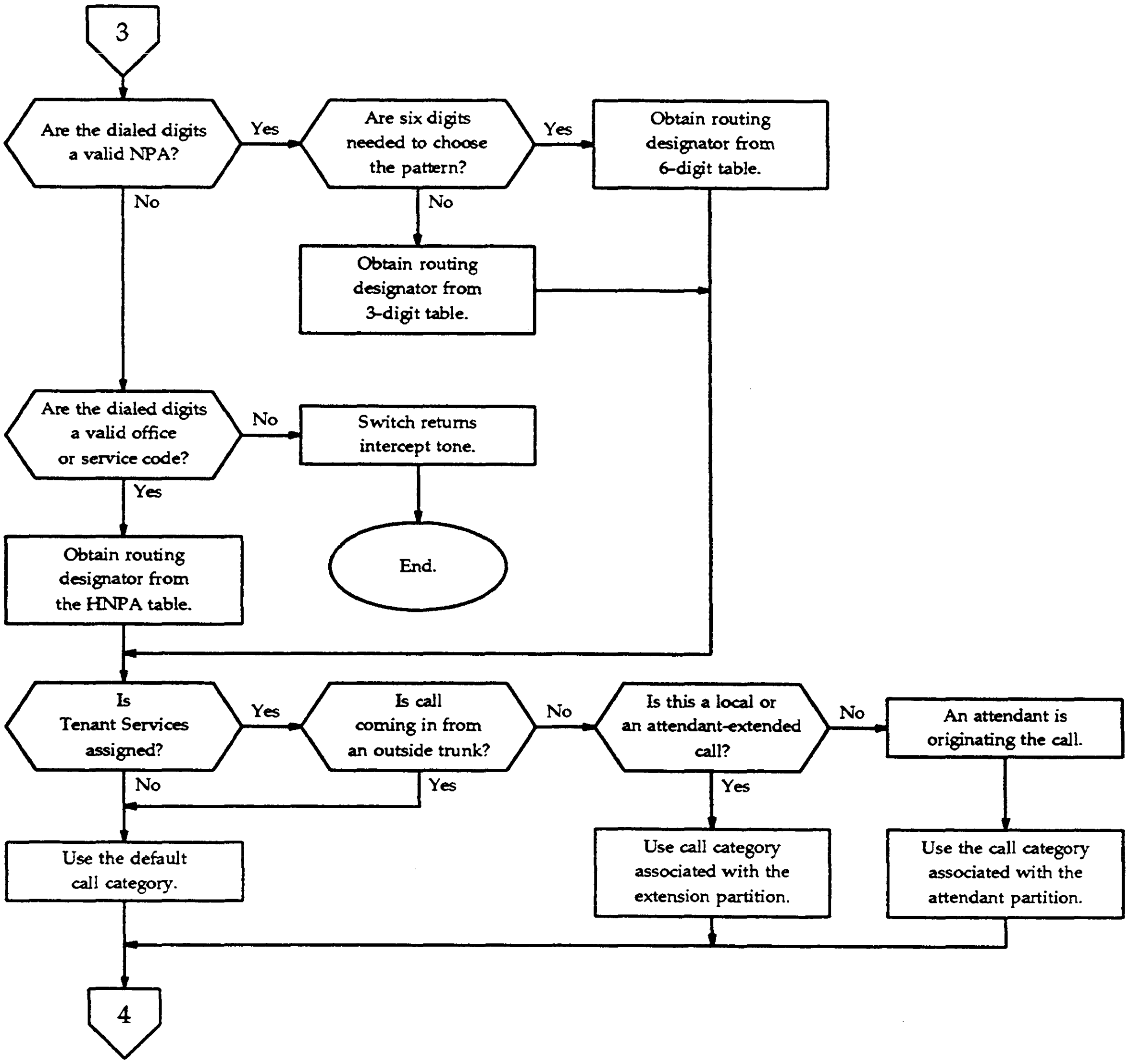


Figure 21-4. ARS Feature Flow (Sheet 4 of 8)

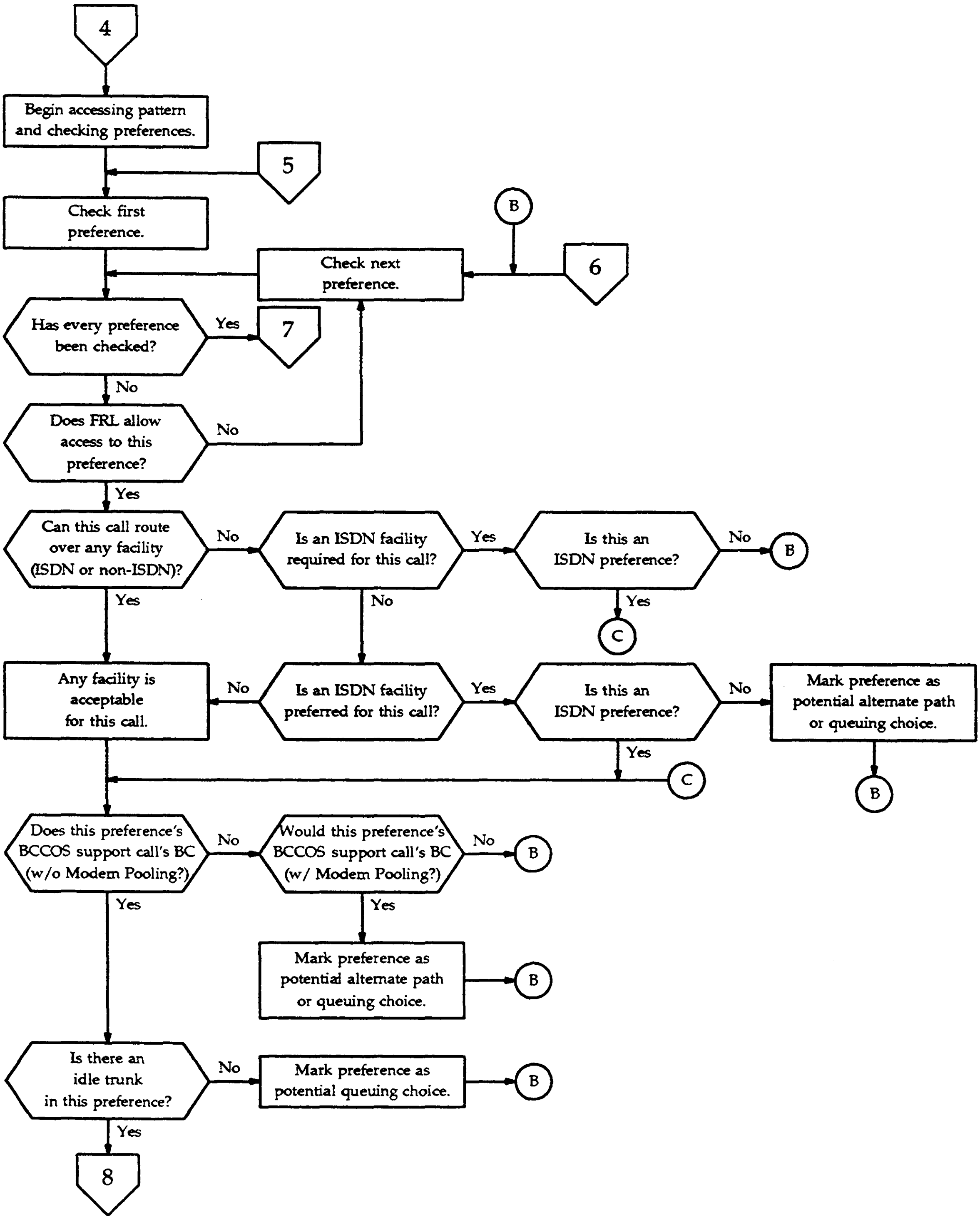
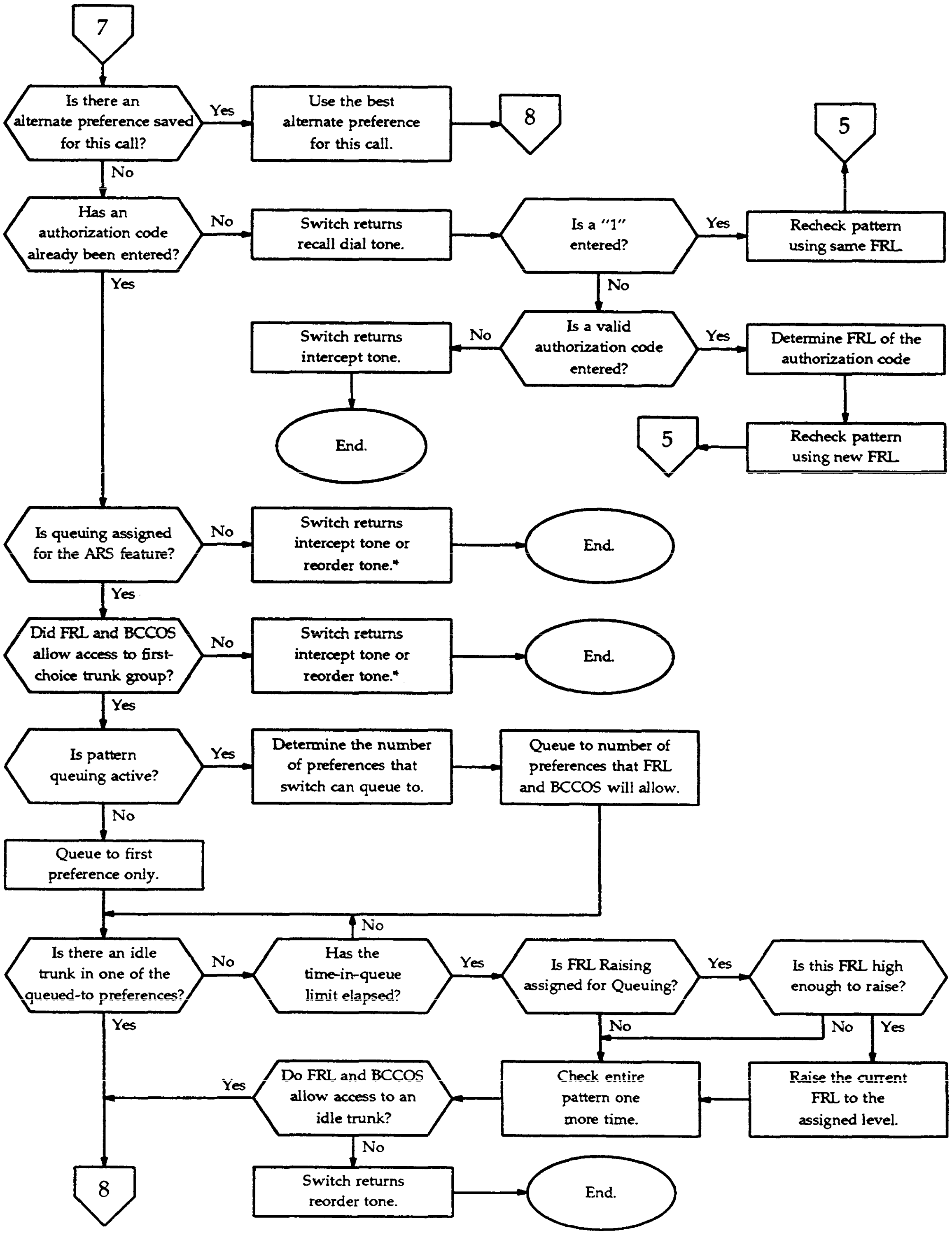


Figure 21-4. ARS Feature Flow (Sheet 5 of 8)



* Intercept tone if the call is not allowed access to any preferences. Reorder tone if the accessible preferences are busy.

Figure 21-4. ARS Feature Flow (Sheet 6 of 8)

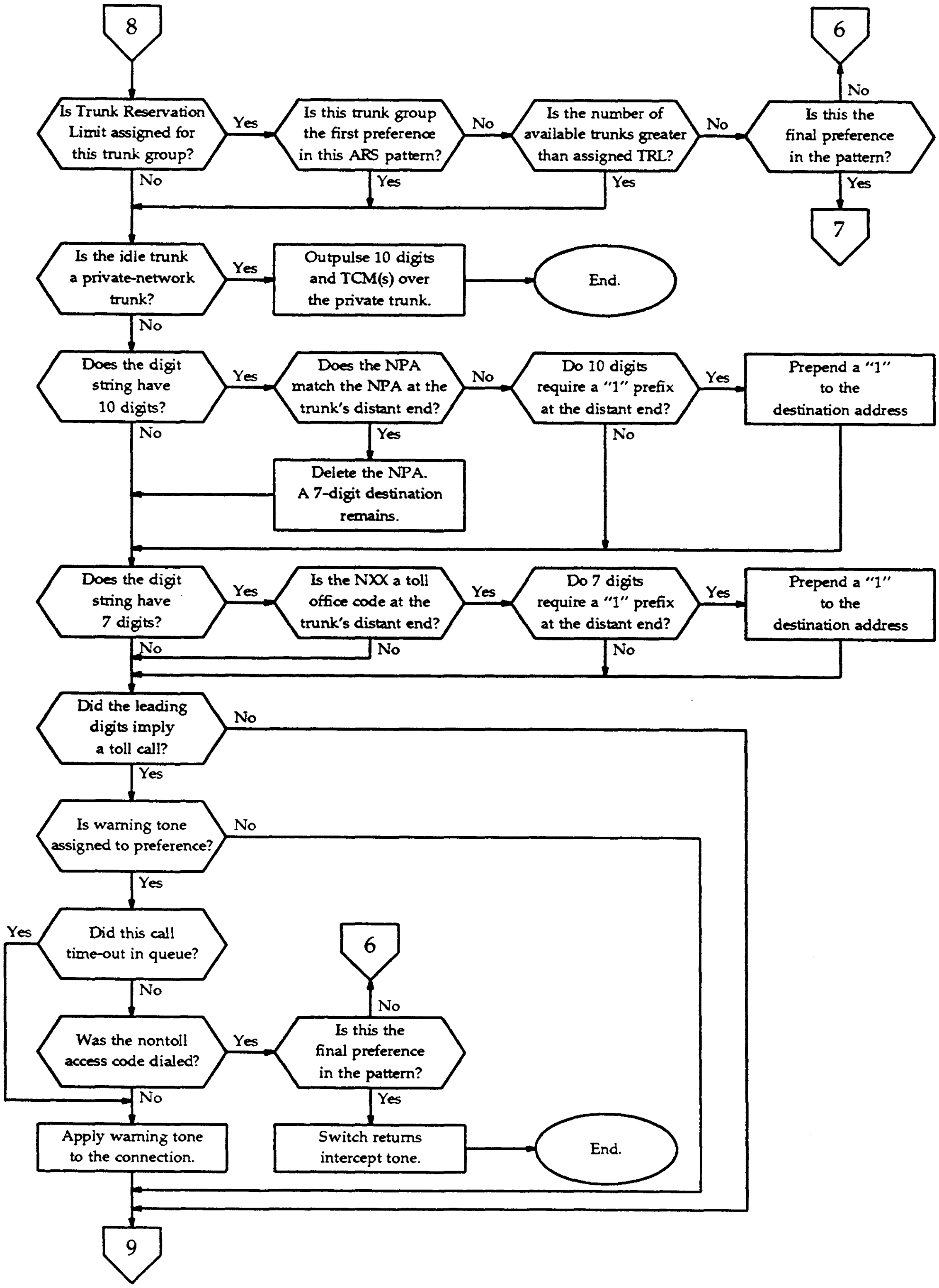


Figure 21-4. ARS Feature Flow (Sheet 7 of 8)

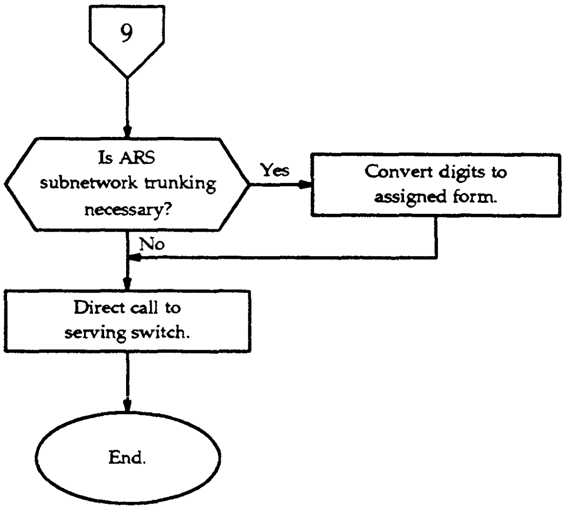


Figure 21-4. ARS Feature Flow (Sheet 8 of 8)

User Operations

For voice- and data-terminal users, the ARS feature is fully automatic once administered. No user operations are required.

However, a System 85 or DEFINITY Generic 2 attendant can activate Manual Override to change the current time-of-day plan for ARS routing.

To Activate Manual Override

An attendant should:

1. Press an idle loop button.
2. Press the **[DISPLAY ROUTE PLAN]** button.

   or

   Dial the Network ARS Plan Switch access code. [Alphanumeric display shows the preset plan number, and the switch returns dial tone].
3. Dial the desired plan number (from 1 to 3). [Alphanumeric display shows "A" and the new plan number, and the DISPLAY ROUTE PLAN lamp lights].

To Deactivate Manual Override

An attendant should:

1. Press an idle loop button.
2. Press the **[DISPLAY ROUTE PLAN]** button.

   or

Dial the Network ARS Plan Switch access code. [Alphanumeric display shows "A" and the current plan number, and the switch returns dial tone].

3. Dial **[0]** as the desired plan number. [Alphanumeric display shows the preset plan number].

Considerations

"Standard Network" Field in Procedure 276

Field 1, the Standard Network field in Procedure 276, need not be assigned to provide the ARS feature. This field is assigned to provide AAR (Automatic Alternate Routing) and a private-network Uniform Numbering Plan for tandem switches in an ETN (Electronic Tandem Network).

However, to provide 10- to 7-Digit Conversion function for ARS, the Standard Network field ***must*** be assigned in Procedure 276. This assignment is necessary since 10- to 7-Digit Conversion transforms a public-network number into a private-network number and passes the call to the AAR software.

Inferred Routing

Prior to R2 V4 System 85, the ARS dial access code(s) can have from one to three digits. Beginning with System 85 R2 V4, the ARS dial access code can have from one to four digits.

However, whenever a System 85 or DEFINITY Generic 2 infers an ARS dial access code for incoming calls from over a trunk group, the ARS dial access code must be a single digit between "0" and "9". This inferred dial access code, which is usually assigned as "9", cannot be assigned as "*" or "#". This limitation exists because the ARS Prefix Digit in Field 9 of Procedure 103 is limited to a single digit.

FRL Access to ARS Routing Patterns

In Release 2, System 85 and DEFINITY Generic 2, the caller's FRL need not be greater than or equal to the FRL of the first preference in an ARS pattern. The caller can access another preference in the pattern when a subsequent preference has an FRL that is less than or equal to the caller's FRL. (Therefore, the trunk groups in an ARS routing pattern need not be arranged with ascending FRLs).

FRLs and Intertandem Tie Trunks

The preceding flowchart shows that receiving System 85 or DEFINITY Generic 2 tandem switches always use the FRL associated with the TCM (for incoming calls from over intertandem tie trunks) to select an outgoing ARS preference. However, this flowchart presents a simplified picture of the preference-selection process at the receiving tandem. The following paragraphs are a more detailed description of this process.

The FRL TCM is the last (or the next to last) digit that the receiving tandem gets, and the ARS software does not wait for this digit to begin its preference-selection process. During the ***first*** check of the preferences in the assigned pattern, the receiving tandem actually uses the FRL assigned to the incoming intertandem trunk group in Procedure 103, Field 2. (If this FRL is not assigned, the FRL value for the trunk group defaults to "0").

If the ARS software ***can*** select a preference using the trunk group's FRL, it does. If, however, the ARS software selects another intertandem tie trunk to continue the routing the System 85 or DEFINITY Generic 2 sends the FRL TCM that was subsequently received when the digits are sent over this outgoing trunk group.

If the ARS software ***cannot*** select a preference using the trunk group's FRL, the ARS software needs to check the pattern's preferences again. By this time the TCM FRL has arrived, and the System 85 or DEFINITY Generic 2 unconditionally replaces the trunk-group FRL with the TCM FRL. From this point on*, the ARS feature uses the TCM FRL to select a preference and sends the TCM FRL with the outgoing digits.

Authorization Codes and Intertandem Tie-Trunk Calls

For calls where an authorization code has not already been entered, the preceding flowchart shows that a receiving tandem switch always prompts for an authorization code (when assigned) if the default FRL cannot access an available facility and a higher FRL would allow access to additional preferences. However, this flowchart presents a simplified picture of ARS authorization-code prompting.

For incoming calls from over intertandem tie trunks (Procedure 103, Fields 3 and 4 assigned as "1") where the ARS feature tries to select an outgoing preference, an ETN tandem switch does not prompt for an authorization code. First, the calling party may have already entered an authorization code at the previous tandem. Also, the list of valid authorization codes could be different at each tandem within the network, and the calling party would have no way of knowing which tandem is requesting the code.

Therefore, when the FRL for these calls is too low to access an available trunk facility, the ETN tandem does one of the following:

- Queues the call to busy accessible preferences in the preference depth (when ARS Queuing is assigned)
- Returns reorder tone if the accessible preferences are busy (when ARS Queuing is not assigned)
- Returns intercept tone if the FRL is too low to access any preference or the first queuing preference.

* Until the time-in-queue limit elapses, and FRL Raising can be invoked.

Shared Software Tables

The 10- to 7-Digit Conversion, Unauthorized Call Control, and the International Call Routing functions of the ARS feature share the same table structure (containing 6 tables) in the translation portion of switch memory. The following list shows the five limiting factors for this pair of ARS functions.

The shared table space allows for:

- Any number of unique 3-digit NPAs,

  plus

- 500 unique 6-digit NPA-NXX combinations,

  or

- 2048 unique 7-digit NPA-NXX-X combinations,

  or

- 2048 unique 8-digit NPA-NXX-XX combinations,

  or

- 2048 unique 9-digit NPA-NXX-XXX combinations,

  or

- 2048 unique 10-digit NPA-NXX-XXXX combinations

  plus

- 256 unique 11- to 18-digit combinations for international call routing.

This table structure optimizes the System 85 or DEFINITY Generic 2 processor's speed of access without unduly limiting the memory capacity of the structure.

A switch administrator will not usually be able to fill all six tables at the same time. Rather, the entire table structure fills when ***any one*** of the five limiting conditions is reached. As a minimum, the table structure can hold:

- 2048 10-digit numbers,

  or

- 2048 blocks of 10 numbers (effectively 20,480 numbers),

  or

- 2048 blocks of 100 number (approximately 200,000* numbers),

or

- 2048 blocks of 1000 numbers (approximately 2,000,000† numbers).

Normally within a private network, a mixture of block sizes are assigned yielding a table capacity somewhere between 2048 and 2,000,000 numbers.

Therefore, to optimize the capacity of this structure, the switch administrator should assign controlled and/or converted numbers in ***blocks*** (that is, as 7-, 8-, or 9-digit combinations) whenever possible. Conversely, an effective way to free up the structure's space (if necessary) is to begin by removing 10-digit combinations and then working back to 9-digit combinations, etc.

Hard and Soft Processor Swaps

The contents of the ARS routing plans are stored in a translation portion of switch memory. Therefore, if Routing Plan 3 is active when a hard processor swap occurs, the same routing plan is active after the hard swap is finished.

The contents of the ARS routing patterns are stored in a translation portion of switch memory. Therefore, these patterns will endure a hard processor swap. The contents of the shared tables for Unauthorized Call Control and 10- to 7-Digit Conversion are stored in a translation portion of switch memory. Therefore, these assigned digit strings will endure a hard processor swap.

ARS queues are stored in a status portion of memory. Therefore, if an ARS call is queued to a pattern when a hard swap occurs, the call is never routed to the public network. The queue is cleared.

Stable ARS calls will endure a hard processor swap. However, an ARS call cannot be placed during a hard swap.

The ARS feature operates normally during a soft processor swap.

Interactions with Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

* The exact limit is not 204,800. There are some losses in this table structure due to overlap.

† The exact limit is not 2,048,000. There are some losses in this table structure due to overlap.

Attendant Control of Trunk Group Access

The ACTGA feature takes precedence over ARS. A call directed by ARS to a controlled trunk group is routed to an attendant.

Authorization Codes

The Authorization Codes feature can be used to raise the FRL of a call if a default FRL is not high enough and a higher FRL may be successful in finding an available and accessible trunk. Authorization codes (if the feature is available) can be used for any locally originated station call. Authorization codes may also be used with Remote Access calls as long as the specific authorization code has off-net origination permission. Off-network permission is available if the network access flag for that authorization code is set to "1" in Procedure 282, Word 1, Field 3.

AAR (Automatic Alternate Routing)

If a caller dials the AAR dial access code and then dials a public-network telephone number of the form "NPA-NXX-XXXX", the switch does not immediately deny the call. Instead, the AAR software passes the ten digits to the ARS software for routing. During this process, however, the caller never specifies either toll or nontoll access to the public-network facilities by dialing the toll or nontoll ARS access code. In this situation, the ARS software selects a preference for the call as if the calling party had dialed the ***toll*** ARS access code.

For International (01X) Calls, selective international call routing (or 01X screening) applies to all calls using the "01X" prefix, whether the AAR or ARS access code is used. The 01X exception list is administered only once for the ARS feature and in fact cannot be administered separately for the AAR feature.

Bearer Capability

Bearer capability defines the type of calls that a carrier (such as a trunk) or service (for example, modem pool or host access port) facility can support. In ***System 85, Release 2, Version 4,*** ARS routing searches for a preference that supports the bearer capability requirement of the call being routed. Five bearer capabilities are recognized:

Bearer Code	Type of Traffic That Can Be Carried
0	Voice and voice-grade data allowed
1	Mode 1 data, 56 Kbps allowed
2	Mode 2 data, 64 Kbps allowed
3	Mode 3 data
4	Mode 0 data.

If a request is made for a trunk group to route a Mode 2 call, only preferences that can support Mode 2 data are searched.

BCCOS (Bearer Capability Class of Service)

In DEFINITY Generic 2, routing is based on BCCOS. Two methods are available for routing the call:

1. The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (for example, Mode 2 data, 1200 bps, restricted channel, etc.). If a match is found and a trunk is available, other factors such as FRL permitting, the action taken is to circuit-switch the call.
2. If a match is not found, the algorithm attempts to connect the call to a preference for which the action to take is not block the call. With currently available options, this would be a preference where the action is to insert a modem pool.

Bridged Call

The ARS Toll Restriction is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When ARS Toll Restriction is assigned to a ***shared extension,*** the restriction applies to every image of the extension.

Call Detail Recording

When optional account codes are used, a public-network caller (after dialing the Account Code access code) ***can*** dial an account code before dialing the ARS access code.

When FEAC (Forced Entry of Account Codes) is assigned to the system for ARS, a public-network caller (after dialing the Account Code access code) ***must*** dial an account code before dialing the ARS access code.

When FEAC is assigned to specific trunk groups in an ARS pattern, a public-network caller must dial an account code (after dialing the Account Code access code) for access to these FEAC trunk groups. If the ARS user does not dial an account code, the FEAC trunk groups are skipped within the ARS pattern.

Call Forwarding—Follow Me

When administering Call Forwarding Off Net, all three ARS plans should be administered to contain patterns with at least one preference in Procedure 309, Word 1. Otherwise, when activating the Call Forwarding feature for Off Net routing, the ARS access code cannot be used as part of the destination's telephone number. Rather, the appropriate trunk-group dial access code would have to be dialed.

Also, when administering Call Forwarding Off Net, the desired local office codes should be specified in the ARS Toll Table (Procedure 309, Word 2 and Procedure 309, Word 1, Field 9). Otherwise, when activating the Call Forwarding—Off Net feature, an office code that is not specifically assigned as ***local*** is presumed by the Call Forwarding—Off Net software to be a ***toll*** office code. Since forwarding to the toll network is not provided, the switch would return intercept treatment.

Call Vectoring

In Procedure 010, Word 3, an FRL can be assigned to VDNS for use with the "route to" command. The FRL of a VDN is used to determine whether the call is allowed to route over available public network facilities.

The ARS Toll Restriction does not limit the routing of "route to" steps to destinations outside the switch. If ARS Toll restriction is assigned to a VDN's class of service, this assignment is ignored.

Data Call Setup

The ARS feature can be used to improve the routing of data calls as well as voice calls. However, like the AAR feature, if ISDN is not used, separate routing patterns must be used to prevent voice calls from terminating on data-only extensions and vice versa.

ISN (Information Systems Network) Interface

ISN data stations can use the circuit-switch feature ARS when placing public-network calls through the circuit switch. When this is done, Modem Pooling is required if the trunk groups to be accessed contain analog trunks.

ISDN—PRI (Primary Rate Interface)

Calls placed over ISDN facilities use the ARS or AAR features for ISDN access. To work for ISDN, the ***bearer capability classes*** must be assigned. In this way placing an ISDN call is transparent to the user. ARS and AAR patterns are selected according to the calling party’s COS (class of service) and the trunk-group bearer capability.

COS (Class of Service)

The AAR and ARS pattern searches work differently for ISDN calls based on the calling station’s COS. For ISDN, a calling party COS specifies one of three options as follows:

ISDN Facilities Required

If ISDN facilities are required by a user's class of service, only ISDN end-to-end facilities are to be used for AAR or ARS routing. If ISDN end-to-end connections are not available, the call will either queue or receive reorder tone.

ISDN Facilities Preferred

If ISDN facilities are preferred but not required by the COS, ISDN facilities are checked first, followed by a check for non-ISDN facilities.

Any Facilities Available

Any available facilities can be used to complete the call. In this case AAR and ARS pattern searches work like they do for a non-ISDN call.

IXC (Interexchange Carrier Access)

The subnetwork trunking function of the ARS feature is used to implement the IXC selection automatically. An IXC code (10XXX) cannot be dialed by a station user, and the IXC digits cannot be passed over the ETN network.

LND (Last Number Dialed)

When the LND feature is used to redial an ARS call, the LND feature sends all of the originally dialed digits (including the ARS access code) for routing. If an ARS function such as: 10- to 7-digit conversion, subnetwork trunking, or IXC (Interexchange Carrier) Access needs to modify the digit stream, the ARS feature performs these modifications each time the number is redialed. For switch security the LND feature does not store or redial Authorization Codes.

Look-Ahead Interflow

For System 85 and DEFINITY Generic 2.1 switches, the ARS feature is required to route Look-Ahead Interflow calls through the public network. For Generic 2.2 switches, the ARS feature is replaced by the WCR (World Class Routing) feature.

The "Standard Networking" field (in Procedure 276) must also be assigned. This is required provide the 10- to 7-Digit Conversion function needed to route Look-Ahead Interflow calls (with public-network destinations) through the private network.

When the ARS feature is used to route Look-Ahead Interflow calls, the digit contents of a vector-group list item for a "route to" step must conform to the rules for DDD (Direct Distance Dialing). These rules allow one of three forms:

- NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- NPA (3-Digit Area Code) + NXX (3-Digit Office Code) + XXXX ((4-Digit Extension Number)
- International Telephone Number.

Besides conforming to the public-network rules for DDD, the vector-group list items for Look-Ahead Interflow "route to" steps must be prefixed by the 1- to 4-digit ARS dial access code and (if required in Procedure 275, Word 3) a "1" prefix digit for toll calls.

Besides conforming to the dialing plan for the public network, an ARS pattern must ***not*** be translated to the Intercept Pattern for the first three or six digits specified within the destination digits of a "route to" step. When this is done, the "route to" step is treated as having an invalid destination. If the "route to" step is the final effective step in the sending (or tandeming) vector, vector processing treats the step as a "stop" step. Otherwise, vector processing continues with the next sequential step in the vector.

The ARS pattern selection for "route to" steps conforms to the currently active ARS routing plan. (This is the case whether the currently active ARS plan was invoked by an automatic plan change, clocked manual override, or manual override). Therefore, whenever an ARS plan is active where the first three or six digits of a "route to" destination are translated to the Intercept Pattern, the "route to" step is considered to have an invalid destination. Also, whenever an ARS plan is active where a pattern's first-choice preference is not an ISDN—PRI trunk group, "route to" steps (if successful in diverting calls) will route the calls on a non-Look-Ahead basis.

The ARS routing of Look-Ahead Interflow "route to" steps can be blocked by Unauthorized Call Control. Whenever a vector-group list item for the Look-Ahead Interflow feature contains an ARS digit string that is marked for call control, vector processing at the sending switch either treats the "route to" step as a "stop" step (if the final effective step) or continues with the next sequential step in the vector.

For voice calls, the Bearer Capability Class of Service (BCCOS) is not usually a significant consideration. This is because voice calls are compatible with any carrier facility. However, the ARS feature does check the BCCOS for calls that are diverted outside the switch by a "route to" step. Therefore, when applicable, the BCC of outgoing (first-choice) preference must be compatible with the BCC in the local calling party's COS or the BCC assigned to the incoming trunk group.

At a sending (or tandeming) switch, ARS Toll Restriction does not limit the routing of Look-Ahead Interflow "route to" steps to an answering destination. If ARS Toll Restriction is assigned to a VDN's class of service, this assignment is ignored.

At a sending (or tandeming) switch, ARS Queuing and Pattern Queuing do not apply to Look-Ahead Interflow calls. Instead, if every preference in the first-choice AAR preference is busy, the Look-Ahead Interflow software will either retry routing the call at 2-second intervals (if final effective step) or continue vector processing with the next sequential vector step.

Since ARS Queuing does not apply to Look-Ahead Interflow calls, FRL Raising (which is invoked after the queue times out) would also not apply to Look-Ahead Interflow calls.

The Look-Ahead Interflow feature is compatible with the 10- to 7-Digit Conversion function of the ARS feature. When 10- to 7-Digit Conversion applies to the programmed destination of a Look-Ahead Interflow "route to" step, the 10-digit NPA-NXX-XXXX is converted to the corresponding RNX-XXXX, and the new digits are passed to the AAR software for routing.

A Look-Ahead Interflow "route to" command always selects the first preference of the ARS pattern needed to route the calls. Therefore, the ARS Trunk Reservation Limit (assigned in Procedure 103, Word 1) does not prevent interflow calls from accessing the preference. Rather, assigning an ARS Trunk Reservation Limit to the trunk group has the effect of reserving trunks in the preference to ensure the routing of Look-Ahead Interflow calls.

The Look-Ahead Interflow feature is compatible with ARS subnetwork trunking. For Look-Ahead Interflow calls, the ARS subnetwork trunking function can internally modify the acceptable digit formats for vector-group list items so that the next switch receives the expected digits.

As part of the Look-Ahead Interflow SETUP message, an intervening public-network switch is always requested to route the interflow call on an ISDN-Preferred basis. Then, according to its routing algorithm, the intervening switch gives ISDN routes first preference during its route-selection process.

If the public-network intervening switch cannot find an available ISDN route, the intervening switch returns a "Public-Network Interworking" message to the sending switch. Upon receiving this message, the sending switch will retry the "route to" step at 2-second intervals if the "route to" step is the final effective vector step. If it is not the final effective step, the sending switch will continue vector processing with the next sequential step.

Precedence Calling

AUTOVON (Precedence Capable) trunk group must not be included in ARS patterns. The ARS feature is not used for AUTOVON Access and Precedence Calling does not work for ARS calls.

Queuing

Based on administration, Queuing (including pattern queuing) can apply to ARS routing routing patterns. An incoming tie trunk that infers ARS routing cannot have ringback queuing.

Remote Access

The Remote Access feature can be used to originate ARS calls from off-net stations (such as public network telephones) as long as the appropriate class-of-service permissions and an adequate FRL are assigned. If the default FRL assigned for Remote Access calls is not adequate to select a require preference, the Authorization Code feature can be used to raise the FRL of the call in the same way as for a local station originated call. However, with Remote Access, the network access flag for the specific authorization code used must be set to "1" to allow that authorization code to be used with the Remote Access feature.

Restriction—Code Restriction

The Code Restriction feature has no effect on calls placed using the ARS feature. When a terminal user dials the ARS access code to access the public network, this access is controlled by the user’s FRL or by ARS Toll Restriction (not by the Code Restriction feature).

Restriction—Miscellaneous Trunk Restrictions

The Miscellaneous Trunk Restrictions feature does not restrict access to trunk groups in an ARS routing pattern.

Restriction—Toll Restriction

The Toll Restriction feature has no limiting effect on ARS calls placed using the ARS access code. Toll Restriction denies toll calls placed over specific trunk groups using the trunk-group access code. To restrict an ARS user from placing toll calls, ARS Toll Restriction should be assigned to the user's class of service in Field 22 of Procedure 010, Word 3.

Route Advance

The Route Advance feature has no effect on the way the ARS (Automatic Route Selection) feature selects a preference within an ARS pattern. If a trunk group in an ARS pattern is also the first trunk group in a Route Advance sequence, the ARS software ignores the alternate Route Advance trunk groups while selecting an available (and accessible) trunk group in the ARS pattern.

Tenant Services

Automatic Route Selection is a partitioned feature. In a partitioned switch, one or more extension partitions are assigned to one of 64 call in Procedure 320, Word 2.1 (Each extension partition can only be assigned to one call category). In turn, each unique routing designator/call category pair maps to one of 64 ARS patterns in Procedure 314, Word 1. The result is that an ARS call can receive treatment by a different pattern depending on the partition's assigned call category. Further, once an ARS call enters the assigned pattern, the ARS call-processing software checks successive trunk groups in the pattern to determine whether the call is allowed to use the trunk group.

The three ARS time-of-day routing plans are a system-wide resource and are not partitioned. Using an unpartitioned switch, any attendant is allowed to change the routing plan using Manual Override. For a partitioned switch, an attendant in any partition can also change the system-wide routing plan.

> **NOTE:** For a partitioned switch, it is recommended that the dial access code (Encode 60) be used to change the ARS routing plan and that only the attendant(s) in Attendant Partition 0 is informed of the dial access code. If buttons were assigned to every attendant console, then the routing plan might be changed at undesirable times.

Touch-Tone Calling Senderized Operation

When the Automatic Route Selection feature uses subnetwork trunking on non-ISDN—PRI trunk groups, each call requires a touch-tone calling sender. If a sender is not available, the switch denies the call.

Restricting Feature Use

Attendant controlled restrictions that deny access to ARS are the following:

- Controlled Outward Restriction
- Controlled Total Restriction.

Terminal restrictions that deny access to ARS are the following:

- Origination Restriction
- Outward Restriction

- Terminal-to-Terminal Only Calling.

A class of service restriction that limits access to ARS is the following:

- ARS Toll Restriction.

Certain office codes (NXXs), area codes (NPAs), and NPA-XXXs can also be restricted in Procedure 311, Words 1, 2, and 3. To restrict access to a specific office code, area code, or NPA-NXX, assign the code to the Intercept Pattern (usually Pattern 1) in the appropriate word of Procedure 311.

Hardware Requirements

The ARS feature requires the following specific hardware items.

For Traditional Modules:

- SN252 Touch-Tone Calling Sender Circuit Packs (four circuits per SN252)

  Reduce call-completion time.
- SN251 Touch-Tone Dialing Register/Receiver Circuit Packs (four circuits per SN251)

  Reduce call-completion time.

For Universal Modules:

- TN748C Tone Detector Circuit Packs (2 senders and 4 receivers per TN748C).

Regardless of the Module Type:

- TN492B Real-Time Clock Circuit.

  Allows ARS plan changes on a time-of-day basis.

Feature Administration

Assignment of the ARS feature is on a per-switch basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal), TCM (Terminal Change Management) feature, or FM (Facilities Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using Manager IV.

The following are the applicable administration procedures.

Administration Procedures — Automatic Route Selection			
Procedure	**Word**	**Purpose**	**SMT**
010	1,2,3,4	Assigns features and restrictions (including FRL and ARS toll restriction) to a class of service for a voice terminal.	Yes
012	1,2,3	Assigns distinctive names to trunk groups.	Yes
014	1 & 2	Defines a BCCOS and its default Bearer Capability requirements.	N/A
100	1	Assigns a dial access code and trunk type to a trunk group.	No
101	1	Administers the characteristics of trunks assigned to a trunk group.	No
103	1	Administers network trunk-group parameters including minimum FRL, authorization code requirements, incoming tie trunk access to AAR and ARS, and the number of trunks that are reserved for first-choice AAR and ARS preferences.	Yes
150	1	Assigns trunks (via equipment location) to a trunk group.	No
176	1	Prior to System 85, R2 V4, displays the equipment locations of trunks associated with a specific trunk-group number.	No
177	1	Prior to System 85, R2 V4, displays the equipment locations of the set of trunks (or specific trunks) associated with a specific trunk-group dial access code.	No
178	1	Beginning with system 85, R2 V4 displays the equipment locations, trunk types, and signaling types of the set of trunks (or specific trunks) associated with a specific trunk-group number or trunk-group dial access code.	No
200	1	Administers attendant console features including Direct Trunk Group Selection feature, class of service display, trunk test, and FRL for the consoles as a group.	No
202	1	Administers the Direct Trunk Group Selection buttons (attendant console) and the BUSY/WARNING level for a trunk group(s).	No
203	1	Assigns the Display Route Plan button to the attendant console(s) (manual override). The applicable encode is: 27 Display Route Plan.	No
275	3	Defines dial 1 requirements for toll calls and assigns the HNPA (local area code) and an FRL to number marked for Unauthorized Call Control.	Yes

(Continued)

Administration Procedures
Automatic Route Selection *(Continued)*

Procedure	Word	Purpose	SMT
284	1	Sets the software clock (the switch uses this clock if the real-time clock circuit [TN492] is not provided). This procedure also displays the presence or absence of the real-time clock circuit.	Yes
286	1	Administers the ARS plan in effect and the control mode (automatic, manual override, or clocked manual override).	Yes
287	1	Administers clocked manual override for the ARS plan.	Yes
309	1	Administers routing-pattern parameters for ARS including trunk-group number, FRL, warning tone, home NPA at distant end of route, dial 1 for toll requirements, and toll-table index. (To forward calls off-net, all three ARS plans must be defined using this procedure).	Yes
309	2	Administers the ARS toll tables and defines the office code(s) at the distant end of an ARS route (see Procedure 309, Word 1) as either local or toll. (To forward calls off-net, the local office codes must be specified using this procedure).	Yes
309	3	Defines the digit grouping, dialing format (touch-tone or rotary) that an ARS preference requires for subnetwork trunking. Also, prior to System 85, R2 V3, defines the inserted digits that an ARS preference requires for subnetwork trunking.	Yes
309	4	Beginning with System 85, R2 V3, defines the inserted digits that an ARS preference requires for subnetwork trunking.	Yes
309	5	Assigns the BCCOS and an NSF (Network Specific Facility) to an ARS preference.	Yes
311	1	Associates ARS pattern numbers with the local central office code(s) and service codes.	Yes
311	2	Associates an NPA (area code) with routing pattern numbers that point to 3- or 6-digit ARS route tables.	Yes
311	3	Associates a combined office code and NPA (6-digit translation) with an ARS pattern number.	Yes
312	1 or 2	Assigns the 10- to 7-Digit Conversion function. (Words 1 and 2 of Procedure 312 provide a choice of perspectives for assigning this function).	Yes
312	3	Assigns "01X Screening" for routing of selected international calls.	No

(Continued)

Administration Procedures Automatic Route Selection *(Continued)*			
Procedure	**Word**	**Purpose**	**SMT**
313	1	Administers the Unauthorized Call Control function.	Yes
314	1	For Tenant Services, assigns an ARS pattern number (1 to 64) to a call category (0 to 63) and a routing designator (1 to 64).	Yes
316	1	Administers the ARS 7-day clock for automatic plan change times.	Yes
330	1	Assigns the Pattern Queuing function. This procedure also assigns FRL Raising for ARS.	Yes
350	1	Assigns the first digit of a trunk-group dial access code, a feature dial access code, or an extension number. The first digit is defined in terms of the number of digits the switch expects to receive and call type.	No
350	2	Assigns the ARS dial access codes. The applicable encodes are as follows: 32 ARS Nontoll Route 33 ARS Toll Route 60 Network ARS Plan Switch.	No
652	3	Sets the hardware and software clock for switches that have real-time clock circuit (TN492).	No

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Automatic Route Selection	
Path Name	**Purpose**
terminal-change class-of-service attributes	Assigns restrictions (including ARS toll restriction) to an extension class of service. This screen also assigns an FRL to an extension class of service.
terminal-change system parameters (select the Access-Codes option)	Assigns an authorization code for access to ARS routing patterns and the minimum FRL numbers on the Unauthorized Call Control list.
terminal-change names trunk-group-names	Assigns distinctive names to trunk groups.

The following are the applicable FM path names used with the AP 16. A printed report of the displayed information can also be generated.

FM Screens — Automatic Route Selection	
Path Name	**Purpose**
facilities-mgmt routing route-selection home-NPA	Assigns the ARS pattern assignments for the office codes within the home-NPA (free-calling area).
facilities-mgmt synchronization	Assigns (synchronizes) the switch and Applications Processor clocks.
facilities-mgmt routing route-selection plan-change	Assigns the automatic and override schedules that determine which of the three ARS plans is currently in effect.
facilities-mgmt routing route-selection foreign-NPA	Displays and changes the ARS foreign-NPA routing for each of the three plans. When an NPA is entered, the routing information for each plan is displayed. If an NPA uses 6-digit routing, the allowed office code routing patterns are displayed.
facilities-mgmt routing route-selection office-codes	Displays and changes the pattern number-to-office-code assignments for an NPA that uses 6-digit routing.
facilities-mgmt routing route-selection toll-table	Displays and changes the office codes that are considered toll. Enter a toll table number and a list of office codes, and the toll indication for each office code is displayed.
facilities-mgmt routing route-selection patterns	Assigns attributes associated with the trunk groups that make up an ARS pattern. Enter plan, pattern, and preference (a trunk group's position in a routing pattern, that is, first-, second-, third-choice, etc). and the following information is displayed: trunk group number, minimum FRL, warning tone, send 1 for toll requirements, and toll table number.
facilities-mgmt routing route-selection rearrangement	Displays and changes the order of the trunk groups that make up an ARS pattern.
facilities-mgmt routing conversion	Displays and changes the correspondence between a private-network location code and a public-network destination code that routes to that location code (10-Digit Conversion). Either the location code or the public-network destination code (telephone number) can be entered, and the corresponding values are displayed.
facilities-mgmt routing call-control	Assigns the Unauthorized Call Control list.

Notes:

Automatic Transmission Measurement System

Description

The ATMS (Automatic Transmission Measurement System) feature allows the customer to measure transmission characteristics of private-network and public-network trunk facilities. The transmission characteristics that can be measured include loss, noise, and echo impairments. The customer can make transmission measurements on demand or set up an automatic test schedule.

Transmission measurements are made by setting up a connection between an OTL (Originating Test Line) in the customer's System 85 or Generic 2 switch (near-end switch) and a TTL (Terminating Test Line) in the far-end switch through the trunk to be tested.

Transmission measurements can be made to the following types of TTLs:

- 105-type TTL with return loss
- 105-type TTL without return loss (i.e., 56A mini-responder)
- High level tone source (i.e., LC145)
- Low level tone source (i.e., SN260B)
- 102-type TTL
- 100-type TTL.

The following is a list of all possible transmission measurements that a System 85 or Generic 2 OTL can make. The measurements that can be made by the OTL on a specific test call depend on the type of TTL that terminates the connection and may be a subset of this list.

- Two-way loss at 404, 1004, and 2804 Hz at a level of -16 dBm
- Two-way loss at 1004 Hz at a level of 0 dBm
- Two-way C-message weighted noise
- Two-way C-notched noise
- Two-way singing return loss — low frequency
- Two-way singing return loss — high frequency
- Two-way echo return loss.

Feature History and Development

The ATMS feature was first available on Release 2, Version 2 of System 85.

With Generic 2.1, Issue 2.0, the MTCP (Maintenance Test Circuit Pack), TN771B was introduced. This circuit pack allows the use of the ATMS feature on switches with all Universal Modules. Previously, there had to be at least one traditional module for the ADFTC (Analog Digital Facility Test Circuit), SN261.

User Operations

The primary customer interface to the ATMS feature is through the Facilities Management application of the AP (Applications Processor) 16 for System 85, Release 2, Version 3 and earlier switches. Beginning with Release 2, Version 4 of System 85, the primary customer interface to the ATMS feature is through the Manager IV. Information on user operations for these applications can be found in the following guides:

Automatic Transmission Measurement System Administrators Guide for the AP 16	585-201-704
Centralized System Management Terminal Change Management User's Guide	585-220-701
Centralized System Management Facilities Management User's Guide	585-220-702

Interactions With Other Features

The following System 85 and Generic 2 feature affects or is affected by the operation of this feature.

Trunk Verification—Voice Terminal

The ATMS feature uses the TVVT (Trunk Verification—Voice Terminal) feature to set up trunk test calls. Refer to the TVVT feature description in this manual for information about interactions between the TVVT feature and other switch features.

Hardware Requirements

Specific hardware requirements for the ATMS feature depend on the Module type being used. One of the following is required:

With Traditional Modules:

- SN261 ADFTC (Analog/Digital Facility Test Circuit)

  The ADFTC is used with the Traditional Module. The ADFTC can function as either an OTL, or a TTL, or both. It cannot function simultaneously as both OTL and a TTL.

With Universal Modules:

- TN771B MTCP (Maintenance Test Circuit Pack)

  The MTCP is available beginning with Generic 2.1, Issue 2.0 and is used with the Universal Module. When used, the MTCP replaces the ADFTC and performs a superset *(see* the System Description, 555-105-201, for details) of the same functions. The MTCP can also be used with earlier Generic 2 switches provided that the bus interface is also upgraded to the UN154B Universal Bus Interface.

Feature Administration

The ATMS feature is assigned on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the FM (Facilities Management) feature (R2V3 and earlier).

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

The customer can also administer this feature using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
AUTOMATIC TRANSMISSION MEASUREMENT SYSTEM

PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns an extension number, equipment location, and class of service to an ADFTC circuit.	Yes
010	1	Defines the class of service for the ADFTC circuit as rotary dial (Field 15	
010	3	Assigns Data Protection-Permanent to a class of service.	
051	1	Associates a terminal type with an equipment location. R2 V1 to R2 V4: Field 6 = 8 Generic 2: Field 6 = 5 for an ADFTC, 8 for an MTCP analog, or 9 for an MTCP digital.	Yes
052	1	Associates a device type with an equipment location. Field 6 = 4 (for either an ADFTC or a MTCP) Field 7, range = 0-1 for an ADFTC or 0 for a MTCP.	Yes

ADMINISTRATION PROCEDURES AUTOMATIC TRANSMISSION MEASUREMENT SYSTEM (Contd)			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
070	1 to 5	Displays terminal information. (Word 5 is not used before Generic 2).	Yes
106	1	Displays information about trunks that are maintenance busy.	Yes
107	1	Assigns a test line type and TTL telephone number to a trunk group.	No
107	2	Assigns marginal thresholds for ATMS tests.	No
107	3	Assigns automatic testing parameters to a test schedule including: Schedule (once or number of week between tests) Starting hour Duration of test (hours) The days of the week tests will be made. Also displays the number of weeks since last test.	No
107	4	Assigns a trunk group to a test schedule.	No
107	5	Displays the trunk groups assigned to a test schedule.	No
107	6	Defines busy out thresholds (maximum and minimum loss and deviation) for automatic busy out and the maximum percentage of trunks that may be busied out automatically for a trunk group.	No
107	7	Defines the number of trunks (per trunk group) that are allowed to be maintenance busied (that is, failed the unacceptable transmission thresholds twice) before a minor alarm is raised.	No
290	1	Displays circuit status of assigned circuit packs.	Yes
350	2	Assigns the dial access codes for the Trunk Verification —Voice Terminal feature. The applicable encodes are as follows: 42 Maintenance busy a trunk 43 Maintenance release a trunk 44 Trunk Verification — Voice Terminal.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS AUTOMATIC TRANSMISSION MEASUREMENT SYSTEM	
PATH NAME	**PURPOSE**
terminal-change connection	Establishes or releases a connection from the AP to a switch in the private network.
terminal-change terminal equipment	Assigns an extension number to an ADFTC circuit.
terminal-change extension attributes	Assigns a class-of-service to an extension number.
terminal-change class-of-service attributes	Defines a class of service for the ADFTC circuit as rotary dial (TOUCH TONE = n) and assigns the Data Protection-Permanent feature.
terminal-change terminal extensions	Requests an on-line listing or a hard copy report of the dial plan, assigned extensions, and unassigned extensions.
terminal-change terminal unassigned-equipment	Requests an on-line listing or hard copy report of unassigned equipment locations.

The following are the applicable FM path names used with the AP 16.

FM SCREENS AUTOMATIC TRANSMISSION MEASUREMENT SYSTEM	
PATH NAME	**PURPOSE**
facilities-mgmt connection	Establishes or releases a connection from the AP to a switch in the private network.
facilities-mgmt transmission thresholds	Assigns the test line type and TTL telephone number and defines marginal and unacceptable transmission measurement thresholds for a trunk group.
facilities-mgmt transmission schedule	Assigns automatic testing parameters to a test schedule, including the days of the week tests will be made, starting hour, duration, and weeks between tests. Also assigns trunk groups to a test schedule.
facilities-mgmt transmission exec-summary	Requests an executive summary report. The report can be sorted by test schedule number, trunk group number, or test date.
facilities-mgmt transmission summary	Requests a summary report of trunk failures.
facilities-mgmt transmission details	Displays detailed transmission data for trunk transmission failures on a trunk-by-trunk basis. After viewing, entries can be saved or deleted.
facilities-mgmt transmission details	Displays reasons for trunk failures. After viewing, miscellaneous entries can be saved or deleted.
facilities-mgmt transmission demand supervision	Initiates a supervision test on a particular trunk.
facilities-mgmt transmission demand	Initiates all possible transmission tests (determined transmission by type of the TTL) on a particular trunk. After a trunk has been tested, the status of the trunk can be obtained and the trunk can be either made maintenance-busy or released from a maintenance-busy condition.

Bearer Capability

Description

The Bearer Capability feature allows the switch to match the calling requirements of a specific call with the best available resources to support that call. Bearer Capability has evolved from, and supports, the ***interworking*** function of ISDN. The Bearer Capability feature uses information that will normally be available in an ISDN environment (and provides equivalent information for non-ISDN situations) to effectively apply available resources for the best support arrangements to meet the service needs of each call.

Elements

The Bearer Capability feature uses two fundamental elements: the ISDN (Integrated Services Digital Network) Bearer Capability IE (Information Element) and the BCCOS (Bearer Capability Class of Service).

Application

Bearer Capability applies to all calls but ***is of primary significance in data calling situations.*** The bearer capability concept was used for the System 85 with the Release 2, Version 4 ISDN—PRI feature. With the DEFINITY Generic 2 switch, the scope and function of Bearer Capability is expanded and the BCCOS is added. Call completion, at the best available level of functionality (particularly for data calls), will increase.

A BCCOS is assigned to each extension (line), trunk group, and routing preference (AAR [Automatic Alternate Routing], ARS [Alternate Route Selection] or WCR [World Class Routing]). The BCCOS provides the following information and services:

- Identifies the types of calls and call characteristice that the assigned facility can support.
- Provides call processing instructions on how to handle incoming calls with a specific BCCOS.
- Contains call requirements and characteristics for non-ISDN calls and non-data module calls originating on the assigned facility. These assigned call requirements and characteristics are used as a last resort for all calls that do not contain needed bearer capability information in the call setup message.

Feature History and Development

Although the bearer capability concept was used, on a limited scale, in System 85 Release 2, Version 4, the Bearer Capability feature is first introduced in Generic 2.

System 85, Release 2, Version 4

In System 85, Release 2, Version 4, bearer capability is a function of the ISDN—PRI feature. In this version, five bearer capability codes are available: one for analog calls (including voice grade data) and four for data calls (mode 0 through mode 3). Also, three levels of trunking service can be specified:

- ISDN Required
- ISDN Preferred
- Any Facility.

In System 85, R2 V4, bearer capability is used only as a route selection criterion for the AAR (Automatic Alternate Routing) and ARS (Alternate Route Selection) features.

Generic 2

In Generic 2, the functionality available in System 85, R2 V4 continues. That is, the ISDN trunking services specification is still made (ISDN required, ISDN preferred, any facility). Bearer capability designation, however, is expanded to provide for 256 possible BCCOSs, of which 9 are predefined. All BCCOSs are user administrable and can be assigned or changed as local requirements dictate.

The application of Bearer Capability is also expanded in DEFINITY Generic 2. The network routing functions still apply, but on a much more sophisticated level. BCCOS applies to all extensions, trunk groups, and network routing preferences. Bearer Capability also plays a significant role in other features such as Modem Pooling and Host Computer Access.

General Concept

The Bearer Capability feature is largely based on ISDN calling reuirements; however, because of its close relationship with ***interworking,*** this feature extends to non-ISDN calls as well. The Bearer Capability feature uses two closely related concepts: ***Bearer Capability*** and ***BCCOS.***

- ***Bearer Capability***

  The term Bearer Capability is derived from the ISDN specifications and standards. It is actually the name of an IE in the ISDN call setup message (codeset 0, IE BC). Contrary to the generic meaning of the name, this is the IE that identifies support ***requirements*** of the call. That is, the bearer capability applies to the call and identifies the types of resources needed to support that call. To avoid confusion we will generally refer to this concept as the ***BCR (Bearer Capability Requirement)*** except when talking about the actual IE by name.

- ***BCCOS (Bearer Capability Class of Service)***

  The BCCOS is an administered value (Procedure 014) that is assigned to extensions, trunk groups, and routing preferences. The BCCOS specifically identifies the types of calls and the data call characteristics that a facility (line, trunk, conversion resource, etc.) can support.

Bearer Capability Requirement

All calls have a ***bearer capability requirement.*** That is, every call has a certain set or range of requirements needed to support that call.

Voice Call Requirements

For voice calls, the range of requirements is usually broad and very general. Most facilities common to telecommunications networks and switches are capable of providing adequate support to voice calling needs. Essentially, voice calls require a talking path and a call control protocol that provides adequate switching and supervision to properly connect and disconnect the call. Other services [such as, ACD (Automatic Call Distribution), Conferencing, Call Coverage, Call Forwarding, etc.] may also be included in specific call setup, but these extend beyond the minimum essential calling requirements.

Data Call Requirements

For data calls, the range of requirements is generally narrower and more specific than with voice calls. Data call requirements include such factors as:

- ***Data Rate***

  Data calls operate at a specific data rate. For a given call, the data rate may be specific (only one rate is acceptable) while for other calls a range of data rates, for example, from 300 bps to 19.2 Kbps, may be acceptable.

- ***Synchronization***

  Data calls operate in either of two synchronization modes: synchronous or asynchronous. Some facilities are capable of operating in either mode while others operate only in one mode or the other.

- ***Channel Type***

  Data communications channels are for either ***restricted*** or ***clear*** data. For a specific call, a clear channel can support either restricted or unrestricted data calls while a restricted channel can support only restricted data traffic.

Sources of Bearer Capability Requirements Information

Bearer capability requirements are used to determine routing requirements, both external (over trunks) and internal (such as, Modem Pooling). In previous versions, information about call requirements comes exclusively from switch translation. In Generic 2, identification of both call type (voice or data) and resource requirements is obtained from the best available sources as follows:

- ***Call Setup Messages***

  For ISDN originated calls, call control information contained in the ISDN call setup message associated with each specific call is the primary source for bearer capability requirements. These call control setup messages contain IEs (Information Elements) that indicate the type of call (such as, voice, Mode 0 data, etc.), protocol used, data rate, and other information needed to determine resource requirements.

- ***Data Module Query***

  For digital data calls, all AT&T manufactured data modules (both BRI and DCP) have the ability to respond to switch requests for additional information. This query response capability may or may not be present in non-AT&T data modules. When needed information is not present in a specific call setup message, the switch can request additional information from data modules (for example, synchronous or asynchronous).

- ***Default Values***

  As a last resort for all calls, the customer administered BCCOS can be used as a source for bearer capability requirements information. One element of the BCCOS (Procedure 014, Word 1, Field 16) defines the default ***Bearer Capability Requirement*** assigned to each BCCOS. This default BCR is used for outgoing calls when the other sources for this information are not available.

  A BCCOS is assigned to call origination and termination points, and to carrier and support facilities. This administrable BCCOS provides default requirements and characteristics for specific facilities. The default BCCOSs are associated with the facility (lines, trunks, etc.) and not with a specific call.

If call processing must, for any reason, use only the default BCCOS (unchanged) values, the resulting DEFINITY Generic 2 call handling will be consistent with call handling provided by a System 85 R2 V4 switch for the same call. In effect, the default processing for Generic 2 equates to the basic bearer capability call handling provided by the System 85 R2 V4 switch.

Default Values for BCCOSs (Bearer Capability Classes of Service)

The Generic 2 switch provides for up to 256 BCCOSs (0 through 255), of which nine are provided initially as defaults. Table 23-A lists the nine initial default BCCOSs and the closest corresponding System 85, Release 2, Version 4 bearer capability code (if any).

TABLE 23-A. Default Bearer Capability Classes of Service

Default BCCOS DEFINITY G2	BCC System 85 R2V4	Type of Call Supported
0	(0)	Voice only*
1	2	Mode 2 Data
2	—	BRI Voice / Data
3		Unknown Digital
4	(0)	Unknown Analog
5	(0)	Voice Grade Data*
6	4	Mode 0 Data
7	1	Mode 1 Data
8	3	Mode 3 Data

* Bearer services available on System 85, Release 2, Version 4 but in a single BCC.

All origination, termination, and call processing support facilities on the DEFINITY Generic 2 switch come from the factory with one of these default BCCOSs assigned. Note that BCCOS applies to non-ISDN facilities (such as, analog and DCP lines and trunks) as well as to ISDN facilities.

Administrators can create new BCCOS using Procedure 014, or they can change the values assigned to the default BCCOS.

CAUTION: *Changing the values of the default BCCOSs is* **not recommended.** *This usually has unanticipated side effects that may be difficult to detect and correct.*

Not only are there more bearer capabilities available in DEFINITY Generic 2 than in System 85, R2 V4, but the new BCCOSS provide a wider range of information. The following series of tables shows the variety of information provided by the BCCOS. It also shows the initial values assigned and the intended application for each of the default BCCOSs.

TABLE 23-B. Default Values BCCOS 0 — Voice Only

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	0	Circuit Switched
Mode 1	0	Circuit Switched
Mode 2	0	Circuit Switched
Mode 3	0	Circuit switched
Voice	0	Circuit Switched
Voice Grade Data	0	Circuit Switched
Unknown Digital	0	Circuit Switched
Unknown Analog	0	Circuit Switched
Mode 3/2	0	Circuit Switched
X.25	0	Circuit Switched
Default Bearer Capability	0	Voice
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	0	Not Supported
56 K	0	Not Supported
19.2 K	0	Not Supported
9.6 K	0	Not Supported
4.8 K	0	Not Supported
2.4 K	0	Not Supported
1.2 K	0	Not Supported
300	0	Not Supported
low	0	Not Supported

Bearer capability Class of Service 0 is intended for analog lines and voice-only extensions. It is the default for voice terminal extension numbers. All extension numbers are assigned BCCOS 0 when they are initially assigned to a Terminal Equipment Location in Procedure 000, Word 1.

TABLE 23-C. Default Values BCCOS 1 — Mode 2 Data

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	0	Circuit Switched
Mode 1	0	Circuit Switched
Mode 2	0	Circuit Switched
Mode 3	0	Circuit Switched
Voice	0	Circuit Switched
Voice Grade Data	1	Mode 2/Analog Modem Pool
Unknown Digital	0	Circuit Switched
Unknown Analog	1	Mode 2/Analog Modem Pool
Mode 3/2	0	Circuit Switched
X.25	0	Circuit Switched
Default Bearer Capability	2	Mode 2
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	1	Supported
56 K	1	Supported
19.2 K	1	Supported
9.6 K	1	Supported
4.8 K	1	Supported
2.4 K	1	Supported
1.2 K	1	Supported
300	1	Supported
low	1	Supported
Default Data Rate	3	1200 Baud
Synchronization	0	Asynchronous
Duplex	0	Full Duplex

Bearer Capability Class of Service 1 is the default for DCP data modules administered as lines and trunks. Extensions administered for data modules in Procedure 052 default to using this class of service. HCA (Host Computer Access) trunks (trunk types 102 — 109) also default to BCCOS 1. This BCCOS should also be used for BRI data modules that have separate extension numbers.

The administrator can modify BCCOS values. For example, if the user wants to block Mode 3 calls, the administrator can create a new BCCOS to do so using Procedure 014 Word 1. This can be done simply by copying BCCOS 1 (for use as a template), changing the BCCOS number and changing Field 9 to a 2 (block mode 3 data calls).

TABLE 23-D. Default Values BCCOS 2 — BRI Voice / Data

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	0	Circuit Switched
Mode 1	0	Circuit Switched
Mode 2	0	Circuit Switched
Mode 3	0	Circuit Switched
Voice	0	Circuit Switched
Voice Grade Data	1	Mode 2 / Analog Modem Pool
Unknown Digital	0	Circuit Switched
Unknown Analog	0	Circuit Switched
Mode 3/2	0	Circuit Switched
X.25	0	Circuit switched
Default Bearer Capability	0	Voice
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	1	Supported
56 K	1	Supported
19.2 K	1	Supported
9.6 K	1	Supported
4.8 K	1	Supported
2.4 K	1	Supported
1.2 K	1	Supported
300	1	Supported
low	1	Supported
Default Data Rate	3	1200 Baud
Synchronization	0	Asynchronous
Duplex	0	Full Duplex

Bearer Capability Class of Service 2 is not the default for any facility. It is intended to serve as a template for single extension BRI voice / data stations.

TABLE 23-E. Default Values BCCOS 3 — Unknown Digital

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	0	Circuit Switched
Mode 1	0	Circuit Switched
Mode 2	0	Circuit Switched
Mode 3	0	Circuit Switched
Voice	0	Circuit Switched
Voice Grade Data	0	Circuit Switched
Unknown Digital	0	Circuit Switched
Unknown Analog	0	Circuit Switched
Mode 3/2	0	Circuit Switched
X.25	0	Circuit Switched
Default Bearer Capability	6	Unknown Digital
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	1	Supported
56 K	1	Supported
19.2 K	1	Supported
9.6 K	1	Supported
4.8 K	1	Supported
2.4 K	1	Supported
1.2 K	1	Supported
300	1	Supported
low	1	Supported
Default Data Rate	3	1200 Baud
Synchronization	0	Asynchronous
Duplex	0	Full Duplex

Bearer Capability Class of Service 3 is defined, but is not the default for any extension, trunk, or routing preference. Thk BCCOS corresponds to the AVD flag on DS1 trunks in System 85. This BCCOS allows TRACS a simple mechanism for upgrading a System 85, R2 V4 switch to DEFINITY Generic 2.

For call processing purposes, a BCCOS 3 call (incoming) could be any one of three call types: digital data, digitized voice, or voice grade data. Neither the switch nor the BRI has any way to determine which type of service is required. The switch offers these calls to the end point BRI or DCP. If the station has a separate extension for voice and data, the call is offered to the station and routed to the addressed extension. If the called extension is an analog data endpoint, the call is handled as voice grade data.

TABLE 23-F. Default Values BCCOS 4 — Unknown Analog

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	1	Mode 2/Analog Modem Pool
Mode 1	1	Mode 2/Analog Modem Pool
Mode 2	1	Mode 2/Analog Modem Pool
Mode 3	1	Mode 2/Analog Modem Pool
Voice	0	Circuit Switched
Voice Grade Data	0	Circuit Switched
Unknown Digital	0	Circuit Switched
Unknown Analog	0	Circuit Switched
Mode 3/2	1	Mode 2/Analog Modem Pool
X.25	0	Circuit Switched
Default Bearer Capability	7	Unknown Analog
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	0	Not Supported
56 K	0	Not Supported
19.2 K	0	Not Supported
9.6 K	0	Not Supported
4.8 K	0	Not Supported
2.4 K	0	Not Supported
1.2 K	0	Not Supported
300	0	Not Supported
low	0	Not Supported
Default Data Rate	0	None
Synchronization	0	Asynchronous
Duplex	0	Full Duplex

Bearer Capability Class of Service 4 is the default for analog trunks and network routing preferences.

TABLE 23-G. Default Values BCCOS 5 — Voice Grade Data

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	1	Mode 2/Analog Modem Pool
Mode 1	1	Mode 2/Analog Modem Pool
Mode 2	1	Mode 2/Analog Modem Pool
Mode 3	1	Mode 2/Analog Modem Pool
Voice	0	Circuit Switched
Voice Grade Data	0	Circuit Switched
Unknown Digital	0	Circuit Switched
Unknown Analog	0	Circuit Switched
Mode 3/2	1	Mode 2/Analog Modem Pool
X.25	0	Circuit Switched
Default Bearer Capability	5	Voice Grade Data
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	0	Not Supported
56 K	0	Not Supported
19.2 K	0	Not Supported
9.6 K	0	Not Supported
4.8 K	0	Not Supported
2.4 K	1	Supported
1.2 K	1	Supported
300	1	Supported
low	0	Not Supported
Default Data Rate	3	1200 Baud
Synchronization	0	Asynchronous
Duplex	0	Full Duplex

Bearer Capability Class of Service 5 is the default for modems. The modem side of a Modem Pooling arrangement (trunk type 101) defaults to BCCOS 5.

NOTE: When a Modem Pooling Conversion Resource is inserted in a call path, the bearer capability requirements for that call change (a modem cannot be used in an unbalanced configuration). For example, with an outgoing call the BCCOS of the conversion resource is used (such as, voice grade data, 2.4 Khz, etc.).

TABLE 23-H. Default Values BCCOS 6 — Mode 0 Data

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	0	Circuit Switched
Mode 1	0	Circuit Switched
Mode 2	0	Circuit Switched
Mode 3	0	Circuit Switched
Voice	0	Circuit Switched
Voice Grade Data	1	Mode 2/Analog Modem Pool
Unknown Digital	0	Circuit Switched
Unknown Analog	1	Mode 2/Analog Modem Pool
Mode 3/2	0	Circuit Switched
X.25	0	Circuit Switched
Default Bearer Capability	4	Mode 0
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	1	Supported
56 K	0	Not Supported
19.2 K	0	Not Supported
9.6 K	0	Not Supported
4.8 K	0	Not Supported
2.4 K	0	Not Supported
1.2 K	0	Not Supported
300	0	Not Supported
low	0	Not Supported
Default Data Rate	9	64K Baud
Synchronization	1	Synchronous
Duplex	0	Full Duplex

Bearer Capability Class of Service 6 is the default for Mode 0 data modules. No line or trunk automatically defaults to this BCCOS.

TABLE 23-I. Default Values BCCOS 7 — Mode 1 Data

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	0	Circuit Switched
Mode 1	0	Circuit Switched
Mode 2	0	Circuit Switched
Mode 3	0	Circuit Switched
Voice	0	Circuit Switched
Voice Grade Data	1	Mode Z / Analog Modem Pool
Unknown Digital	0	Circuit Switched
Unknown Analog	1	Mode 2 / Analog Modem Pool
Mode 3/2	0	Circuit Switched
X.25	0	Circuit Switched
Default Bearer Capability	1	Mode 1
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	0	Not Supported
56 K	1	Supported
19.2 K	0	Not Supported
9.6 K	0	Not Supported
4.8 K	0	Not Supported
2.4 K	0	Not Supported
1.2 K	0	Not Supported
300	0	Not Supported
low	0	Not Supported
Default Data Rate	8	56K Baud
Synchronization	1	Synchronous
Duplex	0	Full Duplex

Bearer Capability Class of Service 7 is the default for Mode 1 data modules. No line or trunk automatically defaults to this BCCOS.

TABLE 23-J. Default Values BCCOS 8 — Mode 3 Data

Attribute	Default Value	
Transport Mode	0	Circuit
Channel Type	0	Restricted
Mode 0	0	Circuit Switched
Mode 1	0	Circuit Switched
Mode 2	0	Circuit Switched
Mode 3	0	Circuit Switched
Voice	0	Circuit Switched
Voice Grade Data	1	Mode 2/Analog Modem Pool
Unknown Digital	0	Circuit Switched
Unknown Analog	1	Mode 2/Analog Modem Pool
Mode 3/2	0	Circuit Switchd
X.25	0	Circuit Switched
Default Bearer Capability	3	Mode 3
Default Transport Mode	0	Circuit
Default Channel Type	0	Restricted
Data Rates:		
64K	1	Supported
56 K	1	Supported
19.2 K	1	Supported
9.6 K	1	Supported
4.8 K	1	Supported
2.4 K	1	Supported
1.2 K	1	Supported
300	1	Supported
low	1	Supported
Default Data Rate	3	1200 Baud
Synchronization	o	Asynchronous
Duplex	o	Full Duplex

Bearer Capability Class of Service 8 is the default for Mode 3 data modules. No line or trunk automatically defaults to this BCCOS.

Call Processing With Bearer Capability

Bearer Capability affects call processing for both incoming and outgoing calls. The ways in which the switch obtains bearer capability information and uses that information to handle specific calls vary, depending on a number of factors. The logic used is actually not complicated. However, there are so many different possibilities to be explained that it at first appears mind boggling. Figures 23-1 through 23-3 are flowcharts showing the basic logic that applies for different call processing situations.

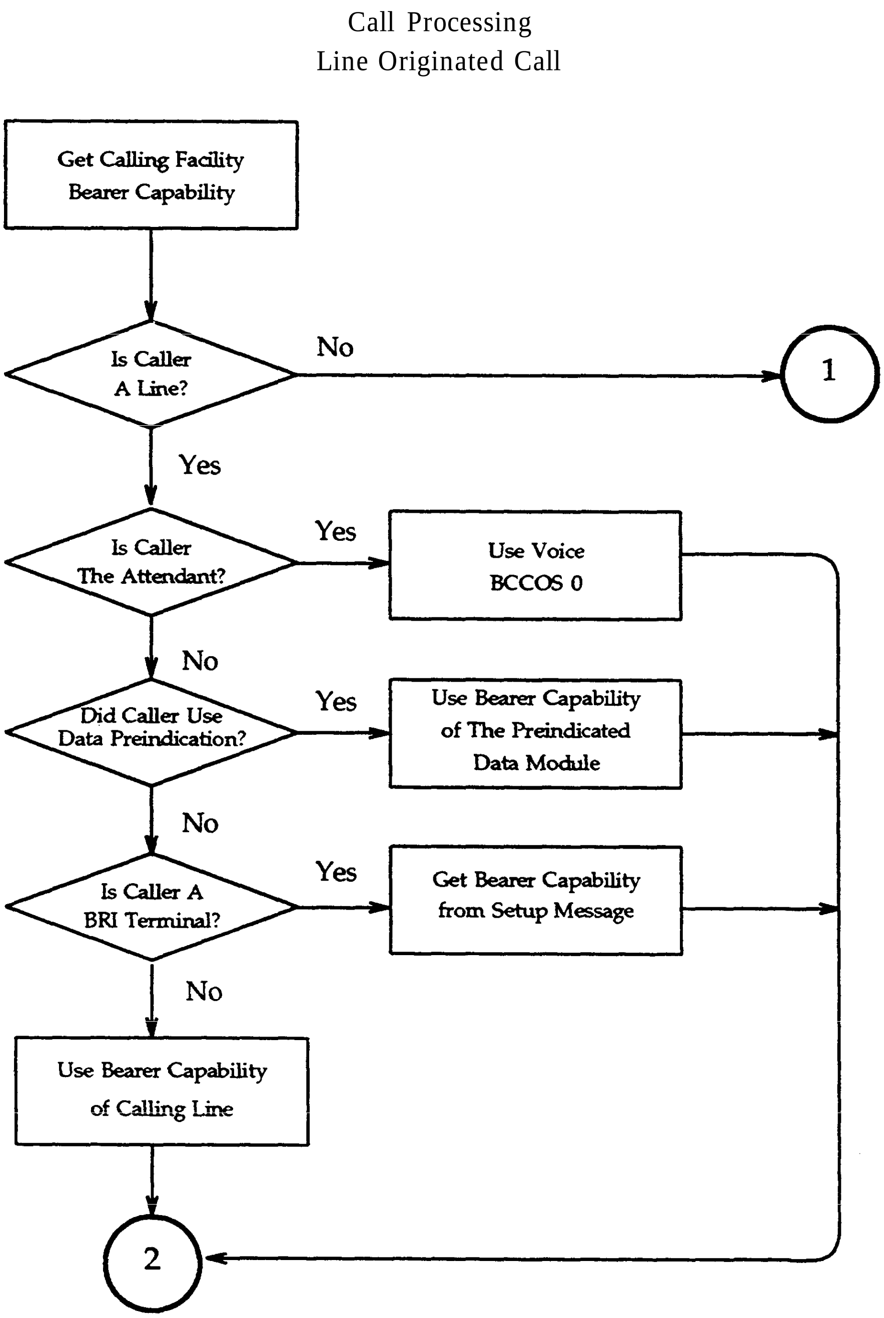


Figure 23-1. Call Processing With Bearer Capability - Incoming Line Call

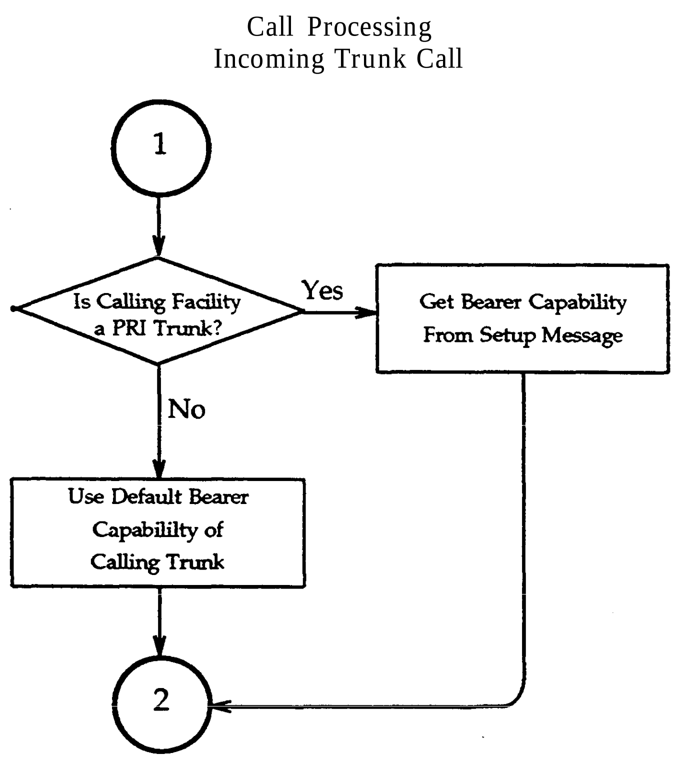


Figure 23-2. Call Processing With Bearer Capability - Incoming Trunk Call

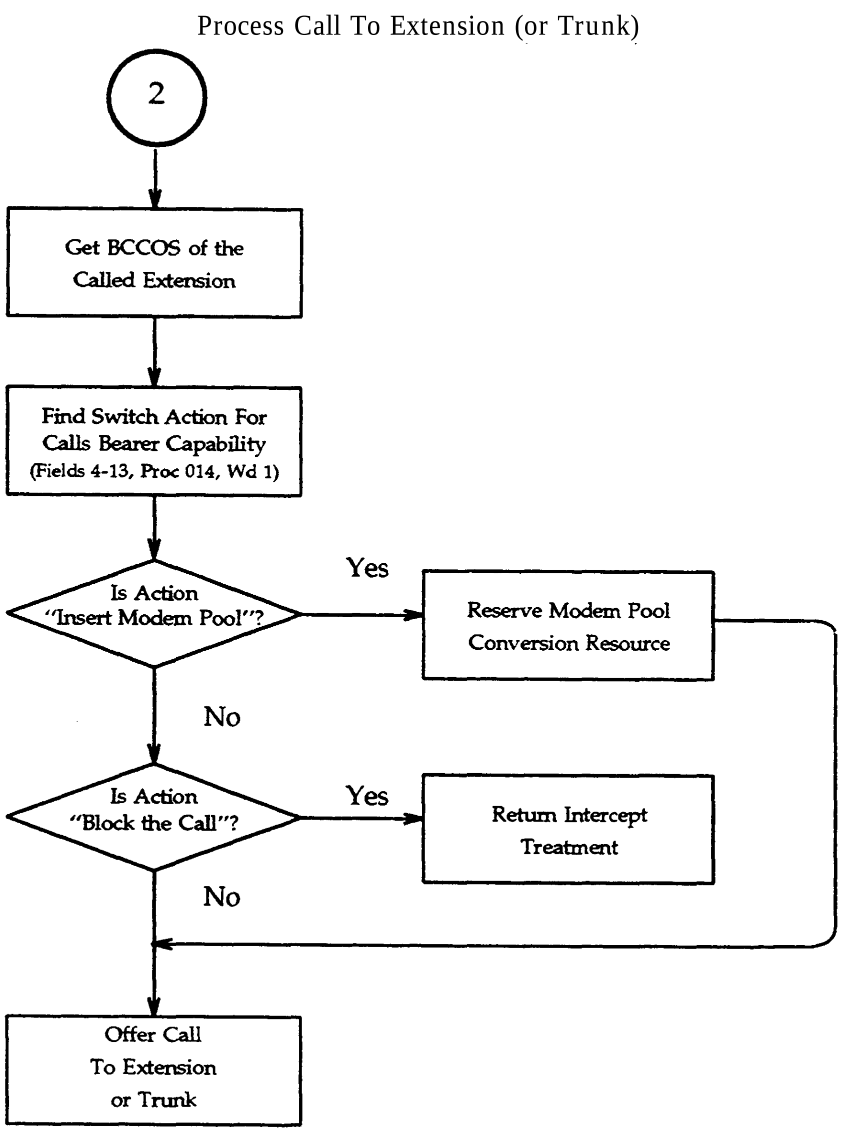


Figure 23-3. Call Processing With Bearer Capability - Offer to Extension or Trunk

The process logic shown in Figure 23-3 applies to two situations. These are the ***incoming call to an extension*** scenario and the ***outgoing call to a trunk*** scenario. The difference ***between these two cases*** is that for outgoing calls to a trunk, a networking feature (either AAR, ARS, or WCR) is involved in route selection and call processing. The additional flow logic introduced by these features is shown in flowcharts provided in each separate feature chapter in this manual.

Incoming Call Processing Based on BCCOS

Based on the initial default BCCOS, Table 23-K shows the call processing actions for incoming calls to ISDN—BRI stations on a Generic 2 switch. All three ISDN—BRI station configurations are shown in these examples.

TABLE 23-K. Incoming Call Processing Based on BCCOS

Incoming Call BCCOS	BRI Terminal Characteristics		
	Voice/Data Station (BCCOS 0 and 1)	Single Extension Voice Only (BCCOS 0)	Single Extension Data Only (BCCOS 1)
0 (Analog Voice)	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call Rejected at BRI*
1 (Mode 2 Data)	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call Rejected at BRI*	Circuit Switch Call w/ ISDN Signaling
2 BRI Voice/Data)	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call Rejected at BRI*	Circuit Switch Call w/ ISDN Signaling
3 (Unknown Digital)	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call Rejected at BRI*	Circuit Switch Call w/ ISDN Signaling
4 (Unknown Analog)	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call w/ ISDN Signaling	Insert Modem Pool
5 (Voice Grade Data)	Insert Modem Pool	Circuit Switch Call Rejected at BRI*	Insert Modem Pool
6 (Mode 0 Data)	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call Rejected at BRI*	Circuit Switch Call w/ ISDN Signaling
7 (Mode 1 Data)	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call Rejected at BRI*	Circuit Switch Call w/ ISDN Signaling
8 (Mode 3 Data)	Circuit Switch Call w/ ISDN Signaling	Circuit Switch Call Rejected at BRI*	Circuit Switch Call w/ ISDN Signaling

* Calls rejected at the BRI result in ringback tone at the calling terminal but no alerting at the called BRI station.

The following descriptions may help to clarify some of the call cases shown in Table 23-K.

- Incoming Call With BCCOS 0

  This BCCOS indicates the incoming call is an analog voice call.

  — ***BRI Voice/ Data Station With Both Voice and Data Terminals***

  For the BRI voice/data station with both a voice and a data extension, the call is circuit switched (offered directly to the station). With ISDN signaling, only the voice terminal will accept this call regardless of which extension (voice or data) was dialed.

  — ***Single Extension, Voice Terminal Only***

  For the single extension BRI station with only a voice terminal, the call is circuit switched (offered directly to the extension).

  — ***Single Extension, Data Terminal Only***

  For the single extension BRI station with only a data terminal, the call is offered to the station but the BRI rejects it because there is no terminal that can handle a voice call. After the interface rejects the call, the call is given ***intercept treatment*** by the switch.

- Incoming Call With BCCOS 1

This BCCOS indicates a Mode 2 data call.

— ***BRI Voice/ Data Station With Both Voice and Data Terminals***

For the BRI voice / data station with both a voice and a data extension, the call is circuit switched (offered directly to the station). With ISDN signaling, only the data terminal will accept this call, regardless of which extension (voice or data) was dialed.

— ***Single Extension, Voice Terminal Only***

For the single extension BRI station with only a voice terminal, the call is offered to the station, but the interface rejects the call because it doesn't have a terminal that can handle data calls. After the interface rejects the call, the switch returns ***intercept treatment*** for the call.

— ***Single Extension, Data Terminal Only***

For the single extension BRI station with only a data terminal, the call is circuit switched (offered directly to the extension).

- **Incoming Call With BCCOS 5**

This BCCOS indicates a voice grade data call. The signal is in a modulated (analog) form.

— ***BRI Voice/ Data Station With Both Voice and Data Terminals***

For the BRI voice / data station with both a voice and a data extension, the call is offered to the station with a Modem Pooling conversion resource inserted. Only the data terminal will accept the call, and Modem Pooling support is required for analog-to-digital format conversion.

— ***Single Extension, Voice Terminal Only***

For the single extension BRI station with only a voice terminal, the call is offered to the BRI which rejects the call because there is no terminal capable of handling a data call. After the interface rejects the call, the switch returns ***intercept treatment*** for the call.

— ***Single Extension, Data Terminal Only***

For the single extension BRI station with only a data terminal, the call is offered to the station with a Modem Pooling conversion resource inserted. Modem Pooling support is required for analog-to-digital format conversion.

Outgoing Call Processing Based on BCCOS

Outgoing call processing is no more involved than incoming call processing. However, for outgoing calls, one of the networking features (AAR, ARS, or WCR) is generally involved.

These features select call routes based on a number of factors such as FRL (Facilities Restriction Level), time-of-day considerations, and calling station permissions / restrictions (toll, nontolling partitioning, etc.). Detailed descriptions of the networking features are provided separately.

In Generic 2, Bearer Capability is a factor in routing "preference" selection. All three networking features use "patterns", which contain preferences (trunk groups) that support specific routing requirements. Each of these preferences is assigned a BCCOS. For outgoing calls, the BCCOS of these preferences is considered in the route selection process, and under some circumstances may be considered twice. The following example illustrates how the routing pattern / preference searches are affected by the Bearer Capability feature.

Routing Pattern Search With Bearer Capability

Table 23-L shows Pattern 29, a hypothetical routing pattern containing five preferences.

TABLE 23-L. Routing Preference With BCCOS Actions

Routing Pattern 29									
	Data Modes 0				Voice 1	Voice Grade 2	Unknown 3	Unknown Only	Channel Data
Preference 1 ***(BCCOS 4)***	Modem Pool	Modem Pool	Modem Pool	Modem Pool	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Restricted
Preference 2 ***(BCCOS 21)***	Block	Block	Block	Block	Circuit Switch	Block	Circuit Switch	Block	Restricted
Preference 3 ***(BCCOS 3)***	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Restricted
Preference 4 ***BCCOS 24)***	Block	Block	Modem Pool	Block	Circuit Switch	Circuit Switch	Circuit Switch	Modem Pool	Restricted
Preference 5 ***BCCOS 31)***	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Circuit Switch	Clear

Some predefined BCCOSs are used (Preferences 1 and 3) while some new BCCOSs have been defined (Preferences 2, 4, and 5).

For our example, a Mode 2 restricted data call is being routed over Pattern 29. Network routing software makes all the checks that it did in earlier systems (FRL, etc.) and also checks the switch actions indicated for each BCCOS and the type of call being routed, in this example, the Mode 2 data column. These switch actions for each BCCOS are specified in translation using Procedure 014, Word 1. Fields 4 through 13.

For a data call, a connection that does not require Modem Pooling is generally considered preferable to one that does. However, a connection that uses Modem Pooling is usually preferable to denying the call. The pattern search algorithms take both these factors into consideration. As each pattern preference is checked, "circuit switching" the call is the action that is sought. However, if a preference indicates "insert Modem Pool", and there is an available trunk and appropriate channel type, this option is recorded for future reference, if needed. In this example:

- Preference 1 shows ***insert Modem Pool*** as the action for Mode 2 data calls and the channel type (Restricted) is acceptable.

  If a trunk is available, this fact is recorded for future reference if needed.

- Preference 2 shows ***block the call*** as the action to take and is eliminated from consideration.

- Preference 3 shows ***circuit switch the call*** as the action to take for Mode 2 data calls. If a trunk is available, the search stops here and that trunk is selected. If no trunk is available in preference 3, the search continues.

- Preference 4 shows ***insert Modem Pool*** as the action for Mode 2 data calls. This is the same as for preference 1.

  Again, if a trunk is available, this fact is recorded for future reference if needed.

- Preference 5 shows circuit switch the call for Mode 2 data, as did preference 3.

  In this case, the channel type is ***clear.*** A clear channel can support a restricted data call so this preference is acceptable. If the situation had been reversed (if the call had been clear channel and the preference had been restricted), the match would not have been satisfactory and this preference would have been skipped.

  Again, (as with preference 3) a trunk is selected for the call if a trunk is available, and the search ends. Otherwise, the search continues.

All the preferences have been checked and no trunk group has been selected. This is where available trunks that were "recorded for future reference" in preferences 1 and 4 come into play. If the ***most desirable*** option (circuit switch the call) is not found, an ***acceptable*** option can now be selected from any available trunks located where ***insert Modem Pool*** was indicated.

User Operations

The operation of the Bearer Capability feature is automatic. Once this feature has been administered, no further user actions are required.

Considerations

Attendant Console Positions

An attendant console is always assigned to BCCOS 0 (voice only). This is automatic and cannot be administered or changed.

Information Transfer Capability Conversion

Because different vendors have interpreted international standards differently, an option was added to the Bearer Capability feature beginning with Generic 2.1, Issue 3.0.

Incoming or station originated calls are given a *call Type,* for call processing purposes, based either on ISDN messaging or on the default value assigned to the origination point's BCCOS. The standard choices are: ***Option 0,*** circuit switch the call, ***Option 1,*** insert a Modem Pooling conversion resource, or ***Option 2,*** block the call. The new choice, ***Option 3,*** forces the ISDN coding on the outgoing side of the call to a specific bearer capability, based on call type as shown in Table 23-M.

TABLE 23-M. Information Transfer Conversion

	ISDN Information Transfer Capability Coding for BCCOS Options	
Call Type	**Option 0 (Circuit Switch)**	**Option 3 (Circuit Switch-Conversion)**
Voice	Speech	3.1 KHz Audio
Voice Grade Data	3.1 KHz Audio	Speech
Unknown Analog	Speech	3.1 KHz Audio
Unknown Digital	As Determined from the default BCCOS	3.1 KHz Audio

Option 3 is only used in cases where an ISDN protocol conflict is preventing calls from completing. This condition can be suspected when early vintage BRI terminals are involved, or when calls to or from international destinations will not complete (for no obvious reason).

Multiple Appearances of an Extension

When multiple appearances of an extension are administered, each appearance must have the same BCCOS.

System 85, Release 2, Version 4 Equivalency

As stated earlier (under *Sources of Requirements Information),* ***if default BCCOS values are used, call routing will be equivalent to that in System 85, R2 V4.*** It is important to understand that default values refers to the initial values provided for the default BCCOSs with the switch. These values can be changed (although this is not recommended) and new BCCOSs can be added by switch administration. Any changes made will have an effect on call routing thereby causing call handling to differ between System 85, R2 V4 and Generic 2.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

AAR (Automatic Alternate Routing)

System 85, Release 2, Version 4

In System 85, R2 V4, AAR routing searches for a preference that supports the bearer capability requirement of the call being routed. Five bearer capabilities are recognized.

If a request is made for a trunk group to route a Mode 2 call, only preferences that can support Mode 2 data are searched.

Generic 2.1

In DEFINITY Generic 2.1, routing is based partly on BCCOS. Two methods are available for selecting a route based on BCCOS:

- The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (for example, Mode 2 data, 1200 bps, restricted channel, etc.). If a match is found and a trunk is available (other factors such as FRL permitting), the action taken is "circuit switch the call".
- If a match is not found, the algorithm attempts to connect the call to a preference for which the action to take ***is not*** "block the call". With currently available options, this would be a preference where the action is "insert modem pool".

ARS (Alternate Route Selection)

System 85, Release 2, Version 4

In System 85, Release 2, Version 4, ARS routing searches for a preference that supports the bearer capability requirement of the call being routed. Five bearer capabilities are recognized.

If a request is made for a trunk group to route a Mode 2 call, only preferences that can support Mode 2 data are searched.

Generic 2.1

In DEFINITY Generic 2.1, routing is based partly on BCCOS. Two methods are available for selecting a route based on BCCOS:

- The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (for example, Mode 2 data, 1200 bps, restricted channel, etc.). If a match is found and a trunk is available, other factors such as FRL permitting, the action taken is "circuit switch the call".

- If a match is not found, the algorithm attempts to connect the call to a preference for which the action to take ***is not*** "block the call". With currently available options, this would be a preference where the action is "insert modem pool".

ACD (Automatic Call Distribution)

The Bearer Capability feature has no direct impact on the ACD feature. An ACD Split does not have a BCCOS; therefore, Bearer Capability is not checked when routing calls to an ACD Split. Agent extension numbers, however, do have BCCOSs, and Bearer Capability is checked for calls routed directly to an agent extension number or originating from an agent extension.

Automatic Callback

The Bearer Capability feature functions normally with the Automatic Callback feature in most instances. When the callback call is placed by the switch, the Bearer Capability associated with the originally called station is used to place the call. In most instances this will pose no problems.

However, when both the called and calling terminals are ISDN—BRI Voice / Data stations administered with a single extension number (BCCOS 2), a complication can arise. In this instance, there is no way of telling if the original call was voice or data (original bearer capability IE is not preserved). The call back call could be either a voice or a data call, without regard to what type of call was originally placed.

Call Vectoring

The Bearer Capability feature has no direct impact on the Call Vectoring feature. A vector does not have a BCCOS. Therefore, Bearer Capability is not checked when routing calls to a Vector Directory Number. However, if a "route to" step within a vector involves the insertion of a Modem Pooling conversion resource, then the Modem Pooling feature requires that Bearer Capability be checked.

Data Call Setup

The Data Call Setup feature works with Bearer Capability in much the same way as it did before the Bearer Capability feature was introduced. However with Bearer Capability, Data Call Setup will work better. Both on and off net data calls will complete more efficiently and there will be a lower percentage of failed calls.

Voice Terminal Data Call Setup

One effect of Bearer Capability is that ***Data Preindication*** on voice terminal originated calls (from DCP terminals) becomes more important. If Data Preindication is not used, call processing uses the BCCOS of the voice terminal to set up the connection. There are two situations where this could result in a failed call:

- ***Off-Premise Calls***

  If a Modem Pooling conversion resource is required, **it will not be provided,** resulting in call failure. This is because the call setup information (using the voice terminal BCCOS) does not indicate that a conversion resource is needed.

- ***Local Call to a Data Port***

  If data preindication is not used, the call will fail. This is because the call setup information indicates a voice call to a data port which obviously won't work. Reorder tone will be returned.

Host Computer Access

The Bearer Capability feature supports the Host Computer Access feature in much the same way as Modem Pooling. That is, calls are routed to Host Computer Access ports based on the call setup message (for ISDN calls), optional query information when required (for data module originated calls) and, as a last resort, BCCOS for all calls.

The Bearer Capability feature also makes possible the provision of local Modem Pooling support for Host Access calls if required. As mentioned in the Data Call Setup feature interaction, the Bearer Capability feature makes ***Data Preindication*** mandatory for DCP voice terminal originated calls to Host Computer Access ports.

ISDN—BRI (Basic Rate Interface)

The Bearer Capability feature is fully compatible with the ISDN—BRI feature. Generally, calls originating from BRI terminals do not use the BCCOS assigned to the terminals extension. The BRI provides bearer capability requirements information for each call in the call setup message (see "Sources of Bearer Capability Requirements Information" under Description). The BCCOS of incoming calls is, however, used by the BRI to determine the appropriate response to incoming calls. In the case of a voice / data station, the BCCOS may also be used for terminal selection.

ISDN—PRI (Primary Rate Interface)

The Bearer Capability feature is fully compatible with the ISDN—PRI feature. Call setup messages received over ISDN—PRI facilities contain bearer capability requirements information, and BCCOS data is converted (through interworking) to ISDN-type information for calls outgoing on ISDN—PRI facilities.

Modem Pooling

The Bearer Capability future significantly enhances the operation of the Modem Pooling feature. With Bearer Capability, the insertion of an appropriate Modem Pooling conversion resource is more reliable, and there is less likelihood that one will be inserted if it is not really needed. That is, Modem Pooling conversion resources are inserted as a last resort only after all other alternatives have been checked. In DEFINITY Generic 2, the Modem Pooling selection process is part of the call routing process, rather than a separate switch action.

For a Modem Pooling conversion resource, Bearer Capability (data characteristics) is checked only for the digital member of the conversion resource. When assigning BCCOS to a conversion resource, the analog member should always be assigned BCCOS 5 (to properly populate the BC IE in the call setup message). If different characteristics are to be assigned to different groups of conversion resources (synchronous, mode 3, restricted, etc.), these distinctions should be made on the digital side as this is the only side that will make a difference when a conversion resource is selected.

Remote Access

The Bearer Capability feature applies to Remote Access trunks in the same way that it applies to any other type of trunk. For general purpose Remote Access facilities, an unknown type BCCOS (3 or 4) will normally be used. Other BCCOSs can be used when the remote access facility is used for special purposes only. For example, BCCOS 1 can be used for Remote Access trunks that are used only for digital mode 2 data.

WCR (World Class Routing)

On Generic 2.2 switches, the WCR feature replaces the AAR and ARS features. With the World Class Routing feature, route selection is based partly on BCCOS. Two methods are available for selecting a route based on BCCOS:

- The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (for example, Mode 2 data, 1200 bps, restricted channel, etc.). If a match is found and a trunk is available (other factors such as FRL permitting), the action taken is "circuit switch the call".
- If a match is not found, the algorithm attempts to connect the call to a preference for which the action to take ***is not*** "block the call". With currently available options, this would be a preference where the action is "insert modem pool".

Hardware Requirements

None.

Feature Administration

The Bearer Capability feature is assigned on a per extension, trunk group, and network routing preference basis using Manager II. Initial assignment is by ***default*** (see Tables 23-B through 23-J).

The switch administrator can modify the initial default BCCOS assignments and create new BCCOSs using the following administration procedures.

ADMINISTRATION PROCEDURES BEARER CAPABILITY		
PROCEDURE	**WORD**	**PURPOSE**
000	3	Assigns single terminal (extension) translations, including Bearer Capability Class of Service.
014	1	Assigns new BCCOSs (Bearer Capability Classes of Service) and changes call characteristics, transport mode, channel type, and default Bearer Capability Requirements values for existing BCCOSs. Also specifies actions taken for calls with different BCCOSs.
014	2	Assigns or changes data characteristics for a BCCOS.
100	2	Assigns a BCCOS to a trunk group.
288	2	Activates (or deactivates) recording of BCCOS for the Variable Format Call Detail Recording feature (Encode 31).
309	5	For Generic 2.1 switches, assigns a BCCOS (Field 6) to an ARS preference.
318	2	For Generic 2.2 switches, assigns a BCCOS (Field 3) to a WCR preference.
321	5	For Generic 2.1 switches, assigns a BCCOS (Field 5) to an AAR preference.

Bridged Call

Description

The Bridged Call future allows voice terminal users who share an appearance to bridge onto an existing call on that shared appearance. A 2-party call becomes a 3-party call.

This feature is useful for monitoring a call, note taking, or consulting by a third party. Executives, secretaries, and consultants are potential users of this feature.

Shared Appearances

Shared appearances must be assigned to enable the Bridged Call feature. Shared appearances consist of two or more images (points of access to an appearance). In practice, a user who initiates a bridged call enters an active connection by going off hook on an image of one of the appearances already involved in the call.

> **NOTE:** The terms "image", "appearance", and "extension" appear throughout this manual. Refer to Appendix E, ***Images, Appearances, and Extensions*** for a description of how these terms interrelate.

Feature History and Development

The Bridged Call feature was first available for System 85 in Release 1. Initially, single-appearance voice terminals were not allowed to participate in bridged connections.

Beginning with Release *2*, Version 2, Issue 1.2 of System 85, one single-appearance voice terminal [administered as a Straight Line Set (SLS)] was allowed to share an appearance with as many as 15 multiappearance voice terminals.

Beginning with DEFINITY Generic 2.1, Issue 3.0, a limited version of a bridged image is allowed for ISDN—BRI data appearances.

User Operations

The following are the user operating procedures for this feature.

With a Multiappearance Voice Terminal

To Bridge Onto an Existing Call:

1. Press the busy appearance button. [Green status lamp is already lit. The associated red status lamp lights].
2. Go off-hook. [Bridged connection is made].

To Exit From a Bridged Connection:

Go on-hook, press another appearance button, or press the **[HOLD]** button.

With a Straight Line Set

To Bridge Onto an Existing Call:

Go off-hook. [Bridged connection is made].

To Exit From a Bridged Connection:

Go on-hook.

Considerations

Feature Parameters

There can be as many as 12 appearances of an extension and up to 16 images (shared appearances) of each appearance. There can be as many as 192 images of an extension (12 appearances x 16 images per appearance). It is possible (although not recommended) to have each of the 192 possible images assigned to a different voice terminal.

More typically, an extension will have 3 or 4 appearances. An image of each appearance can be assigned to as many as 16 different voice terminals.

Each set of up to 16 images ("shared appearances") can contain 1 image assigned to a straight line set. The same appearance cannot have more than one image on any single voice terminal.

Bridging with Straight Line Sets

A straight line set is a single-appearance voice terminal that ***appears*** to System 85 or Generic 2 as a multiappearance voice terminal. Straight line sets possess ***some,*** but by no means all, of the capabilities of multiappearance voice terminals. The bridging provided by straight line sets is limited by the design of single-appearance terminals. Single-appearance terminals do not have status lamps to indicate the busy/idle status of their single appearance. As such, the user of a straight line set must:

- Go off-hook to determine the busy/idle status of the appearance.
- Wait until the shared appearance is idle to place a call.
- Choose between ringing, abbreviated ringing, delayed ringing, and the absence of ringing for the voice terminal. Visual alerting is not provided.

Bridging by Second Party

Only one bridged connection is allowed per appearance. When a third party attempts to bridge onto a call using an appearance that has already been bridged, the third party receives reorder tone.

In a conference call (established via Conference—Three Party or Conference—Attendant Six Party), one bridged connection is allowed for each conferee. If another party attempts to bridge on to a conference call using a third image of an appearance that has already been bridged, the bridging party receives reorder tone.

Bridging Across Modules

Bridging across modules on a System 85 or Generic 2 is allowed. That is, a voice terminal user located on one module can bridge onto a call connecting voice terminals on different modules (as many as four different modules could be involved).

Class of Service

Class of service is assigned to an extension number. As every image is a representation of an appearance of an extension number, the same class of service applies to all images (of every appearance) of the same extension number. This means that all images of the same extension number have the same class of service permissions and restrictions.

Arrangement of Shared Appearances

When assigning shared appearances (images) of an extension number to multiappearance voice terminals, it is generally preferable to assign images of the same appearances to each terminal. That is, the shared appearances of an extension should be the same on all terminals. This practice avoids confusion and access and status indication problems.

Hard and Soft Processor Swaps

If a voice terminal user is bridged onto a shared appearance when a hard processor swap occurs, this connection will endure the hard processor swap.

A voice terminal user cannot bridge onto a shared appearance during a hard processor swap.

The Bridged Call feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and Generic 2 feature affect or are affected by the operation of this feature.

APLT (Advanced Private Line Termination)

The APLT Restriction is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When the APLT Restriction is assigned to a ***shared extension,*** the restriction applies to every image of the extension.

Attendant Call Waiting

Attendant Call Waiting is partially allowed toward shared extensions with only one appearance. When only the straight line set is active on this type of shared extension, the switch allows an attendant call to wait. However, when a multiappearance terminal is active on the shared extension, Attendant Call Waiting is denied. The switch returns busy tone.

While an attendant-extended call is waiting on an SLS, a multiappearance voice terminal can be used to bridge onto the active call. However, the waiting call cannot be retrieved until the multiappearance voice terminal leaves the connection.

Attendant Call Waiting is denied toward shared extensions with more than one appearance. Instead, attendant calls are routed to an idle appearance (if available) of the shared extension. When every appearance is busy, the switch returns busy tone.

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, the ARS Toll Restriction is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When ARS Toll Restriction is assigned to a ***shared extension,*** the restriction applies to every image of the extension.

Call Waiting

Call Waiting is partially allowed for shared extensions with only one appearance. When only the straight line set is active on this type of shared extension, the switch allows an incoming call to wait. However, when a multiappearance terminal is active on the shared extension, Call Waiting is denied. Busy tone is returned to the calling party. These attendant-extended calls can also return to the attendant-console via the Timed Reminder feature. Further, the attendant is allowed to ***reextend*** the call for further waiting even though a multiappearance voice terminal is bridged to the connection. However, when a multiappearance voice terminal is bridged to the connection, the Attendant Call Waiting tone will not be repeated.

While a call is waiting on an SLS, a multiappearance voice terminal can be used to bridge onto the active call. However, the waiting call cannot be retrieved until the multiappearance voice terminal leaves the connection.

Call Waiting is denied to shared extensions with more than one appearance. Instead, incoming calls are routed to an idle appearance (if available) of this type of shared extension. When every appearance is busy, the switch returns busy tone to the calling party.

Conference—Attendant Five Party

For voice terminals with bridge-on capability, it is possible to have the bridged-on party involved in an attendant established conference, if the conference limit (5) has not already been reached.

Conference—Attendant Six Party

For voice terminals with bridge-on capability, it is possible to have the bridged-on party involved in an attendant established conference though not actually connected to the conference circuit. However, transmission quality may degrade.

Conference—Three Party and Transfer

Bridging is allowed onto an appearance that is being held for conference.

For multiappearance voice terminals while bridging is active, the CONFERENCE and TRANSFER buttons are inoperable for both a controlling terminal (extension that originated the bridged call) and a bridged terminal. A user can bridge on after a conference is established.

For a straight line set while bridging is active, pressing the RECALL button (if provided) and momentarily pressing the switchhook are ineffective for both a controlling terminal and a bridged terminal. A user can bridge on after a conference is established.

Data Protection

Attempting to bridge onto a call that has Data Protection activated is denied. The switch returns reorder tone.

Data Protection—Permanent is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When Data Protection—Permanent is assigned to a shared extension, this protection applies to every image of the extension.

DID (Direct Inward Dialing)

The DID Restriction is assigned to a class of service in Procedure 010, Word 3. of service is then assigned to an extension in Procedure 000, Word 1. When Restriction is assigned to a shared extension, this protection applies to extension.

Extension Number Portability

Every image of an extension number must be removed before the extension number can be ported to another network node.

FRL (Facilities Restriction Level)

An FRL is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. This is the default FRL for all images of that extension. That is, the same default FRL applies to every image of that extension.

Hold

While an SLS has a call on "soft hold" (by pressing the RECALL button, or momentarily pressing the switchhook), bridging is not allowed for a multiappearance terminal sharing that appearance.

While an SLS has a call on "hard hold" (by dialing the Hold dial access code), bridging is allowed for a muitiappearance voice terminal sharing the appearance. The bridging multiappearance voice terminal is added to the active connection. However, the call on hard hold cannot be retrieved until the bridged multiappearance terminal leaves the connection.

ISDN—BRI (Basic Rate Interface)

ISDN—BRI voice terminals can be used with the Bridged Call feature. The feature works the same with a BRI terminal as it does for any other Multiappearance Voice Terminal with the following exceptions:

- Standard BRI (Voice or Data) Appearances

  When a shared ISDN—BRI appearance (not designated as a data appearance) is being used for a voice call, the Bridged Call feature functions normally for any image of that appearance. However, if a BRI appearance is being used for a data call, bridging onto that call is denied (reorder tone is returned). This protection is provided automatically and the Data Protection feature need not be used.

- BRI Data Appearances

  With ISDN—BRI appearances, selected appearances (one or more) on the station can be administered as data appearances. These data appearances are for data calls only and cannot be used for voice calls. While data appearances can be shared (have multiple images), these bridged images cannot be used to bridge onto an active data call (reorder tone will be returned). This protection is automatic and the Data Protection feature is not required on these appearances.

Intercom—Automatic

Bridging onto an existing Automatic Intercom call is allowed. A maximum of three multiappearance voice terminals may be connected to the call at any one time. Additional parties attempting to bridge onto the connection receive reorder tone.

Intercom—Dial

Bridging onto a Dial Intercom connection is not allowed.

Intercom—Manual

Bridging onto an existing Manual Intercom call is permitted. A maximum of three multiappearance voice terminals may be connected to a Manual Intercom call at any one time. Additional calls trying to bridge onto the call receive reorder tone.

Last Extension Dialed

The extension number that the Last Extension Dialed feature stores and redials is kept on a per-equipment location basis. Therefore, even though a voice terminal user shares an appearance with another user, the stored extension number cannot be overwritten (or used) by a user sharing the appearance on a different voice terminal.

Last Number Dialed

The dialed number that the Last Number Dialed feature stores and redials is kept on a per-equipment location basis. Therefore, even though a voice terminal user shares an appearance with another user, the stored number cannot be overwritten (or used) by a user sharing the appearance on a different voice terminal.

Override

The Override feature can be used toward a terminal that is in a normal talking state with another terminal, even if a third terminal is bridged onto the connection.

> **NOTE:** An override call to a straight line set enters the busy connection. An override call to a multiappearance voice terminal usually terminates to an idle appearance. However, if every appearance is busy, the override call enters the connection.

Priority Calling

Priority Calling is partially allowed toward shared extensions with only one appearance. When only the straight line set is active on this type of shared extension, the switch allows a Priority Call to wait. However, when a multiappearance terminal is active on the shared extension, the incoming Priority Call is denied. Busy tone is returned to the calling party.

While a priority call is waiting on an SLS, a multiappearance voice terminal can be used to bridge onto the active call. However, the waiting call cannot be retrieved until the multiappearance voice terminal leaves the connection.

Priority Calling is partially allowed toward shared extensions with more than one appearance. When one or more appearances of a shared extension is idle, incoming priority calls are routed to an idle appearance and 3-burst ringing is provided. When every appearance is busy, the switch returns busy tone to the calling party.

Privacy—Manual Exclusion

Bridging onto an appearance that has Manual Exclusion active is not allowed. The switch returns reorder tone to the party attempting to bridge.

If a party is already bridged onto a call when Manual Exclusion is activated, the switch removes the bridged party from the connection and returns reorder tone.

The Manual Exclusion feature, requiring an XCLUSION button, cannot be assigned to an SLS.

Restriction—Attendant Control of Voice Terminals

The Attendant Control of Voice Terminals feature allows an attendant to apply a restriction to an extension or a predefined group of extensions. When this type of restriction is applied to a shared extension, the restriction applies to every image of that extension.

Restriction—Code Restriction

A Code Restriction level is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When a Code Restriction level is assigned to a shared extension, the level applies to every image of that extension.

Restriction—Miscellaneous Trunk Restrictions

Miscellaneous Trunk Restrictions are assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When these restrictions are assigned to a shared extension, the restrictions apply to every image of that extension.

Restriction—Toll Restriction

Toll Restriction is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When Toll Restriction is assigned to a ***shared extension,*** the restriction applies to every image of that extension.

Restriction—Voice Terminal Restrictions

Voice Terminal Restrictions are assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When one (or more) of these restrictions is assigned to a ***shared extension,*** the restriction applies to every image of that extension.

Serial Calls

Bridging is allowed for a shared appearance that is involved in a Serial Call.

Tenant Services

There are no tests in System 85 or Generic 2 software to ensure that the images of an appearance (or that the appearances of an extension) do not cross partition boundaries. It is the responsibility of the system manager to ensure that every image of each extension resides only on voice terminals that are used by a particular extension partition.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, the WCR Toll Restriction is assigned to an extension class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When WCR Toll Restriction is assigned to a ***shared extension,*** the restriction applies to every image of that extension.

Hardware Requirements

Multiappearance voice terminals are primarily used for this feature. However, one single-appearance terminal can share an appearance with as many as 15 multiappearance terminals.

Also, hard-wire bridging (connecting tip and ring leads together) of up to two single-line terminals is allowed using the SN228, SN229, and TN742 (previously allowed using SN221 and SN222).

Feature Administration

With one exception (ISDN—BRI data appearances), the Bridged Call feature is provided whenever shared appearances (multiple images of an appearance) are provided.

On System 85 switches, the Bridged Call Features administered using the Maintenance and Administration Panel (MAAP). The customer can also administer this feature using the System Management Terminal (SMT) or the Terminal Change Management (TCM) feature.

On DEFINITY Communications System Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

Administration Procedures
Bridged Call Feature

Procedure	Word	Purpose	SMT
051	1	Assigns a single-appearance voice terminal as type "SLS" for use in bridging.	Yes
052	1	Assigns images (shared appearances) of an appearance.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Bridged Call	
Path Name	**Purpose**
terminal-change terminal equipment	Assigns images (shared appearances) of an appearance. (The terminal type of a straight line set is "No-Button.")
terminal-change terminal line-pickup	Displays or prints a report of bridging assignments.

Busy Verification of Lines

Description

This features allows the attendant to check the status of an apparently busy extension number. This check verifies that the called extension number is busy or appears busy. The extension number can appear busy when the terminal is left off-hook or is in need of maintenance. Being able to verify whether a line is really busy or not increases the efficient use of time and equipment.

Before allowing the attendant to connect onto a 2-way connection, the talking parties hear a burst of warning tone. This tone is reapplied at 15-second intervals as long as the attendant is on the connection. The first burst of tone lasts for 2 seconds. Thereafter, the duration is 1/2 second.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Busy Verify a Line From an Attendant Console:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights.]
2. Press **[VERFY]** . [VERFY lamp lights.]
3. Press **[START]** , or press the DXS group selection button. [If START was pressed, dial tone is heard. If DXS group selection button was pressed, the group indicator lamp lights.]
4. Dial the terminal's extension number, or press the DXS specific terminal line button.
5. One of the following situations occur:
   - The line is busy on a normal 2-party call. [A 2-second warning tone is heard by the attendant and both parties. The attendant is bridged on the talking connection and can hear and be heard by the talking parties. A 0.5-second warning tone is heard by all parties every 15 seconds as long as the attendant is on the line.]
   - The line is idle. (The busy verification request is canceled, and the call proceeds as a normal attendant-originated call to an idle voice terminal). [VERFY lamp goes out, RING lamp lights, ringing is heard at the voice terminal, and the attendant hears ringback tone.]

- The line is in an out-of-service state. (An out-of-service state is when the terminal has remained off-hook longer than 10 seconds without dialing remained off-hook longer than 10 seconds between dialing digits, or remained off-hook longer than 10 seconds after the other party has gone on-hook). [Attendant is connected directly to the line. ANS lamp lights, and VERFY lamp goes out.]
- The line is in an unstable state. (See the "Considerations" section for unstable states).

6. When verification is finished, press RELEASE. [ATND and ANS lamps go out, and PA lamp lights.]
7. One of the following situations occur:
   - The verified line was busy. [The VERFY lamp goes out.]
   - The verified line was idle. [RING lamp goes out, ringing stops at the called voice terminal, and ringback tone is removed from the attendant.]

Considerations

Reorder Tone

If reorder tone is heard, the VERFY lamp goes out and the attendant is unable to verify the line because it is in an unstable or transient condition. The attendant should wait a few seconds and try again to verify the line.

Intercept Tone

If intercept tone is heard, the VERFY lamp goes out and the attendant is unable to verify the line.

Busy Verification Not Recognized

If an attendant presses the VERFY button without first selecting an idle loop, the request for busy verification is not recognized.

Dual Use of VERFY Lamp

When the switch is equipped with both the Busy Verification of Lines and the Trunk Verification by Attendant features, the same VERFY button and lamp are used for both features.

Request to Verify Another Line

If the attendant receives a request from a voice terminal user to verify another line, the attendant must release the incoming call or place the incoming call on hold before verifying the other line.

Warning Tones

Both warning tones are 440-hertz.

Hard and Soft Processor Swaps

If an attendant is verifying a line when a hard processor swap occurs, this connection will endure the hard swap.

An attendant cannot verify a line during a hard processor swap.

The Busy Verification of Lines feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

ACD (Automatic Call Distribution)

An attendant can use the Busy Verification of Lines feature to check the busy/idle condition of the terminals in an ACD split. The agent's terminals can be checked as to whether or not the terminals are available for split calls. If Busy Verification of Lines is activated toward an ACD associated extension number, only the split supervisor's line is verified.

Attendant Call Waiting

An attempt to verify a terminal line via the Busy Verification of Lines feature takes precedence over the Attendant Call Waiting feature.

Automatic Callback

While terminal B is on-hook waiting for terminal A to become idle after activating Automatic Callback toward terminal A, both terminals can be busy verified. But, when terminal A goes on-hook and the automatic callback process begins, neither terminal can be busy verified until the talking connection between terminals A and B has been established.

Call Coverage

If an attendant attempts to busy verify an extension classified as "extension busy" and "extension active", verification is allowed. However, verification is not allowed if the Busy Verification call is to an extension where all appearances are on hold.

A busy verification call to a busy extension that has coverage does not redirect to coverage.

Call Forwarding—Busy and Don't Answer

Busy Verification is allowed toward an extension, even if the extension has Call Forwarding—Busy and Don't Answer activated.

Call Forwarding—Don’t Answer

Busy verification is allowed toward an extension even if the extension has Call Forwarding—Don't Answer active.

Call Forwarding—Follow Me

Busy Verification is allowed toward an extension, even if the extension has Call Forwarding—Follow Me activated.

Call Park

An extension that is in Call Park cannot be busy verified.

Call Vectoring

An attendant is not allowed to activate busy verification toward a VDN. If this is attempted, the switch returns intercept tone to the attendant.

Call Waiting

If an attendant attempts to busy verify an extension that already has a call waiting, busy verifiction proceeds normally. However, if the attendant attempts to busy verify an extension that is waiting for another extension, the busy verification attempt is denied.

Code Calling Access—Universal

Busy verification is denied toward an extension that has accessed code calling. Attempts to do this result in reorder tone.

Conference—Attendant Five Party

Busy Verification of Lines is denied when attempted toward a line connected to a Conference—Attendant Five Party call.

Conference—Attendant

Busy Verification of Lines is denied when attempted toward a conference call made by the Conference—Attendant Six Party feature.

Conference—Three Party

Busy Verification of Lines is denied when attempted toward an extension connected to a Conference—Three Party call, unless that extension appears on a multiappearance terminal with more than one appearance. In this case, the verification attempt routes to an idle appearance of the extension.

Data Protection

When Data Protection (either Permanent or Temporary) is active on an extension (or an active connection), Busy Verification of Lines toward that connection is denied.

DDC (Direct Department Calling)

An attendant can use the Busy Verification of Lines feature to check the busy/idle condition of the terminals in a DDC group. However, this feature cannot check for the line "made busy" to group calls. If Busy Verification of Lines is activated toward a DDC group number, only the controlling terminal line is verified.

EUCD (Enhanced Uniform Call Distribution)

An attendant can use the Busy Verification of Lines feature to check the busy/idle condition of the terminals in an EUCD split. The agent's terminals can be checked whether or not the terminals are available for split calls. If Busy Verification of Lines is activated toward an EUCD associated extension number, only the split supervisor's line is verified.

Hold

If terminal A has terminal B on hold, terminal A can be busy verified but terminal B cannot (unless the extension being verified has multiple appearances).

Hunting

Hunting is not performed when the Busy Verification of Lines feature is activated toward a terminal line in a hunting group.

Line Lockout

Busy Verification of a locked-out voice terminal makes an immediate talking connection to the locked-out terminal. No barge-in tone is heard. The ANS lamp lights at the console.

Loudspeaker Paging Access

Reorder tone is received when attempting to busy verify a terminal line that has accessed Loudspeaker Paging (making a page or waiting for answer-back).

Override

When busy verifying a called extension, the switch denies Override to that extension.

Privacy—Attendant Lockout

The attendant is prevented from busy verifying a terminal line that is connected to a loop held on the attendant console.

Privacy—Manual Exclusion

Activating Manual Exclusion on an extension does not prevent an attendant from busy verifying a line.

Priority Calling

A verification attempt made via the Busy Verification of Lines feature has precedence over a priority call on the same line. However, if an attendant attempts to busy verify a terminal line that is waiting for another line, the busy verification attempt is temporarily denied and routed to reorder tone.

Serial Calls

The switch denies Busy Verification of Lines toward a line involved in a serial call.

Tenant Services

An attendant (in a partition other than Attendant Partition 0) is allowed to verify any extension residing in Extension Partition 0 or residing in an extension partition that has been assigned to the attendant's partition. When the attendant tries to verify an extension in another partition, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to verify any extension in the switch.

Timed Recall on Outgoing Calls

The switch denies busy verification if the terminal line being verified is connected to a call which has already been switched to an attendant position by means of the Timed Recall feature. Busy verification is allowed on terminal-to-trunk calls that have not yet been switched to the attendant by this feature. The recall timing is suspended as long as the attendant remains bridged on the connection by busy verification and is resumed when the attendant releases.

UCD (Uniform Call Distribution)

An attendant can use the Busy Verification of Lines feature to check the busy/idle condition of the terminals in a UCD group. However, this feature cannot check for the line "made busy" to group calls. If Busy Verification of Lines is activated toward a UCD group number, only the controlling terminal line is verified.

Restricting Feature Use

The Busy Verification of Lines feature can be denied toward terminal lines which have the Voice Terminal Restriction feature active for the assigned class of service.

Hardware Requirements

None.

Feature Administration

Assignment of the Busy Verification of Lines feature is on a per-console basis.

On System 85 switches, the Busy Verification of Lines feature is administered using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE BUSY VERIFICATION OF LINES		
PROCEDURE	**WORD**	**PURPOSE**
203	1	Assigns the VERFY button to the attendant console(s). The applicable encode is as follows. 7 VERFY Button.

Notes:

Call Coverage

Description

General Usage

This feature provides alternate answering points for calls that might otherwise go unanswered. For a principal (user with Call Coverage active), Call Coverage provides automatic redirection of calls that meet specified conditions. Calls are redirected to a coverage path. The coverage path(s), don't answer interval, and coverage criteria (conditions) combine to make a coverage group. A coverage group is assigned to a principal. The same coverage group can be assigned to more than one principal.

The Coverage Path

A single coverage path can consist of up to three coverage points:

- The primary covering user
- A backup to the primary covering user
- Another individual or general group of pooled answering positions, such as a Message Center or an AUDIX as a final covering point.

A coverage group can have a single or a dual coverage path.

Coverage Points

A coverage point can be:

- An individual voice terminal (covering user)
- Automatic coverage provided by the AUDIX system, or
- Coverage service provided by a group of voice terminals.

Both AUDIX and Message Center group coverage are provided by ACD (Automatic Call Distribution) splits (EUCD, UCD, or DDC in switches prior to Release 2, Version 3).

When highly interactive coverage is needed, the primary covering user should be someone who is aware of the principal's schedule and can determine the relative importance of calls to the principal (such as a personal secretary or executive assistant). A group or pooled answering arrangement can provide general coverage. This type of coverage can provide such services as information on when the principal will return, record who called, and take messages from calling parties to the principal.

Priority Versus Nonpriority Calls

Priority calls (calls that terminate with 3-burst ringing) do not route to coverage. Nonpriority calls are redirected to coverage based on the assigned criteria. These criteria define the conditions under which nonpriority calls, directed to a principal's extension, are redirected to coverage.

Coverage Criteria

There are four specific coverage criteria or types of coverage. The criteria and conditions are:

- ***Cover Active***

  When any appearance of an extension with cover active is busy, all nonpriority calls (of the specific type) to that extension are redirected to coverage. Cover Active can be assigned to a single-line voice terminal, a single-appearance extension, or extensions with multiple appearances.

- ***Cover Busy***

  When a single-appearance extension has Cover-Busy assigned and that extension is busy, incoming nonpriority calls (of the specified type) redirect to coverage. These calls are treated the same as ***Cover Active.*** However, if there are multiple appearances of the extension and any of the appearances are idle (except origination only appearances), calls do not redirect to coverage. Incoming calls alert at an idle appearance. When every appearance of the extension is busy, nonpriority calls (of the specified type) redirect to coverage.

- ***Cover Don't Answer***

  Calls to an extension assigned coverage alert at the principal's voice terminal for the Don't Answer Interval (from two to six ringing cycles). If unanswered during this interval, nonpriority calls (of the specified type) redirect to coverage.

- ***Cover All***

  All nonpriority calls (of the specified type) directed to a principal immediately redirect to coverage. If Cover All is assigned to an extension, this criterion takes precedence over the other criteria.

These criteria can be applied separately to the following ***specific types*** of calls.

- Internal calls (calls originating within the switching system)
- External calls (such as, attendant extended and trunk calls),

  or

- Both internal and external calls.

In this manner, specific types of nonpriority calls can be sent to coverage in different ways. For example, within a single coverage group, Cover Busy can apply to internal calls, Cover Active can apply to external calls, and Cover Don't Answer can apply to both internal and external calls.

Dual Coverage Paths

An extension can have up to two coverage paths. These dual coverage paths can provide separate coverage for different types of calls. For example, calls from within a DCS network can be routed to automatic coverage by the AUDIX system, while calls from the public network are routed to a Message Center agent. This use of dual coverage paths provides external callers, who are less likely to know how to deal with AUDIX, with personal and flexible interaction with agents, while providing internal callers with efficient and effective access to automatic coverage.

Another use of dual coverage paths is to cover calls according to separate criteria. For example, an executive can have one path that routes calls to the secretary and then to Message Center, and another path that routes calls directly to Message Center. Cover Active can be assigned to the first path, while Cover Don't Answer is assigned to the second path. If available, the secretary is instructed to pickup unanswered calls for the executive. In this way, when the executive is busy on a call, subsequent calls will route to the secretary and then to Message Center without ringing the executive's voice terminal. Meanwhile, unanswered calls can be quickly covered either by the secretary or a Message Center agent.

Call Coverage Functions

To satisfy specific Call Coverage needs, seven functions are available to voice terminal users:

- Caller Response Interval

  The caller response interval is an administrable period of time (from 2 to 10 seconds), beginning with a coverage tone, that allows the caller to make a choice of responses to a call that is being redirected to coverage. The response interval can be administered as 0 seconds. When this is done, the tone is given, but calls redirect to coverage immediately. The coverage tone is a single burst of tone that notifies the caller that the call is being redirected to cover. The Caller Response Interval and coverage tone do not apply to outside (trunk) calls or attendant-Originated calls.

  During the Caller Response Interval, the caller can select one of the following options:

  — Terminate the call by hanging up

  — Activate Leave Word Calling

  — Wait for the covering user to answer.

- Consult/Return

  This function allows the covering user to call the principal for private consultation whenever a principal's call is redirected to coverage. The calling party is put on hold and a 2-party connection is established between the principal and covering user. At the principal's discretion, the covering user can add the calling party to the conversation, or can return the call to the principal.

Consult/Return calls are treated as priority calls that override coverage. Like other priority calls, Consult/Return calls can terminate to an originating (only) appearance.

- Coverage Callback

  This function allows covering users to leave a message for the principal to call back the calling party. This function is available only when the incoming call (call being directed to coverage) originated locally (within the same switching system). For external calls (incoming trunk calls including DCS calls originating on a different node), the Leave Word Calling feature can be used.

- Send All Calls (Prior to R2 V4)

  This function allows a principal or covering user to redirect calls to coverage for one extension or all extensions on a given voice terminal. This is essentially the same as ***Cover All*** except that it can be turned on and off as desired at the principal's terminal rather than having to be assigned in translation (see "Feature Administration"). Send All Calls can be activated at any time, including when a call is ringing or when the principal is active on a call. If activated when a call is ringing, the call immediately redirects to coverage.

  Send All Calls can be used in association with the Consult function. In this way, a principal could send every call to coverage, but request to be notified when a specific call arrives.

- Send All Calls (Beginning With R2 V4)

  This function allows a principal or covering user to redirect calls to coverage for ***one*** extension, a ***group*** of extensions, or ***all*** extensions on a given voice terminal. This is essentially the same as ***Cover All*** except that it can be turned on and off as desired at the principal's terminal rather than having to be assigned in translation (see "Feature Administration"). Send All Calls can be activated at any time, including when a call is ringing or when the principal is active on a call. If activated when a call is ringing, the call immediately redirects to coverage.

  Send All Calls can be used in association with the Consult function. In this way, a principal could send every call to coverage, but request to be notified when a specific call arrives.

  — Send All Calls Extension

    In Release 2, Version 3 and earlier switches, only one Send All Calls button was available. This button activates the Send All Calls function for every extension appearing on a voice terminal. For example, a secretary's or assistant's voice terminal has an image of a supevisor's extension. When the secretary or assistant presses the Send All Calls button, Send All Calls is activated for both the secretary's extension and the supervisor's extension. Calls for the supervisor are sent to coverage as well as calls to the secretary. The supervisor may be available to answer calls and may not want calls sent to coverage. The enhanced Send All Calls Extension function allows the secretary to use the Send All Calls function without necessarily activating Send All Calls for the supervisor.

The new Send All Calls Extension button (SAC XXXX) allows calls to a specific extension to immediately redirect to coverage without impact on other extensions that appear on the same terminal. (XXXX represents the extension number; "X" stands for any number 0 through 9).

Each extension on a multiappearance voice terminal can have a separate Send All Calls Extension button (SAC XXXX). Other terminals with shared appearances of the same extensions can have SAC XXXX buttons for the same extensions. Pressing the SAC XXXX button activates or deactivates Send All Calls for ***every appearance of that extension*** (including the shared appearances of that extension).

The feature lamp beside the SAC XXXX button lights when Send All Calls is active. This lamp also lights on all terminals with SAC Extension buttons for the same extension. These lamps go dark when Send All Calls is deactivated.

The same multiappearance voice terminal can have as many Send All Calls Extension buttons as there are individual extensions on the terminal.

If the voice terminal does not have a Send All Calls Extension button, Send All Calls can still be activated or deactivated by dialing the appropriate dial access code. The effect on shared appearances on other terminals is the same as if a SAC XXXX button had been used.

— Send All Calls Group of Extensions

In R2 V4, the Send All Calls Group of Extensions (SAC GROUP) button activates the Send All Calls function for a "designated" group of extensions on a multiappearance voice terminal. The status of other extensions (not in the specifically designated SAC Group) remains unchanged.

A multiappearance terminal can have only one Send All Calls Group of Extensions (SAC GROUP) button.

To "designate" an extension as a member of a Send All Calls group, an image of the first appearance of the extension must appear on the terminal with the SAC GROUP button. If one appearance of the extension is assigned to the group, all other appearances (shared appearances) are also assigned to the group.

Pressing the SAC GROUP button activates or deactivates the Send All Calls function for the group and all individual extensions belonging to the group. Only the terminal assigned the SAC GROUP button can activate or deactivate Send All Calls for the group.

The green status lamp beside the SAC GROUP button lights when the button is pressed to activate Send All Calls. When the button is pressed again, Send All Calls deactivates, and the green lamp goes dark.

There is no dial access code to activate or deactivate send all calls on a group basis.

An ***individual extension*** in a send all calls group can also be assigned a Send All Calls Extension (SAC XXXX button. This button can activate or deactivate Send All Calls for that specific extension. For example, if a secretary presses the SAC GROUP button to redirect all calls to coverage, a supervisor belonging to the group can deactivate Send All Calls by pressing his/her own Send All Calls Extension button. Calls to that specific extension are not then redirected to coverage (on a send all calls basis). This has no effect on the other members of the send all calls group. Their calls still redirect to coverage.

- Temporary Bridged Appearance

This function allows a principal (with a multiappearance terminal or a straight line set) to bridge onto a coverage call by going off-hook on the appearance that has been redirected to coverage. However, the coverage point that answered the redirected call cannot be a coverage group (such as, AUDIX or Message Center) for the Temporary Bridged Appearance function to work. Also, when a VDN (Vector Directory Number) is assigned as the final point in a coverage path, a principal's temporary bridged appearance is removed at the time that vector processing assumes control of a redirected call.

- Implied Principal Addressing

This function returns the extension-number address of the principal to whom a coverage call was originally directed. This address is used for the Consult function and to deliver Coverage Callback and Leave Word Calling messages to the pricipal. For example, if Leave Word Calling is provided, messages can be left for the principal at any point in the coverage path.

- Ring Ping

For R2 V4, an optional ring ping on immediately redirected calls (send all calls function) is a reminder to the principal that send all calls is active. This option is assigned on an extension class of service basis rather than an individual terminal basis. The principal's terminal receives a quick burst of ringing (0.1-second tone) when a call is immediately redirected to coverage. This is an especially helpful reminder to the user that has Send All Calls active on a terminal that does not have feature lamps (such as single-appearance terminals). This is also useful for shared appearances where a different user can activate send all calls.

This tone is identical to the tone provided when Call Forwarding—Follow Me is active. Ring ping is given when calls are directed due to any of the following:

— Cover All

— Cover Active

— Cover Busy

— Send All Calls.

Ring ping is provided on a line class-of-service basis. All voice terminals sharing the terminating appearance of the extension receive ring ping except as follows:

— The terminal is an off-hook, single-appearance set.

— The terminal is an on-hook, single-appearance set that is an image of a multiappearance terminal that is off-hook.

— The terminal is already ringing.

— The terminal is administered not to ring (alert type = 0) for this extension.

— The Ringing—Cutoff feature is active on the terminal.

Use of the Display— Voice Terminal Feature

If an internal caller's voice terminal has a display capability, the caller receives a visual indication that the call has redirected to coverage. If the covering user's voice terminal has a display capability, the display identifies the called principal, the calling party, and the reason the call routed to coverage (see the Display—Voice Terminal feature).

Feature History and Development

The Call Coverage feature was first introduced in System 85 in Release 1, Version 3. Subsequent enhancements that affect this feature include the following:

- Introduction of the "dual coverage paths" functionality providing separate coverage for different types of calls. Dual coverage paths were first available in Release 2, Version 2.
- Introduction of the AT&T Personal Terminal 510D, and the 7407D IDT (Integrated Display Terminal) providing additional terminals with enhanced call coverage capabilities. The 510D was first available in Release 2, Version 2 with a retrofit capability to earlier versions. The 7407D was introduced with Release 2, Version 3.
- Enhancement of the Display—Voice Terminal feature to include scrolling capabilities improved the services available to covering users equipped with this type of terminal. Scrolling was first available in Release 2, Version 3.
- Improvement of the Send All Calls function in Release 2, Version 4, with the introduction of two Send All Calls feature buttons:

  — Send All Calls Extension button

  — Send All Calls Group of Extensions button.

- Addition of an optional "Ring Ping" when calls are immediately redirected using Send All Calls. Ring ping for Send All Calls is first available in Release 2, Version 4.

User Operations

The following are user operations associated with the Call Coverage feature. Except for Send All Calls, these operations are performed by a covering user. The Send All Calls operations can be performed by either the covering user or the principal.

The Consult Function

To activate the Consult function from a multiappearance voice terminal:

1. Press the **[TRANSFER]** (or **[CONFERENCE]**) button. [Dial tone] (Calling party is put on hold).
2. Press the **[CONSULT]** button. [3-burst ringing at principal's extension] (Priority call is directed to principal).
3. Obtain handling instructions from the principal.

To establish a 3-way talking connection (principal, caller, and covering user):

Press the **[CONFERENCE]** button. (Calling party is returned to the connection).

To establish a 2-party connection between principal and calling party:

Press the **[TRANSFER]** button. [DiaI Tone] (Principal and calling party are connected).

To drop the principal and return to the calling party:

After a 3-way connection has been established:

Press **[DROP]** .

Before a 3-way connection is established:

Press the held (incoming) appearance.

To Activate the Coverage Callback Function:

Press the **[COVERAGE CALLBACK]** button.

To Establish a Leave Word Calling Message for the Principal to Call the Covering User:

Press the **[LEAVE WORD CALLING]** button.

Send All Calls Function (Prior to R2 V4)

To activate Send All Calls for all extensions on a principal's terminal:

Press the **[SEND ALL CALLS]** button.

To deactivate Send All Calls for all extensions on a principal's terminal:

Press the **[SEND ALL CALLS]** button.

To activate Send All Calls for a specific extension:

1. Go off-hook on the selected extension. [Dial tone]
2. Dial the Activate Send All Calls access code. [Confirmation tone]
3. Go on-hook.

To deactivate Send All Calls for a specific extension:

1. Go off-hook on the selected extension. [Dial tone]
2. Dial the Cancel Send All Calls access code. [Confirmation tone]
3. Go on-hook.

The Send All Calls Function (Beginning With R2 V4)

To activate Send All Calls for a single extension:

1. Press the appropriate SAC XXXX button. [The lamp for the SAC XXXX button lights.]

   If the extension is part of a Send All Calls Group, activating Send All Calls for the group will not alter the state of the extension. That is, Send All Calls remains active for this extension.

   or

   Go off-hook on the selected extension. [Dial tone]
2. Dial the Activate Send All Calls access code. [Confirmation tone]
3. Go on-hook.

To deactivate Send All Calls for a single extension:

1. Press the appropriate **[SAC XXXX]** button. [The lamp for the SAC XXXX button goes dark.]

If the extension is part of a Send All Calls Group and the SAC GROUP button is pressed to deactivate Send All Calls for a group, the state of this extension is not changed. That is, Send All Calls remains deactivated for this extension.

or

Go off-hook on the selected extension. [Dial tone]

2. Dial the Cancel Send All Calls access code. [Confirmation tone]

3. Go on-hook.

To activate Send All Calls for a group of extensions:

Press **[SAC GROUP]**. [The lamp for the SAC GROUP button lights, and all the assigned SAC XXXX feature button lamps of extensions included in the group light on the terminal. The lamps for the SAC XXXX buttons on all terminals with shared appearances of the extensions included in the group light.]

To deactivate Send All Calls for a group of extensions:

Press **[SAC GROUP]**. [Lamp for the SAC GROUP button goes dark, and all the assigned SAC XXXX feature button lamps included in the group go dark. Lamps for the SAC XXXX buttons on all terminals with shared appearances of the extensions included in the group also go dark.]

To deactivate Send All Calls for a single extension in a group when Send All Calls is active for the group:

Option 1:

Press the **[SAC XXXX]** button for the extension. [The lamp goes dark. Send All Calls has been deactivated for this extension.]

Option 2:

1. Go off-hook at the desired extension. [Dial tone]
2. Dial the SAC-Cancel dial access code.
3. Go on-hook. [The lamp goes dark. Send All Calls has been deactivated for this extension.]

NOTE: Pressing the SAC GROUP button to deactivate Send All Calls for the group, after deactivating Send All Calls for the extension (either option), does not change the status of the extension. That is, Send All Calls remains deactivated for the extension.

Message Retrieval and Display

To access principal's message file as a covering user:

1. Press an idle appearance button.
2. Press **[COVER MSG RETRIEVAL]**. [Display shows WHOSE MESSAGES?]

3. Dial principal's extension number (when display shows MESSAGES FOR [Name]).
4. Press **[NEXT MESSAGE]** .

To access principal's message file during call with principal:

1. Press **[HOLD]** . [Principal put on hold.]
2. Press an idle appearance button. [Dial tone]
3. Press **[COVER MSG RETRIEVAL]** . [Display shows WHOSE MESSAGES?]
4. Dial principal's extension number.
5. Press held appearance. [Principal returns to connection.]

To return call for principal to displayed message originator (Callers extension number has appeared in the message):

1. Press **[TRANSFER]** . [Principal put on hold; dial tone returned.]
2. Press **[RETURN CALL]** . [Callback call is initiated]
3. Press **[TRANSFER]** . [Callback call and principal are connected.]

The Lock / Unlock Function

To lock the terminal (blocks message retrieval):

1. Go off-hook on an idle appearance. [Dial tone]
2. Dial the LOCK access code. [Confirmation tone returned.]

   This terminal cannot now be used to perform any Message Retrieval functions.

To unlock the terminal (return message retrieval):

1. Go off-hook on an idle appearance. [Dial tone]
2. Dial the UNLOCK access code. [Confirmation tone returned.]

Considerations

Unavailable Terminals

A voice terminal in the coverage path is unavailable to receive coverage calls when one of the following conditions exists.

- The covering extension is busy (every appearance of the extension is in use on multiappearance voice terminals or off-hook on a single-line voice terminal).
- Call Forwarding, either Busy and Don't Answer or Follow Me, is active for the covering extension.

- Send All Calls is active for the covering extension or the entire covering user's voice terminal.
- The coverage point is in an out-of-service condition.

If Cover Don't Answer is active on an unattended voice terminal and the entire coverage path is unavailable, the call cannot redirect to coverage. The call continues ringing on the original called extension.

When the principal's voice terminal is busy or Send All Calls is active, and the entire coverage path is unavailable, the call cannot redirect to a covering user. In this case, the roller receives busy tone.

Soft Numbers for Covering Users

Sometimes, a covering user answers redirected calls for numerous (perhaps, as many as 25) principals. In this situation, the covering user answer redirected calls using a voice terminal equipped with a coverage module. Generally, "soft number" are used to supply the identity of the originally called principal to the covering user.

The term "soft numbers" should not be misunderstood. These numbers are ***real*** extension numbers. They are part of the numbering plan, and consume some of the usable extension numbers within a System 85 or DEFINITY Generic 2 switch. However, soft numbers for Call Coverage are typically assigned special characteristics. Usually, soft numbers have only one image of a single appearance. Each soft number is assigned to a principal's coverage path. The image of the soft number is usually located on a coverage module, and the adjacent appearance button is usually labeled with the principal's name (not the number). Ringing is usually (but not necessarily) assigned to the image of the soft number. Also, to prevent unauthorized calling activity from soft numbers, these numbers can be assigned to a common class of service that includes Origination Restriction or Outward Restriction.

Covering User Send All Calls (Button Activated)

Before R2 V4, if the covering user has an image of an appearance of the principal's extension and activates Send All Calls using the SEND ALL CALLS button, the Send All Calls function is in effect for the principal's extension as well as the covering user's extension. This problem can be avoided by using the feature dial access code rather than the SEND ALL CALLS button.

Send All Calls Activated at Coverage Point

If a coverage point has Send All Calls active, the point is not eligible to receive a coverage call.

The following list shows the alternate treatments for cases where a covering user has Send All Calls active. These specified treatments are based on the assumption that the alternate voice terminal has at least one idle appearance.

- If there is only one coverage point, the principal's voice terminal rings.
- If the coverage point is the final (not the only) point, the previous coverage point rings.
- If there is a Subsequent coverage point in the path, the coverage point with Send All Calls active is skipped.

If a situation arises where the alternate voice terminal(s) does not have an idle appearance, the switch returns busy tone to the calling party.

Send All Calls Without a Coverage Path

A multiappearance voice terminal user can activate Send All Calls on an extension that has no coverage path assigned. Calls placed directly to such an extension have nowhere to go so they terminate on an idle appearance on the terminal. However, if this extension is in a coverage path, the extension is ineligible to receive coverage calls while Send All Calls is active. Therefore, a covering user can activate Send All Calls (with no path) as a temporary break from answering coverage calls. (During this time, direct calls to the covering user can still complete to the covering user's voice terminal, but on a different appearance).

Temporary Bridged Appearances and Single-Appearance Terminals

Normally, Temporary Bridged Appearances are not provided for single-appearance voice terminals. Therefore, by default, a principal with an analog voice terminal cannot bridge onto coverage calls. However, when a principal's analog voice terminal is administered as a straight line set, the Temporary Bridged Appearance function is provided (except status-lamp operation).

The DCS (Distributed Communications System) Environment

In a DCS, the Caller Response Interval, Coverage Tone, and Coverage Callback are provided for calls between nodes (for example, between a System 85 and a DEFINITY Generic 2 in a DCS cluster) for calls that are redirected to coverage. An internode call that redirects to coverage uses internal criteria (the criteria for an internal call). Also, the coverage points in a principal's coverage path must be located on the same node as the principal.

Message Waiting Lamps

As many as 10,500 automatic message waiting lamps can be assigned to voice terminals within the switch.

Coverage Groups

As many as 3047 separate coverage groups can be assigned to a switch. The first 1999 groups are reserved as single-path groups, and the remaining 1048 group pairs are reserved as dual-path groups. A given coverage point can be assigned only once in a coverage path or group.

Dual Coverage Path Assignments

Dual coverage paths are assigned using pairs of groups numbered from 2000 to 4095. Each dual path pattern is composed of two groups: one even numbered and one odd numbered. The first group in a dual path pattern must be the even numbered group, and the second path is the immediately following odd numbered group. This results in a total of 1,048 dual path patterns composed of 2,096 groups (two groups to a pattern).

When there is a conflict between criteria in dual coverage paths (for example, both paths are assigned as Cover Active for internal calls), the call will redirect according to the first (even-numbered) path.

Single Path Functionality

If necessary, coverage groups from 2,000 to 4,095 can be given single path functionality. This is done by assigning the same criteria and path for both the even group number and the corresponding odd group number.

Coverage Paths Without Criteria

It is permissible for no coverage criteria to be assigned to a single-path coverage group for an extension. If criteria are not assigned, the principal or covering user can activate Send All Calls to redirect calls to coverage.

It is also permissible to assign no criteria to a dual-path coverage group for an extension. When this is done, Send All Calls redirects internal caIls according to the first coverage path, while redirecting external calls according to the second coverage path.

Attendant-Extended Calls

If an attendant originates or extends a call to a principal and that call redirects to coverage, the Temporary Bridged Appearance is not present on the principal's voice terminal.

Call Coverage cannot be assigned to an attendant position. Also, an individual attendant console cannot be assigned as a coverage point. However, with Call Vectoring, coverage calls can be redirected to the attendant group (queue). See Call Vectoring under "Interactions With Other Features" for further details.

Message Center and ACD Splits

Message Center or other ACD splits can be assigned as coverage points. However, only one split can be assigned to a coverage path, and it must be the last point in the path.

Parallel Coverage Points

Parallel coverage can be set up by hard-wire bridging a second voice terminal to a coverage point (extension). When a call redirects to coverage, both voice terminals alert (ring) simultaneously, and either covering user can answer the call.

Leave Word Calling

Leave Word Calling complements the Call Coverage feature by allowing the calling party to request a return call during or before the coverage process.

Lack of Coverage Path

If Send All Calls is activated for an extension but the extension does not have a coverage path assigned, calls to that extension are not routed to coverage. The appropriate lamps still light. The called party's terminal rings, and the calling party receives ringback tone.

Terminals With One Extension

For R2 V4, a terminal that has appearances of only one extension should be assigned a SAC XXXX button. A SAC GROUP button for a single extension wastes call-processing time by having to search for group members.

Terminals With Shared Appearances

For R2 V4, shared appearances (images) assigned on a voice terminal should not be included in a Send All Calls group. This is not recommended because a person with an image of the same appearance on another terminal may be available to answer calls and may not want Send All Calls activated. Therefore, the Send All Calls Extension feature button should be assigned to these extensions.

Hard and Soft Processor Swaps

Coverage criteria and coverage path translations are stored in a translation portion of switch memory. Therefore, these transitions will endure a hard processor swap.

Send All Calls activations are stored in a status portion of switch memory. Therefore, if a terminal user activates Send All Calls and then a hard swap occurs, Send All Calls will not be activated after the hard swap is finished.

If a principal is already bridged onto a redirected call using a temporary bridged appearance when a hard swap occurs, this connection will endure the hard swap. However, a temporary bridged appearance cannot be used to bridge onto a redirected call during a hard swap.

If a hard swap occurs while a call is being redirected to coverage, the redirected call fails.

The Call Coverage feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Display

After pressing the STA ID button, the attendant display indicates the extension number the call redirected to rather than the originally called extension number.

Attendant Release Loop Operation

When an attendant extends a call to a voice terminal with assigned coverage (with criteria that apply to the call) and releases the call, the call will not return to the attendant queue. Instead, the call will redirect to coverage according to the assigned coverage path.

For a call from an internal voice terminal extended by an attendant to another internal voice terminal, the call can redirect to coverage as an attendant extended call (external call criteria). This occurs when the attendant does not release the call before the call type is checked for redirection to coverage.

ACD (Automatic Call Distribution)

An ACD split (including a Message Center split) can be assigned as the final point in a coverage path. However, the split number, rather than an associated extension number, is used to designate the coverage point. Therefore, all coverage calls that are redirected to an ACD split are placed in the nonpriority portion of the queue.

On an attendant-extended call to a principal with coverage assigned to an ACD split (including a Message Center split), the attendant receives coverage tone before the call is redirected to the ACD split. During the caller response interval, the attendant must release the call to allow the call to be queued. The attendant can release the call prior to the tone.

Automatic Callback

After a voice terminal user with coverage active activates Automatic Callback toward another voice terminal, the 3-burst callback call is treated as a priority call and does not route to coverage.

Automatic Callback cannot be activated toward a voice terminal whenever a regular nonpriority call would have redirected to coverage. Therefore, Automatic Callback cannot be activated toward a terminal that has Cover Active, Cover Busy, or Cover All assigned as the internal coverage criterion. Automatic Callback cannot be activated toward a terminal that has Send All Calls active. Also, Automatic Callback cannot be activated during the caller response interval of a coverage call. However, Automatic Callback ***can*** be activated toward a principal that is off-hook with either Cover Don't Answer or no criterion assigned as the internal coverage criterion.

Busy Verification of Lines

If an attendant attempts to busy verify an extension classified as "extension busy" and "extension active", verification is allowed. However, verification is not allowed if the Busy Verification call is to an extension where all appearances are on hold.

A busy verification call to a busy extension that has coverage does not redirect to coverage.

CDR (Call Detail Recording)

The extension number of the covering user, rather than the principal, is recorded by CDR when a call is redirected and answered by a covering user.

If a covering user answers a redirected call on a soft number, the ***primary extension number*** of the covering user's voice terminal (not the soft number or the principal's number) is recorded in the call record.

Call Forwarding—Busy and Don't Answer

Call Forwarding—Busy and Don't Answer takes precedence over Call Coverage in the following situations.

The following list shows the alternate treatments for cases where a covering user has Call Forwarding—Busy and Don't Answer active. These specified treatments are based on the assumption that the alternate voice terminal has at least one idle appearance.

- If Call Forwarding—Busy and Don't Answer is active for a principal who has coverage, calls to that principal's extension redirect according to call forwarding.
- If a coverage point (other than an ACD split) has Call Forwarding—Busy and Don't Answer active, the point is not eligible to receive a coverage call.
  - If there is only one coverage point, the principal's voice terminal rings.
  - If the coverage point is the final (not the only) point, the previous coverage point rings.
  - If there is a subsequent coverage point in the path, the forwarded coverage point is skipped.

If a situation arises where the alternate voice terminal(s) does not have an idle appearance, the switch returns busy tone to the calling party.

When a call is forwarded to a principal with coverage active, the forwarded call does not redirect to coverage. If the principal doesn't answer, the forwarded call will ring (until abandoned) at the principal's voice terminal. However, if every appearance of the principal's voice terminal is busy, the switch either continues ringing at the forwarding terminal in a don't answer condition or returns busy tone to the calling party if the forwarding terminal is busy.

Also, if Call Forwarding—Busy and Don't Answer is active for a group coverage point (ACD split), calls will cover to the split's queue and then intraflow according to call forwarding.

Call Forwarding—Don't Answer

Call Forwarding—Don't Answer takes precedence over Call Coverage in the following situations.

The following list shows the alternate treatments for cases where a covering user has Call Forwarding—Don't Answer active. These specified treatments are based on the assumption that the alternate voice terminal has at least one idle appearance.

- If Call Fowading—Don't Answer is active for a principal who has coverage, calls to that principal's extension redirect according to call forwarding.
- If a coverage point has call Forwading—Don't Answer active, the point is not eligible to receive a coverage call.
  - — If there is only one coverage point, the principal's voice terminal rings.
  - — If the coverage point is the final (not the only) point, the previous coverage point rings.
  - — If there is a subsequent coverage point in the path, the forwarded coverage point is skipped.

If a situation arises where the alternate voice terminal(s) does not have an idle appearance, the switch returns busy tone to the calling party.

When a call is forwarded to a principal with coverage active, the forwarded call does not redirect to coverage. If the principal doesn't answer, the forwarded call will ring (until abandoned) at the principal's voice terminal. However, if every appearance of the principal's voice terminal is busy, the switch continues ringing at the forwarding terminal in the don't answer condition.

Call Forwarding—Follow Me

Call Forwarding—Follow Me takes precedence over Call Coverage in the following situations.

The following list shows the alternate treatments for cases where a covering user has Call Forwarding—Follow Me active. These specified treatments are based on the assumption that the alternate voice terminal has at least one idle appearance.

- If Call Fowarding—Follow Me is active for a principal who has coverage, calls to that principal's extension redirect according to call forwarding.
- If a coverage point (other than an ACD split) has Call Forwarding—Follow Me active, the point is not eligible to receive a coverage call.
  - — If there is only one coverage point, the principal's voice terminal rings.

— If the coverage point is the final (not the only) point, the previous coverage point rings.

— If there is a subsequent coverage point in the path, the forwarded coverage point is skipped.

If a situation arises where the alternate voice terminal(s) does not have an idle appearance, the switch returns busy tone to the calling party.

When a call is forwarded to a principal with coverage active, the forwarded call does not redirect to coverage. If the principal doesn't answer, the forwarded call will ring (until abandoned) at the principal's voice terminal. However, if every appearance of the principal's voice terminal is busy, the switch returns busy tone to the calling party.

Also, if Call Forwarding—Follow Me is active for a group coverage point (ACD split), calls will cover to the split's queue and then intraflow or interflow according to call forwarding.

Call Park

When the Call Park feature is activated by a covering user, the Temporary Bridged Appearance at the principal's voice terminal is removed, and the principal is unable to bridge onto the parked call.

Call Pickup

Activation of the Call Pickup feature is denied when attempting to:

- Pick up a Temporary Bridged Appearance on a principal's voice terminal (that is, after the call has been redirected)
- Pick Up a coverage call during Coverage Tone or during the Caller Response Interval.

If a member of the covering user's pickup group uses Call Pickup to pick up a redirected call, the temporary bridged appearance on the principal's voice terminal remains intact. After the redirected call is picked up, the principal can still go off-hook on the temporary bridged appearance to join the call with the Call Pickup user.

Call Vectoring

When the Call Vectoring feature is administered, an ACD split number cannot be assigned as the final point in a coverage path. However, a vector directory number can be assigned as the final point in a coverage path. When this is done, the full flexibility of the Call Vectoring feature can be applied to the redirected call. The corresponding vector could be programmed to queue the redirected call to an ACD split (including an AUDIX or a Message Center split). Furthermore, the vector's processing could vary by time of day (to provide night service) or by the status of the split's queue (to provide intraflow or interflow).

A vector can have a "route to" step that routes calls to an extension with coverage assigned. When this is done, the destination extension's coverage is ignored.

If a redirected call covers to a VDN, the Call Vectoring feature screens these covered calls to limit undesirable vector treatment of the covered calls. A Call Vectoring subroutine quickly scans the VDN's vector a per-call basis to be sure that one of a set of vector steps will operate on the call before the covered call can terminate to the VDN in the coverage path. The allowable steps include:

- Queue to main split (with staffed agents) step
- Route to step
- Forced disconnect ***with*** recorded announcement step (beginning with R2 V4, Issue 1.2).

VDN Override does not apply to calls that are redirected to a VDN by the Call Coverage feature. For calls that are redirected to a VDN, the originally called terminal's name remains permanently attached to the call.

Coverage to the Attendant Queue

Call Vectoring can redirect coverage calls to the attendant queue. Attendant coverage can be beneficial for some System 85s or DEFINITY Generic 2s. Using this coverage the attendant group can serve as the final coverage point for an assortment of principals.

A VDN can be assigned as the final point in a coverage path. One of these VDNs can be assigned to a vector with a single "route to" step. The "route to" step within this coverage vector contains an Abbreviated Dialing list item that outpulses the attendant dial access code [usually "0", or a DID LDN (Listed Directory Number)].

Since "route to" steps can direct calls to DID LDNs, partitioned switches can also cover to the ***shared*** attendant queue. Each extension partition desiring attendant coverage can have a vector that directs calls to the LDN for the attendant partition assigned to that extension partition. In this way attendant coverage is a partitioned function of the Tenant Services environment.

Call Waiting

When a principal has a single-line voice terminal with Call Waiting assigned, the call waits on the voice terminal if the principal has Coverage—Don't Answer active and the principal is busy. If any other type of coverage is active, the call goes to coverage.

Centralized Attendant Service

A backup terminal in a Centralized Attendant Service arrangement can be assigned as a point in a coverage path. This terminal can also have coverage assigned. When this terminal has coverage assigned, calls directly addressed to that extension redirect normally. However, incoming attendant-seeking calls route to the backup terminal when coverage is active and do not redirect to coverage.

Conference—Three Party

The switch denies an attempt to activate a 3-party conference from a multiappearance voice terminal during the Caller Response interval of a coverage call. The switch ignores the button press.

When the covering user presses the CONFERENCE button after the call goes to coverage, the temporary bridged appearance is removed from the principal's voice terminal.

Data Call Setup

If a data module is in a coverage path, it is skipped during the search for an available coverage point. If a data module calls an extension with coverage active, the call does not cover.

DDC (Direct Department Calling)

See UCD interaction.

Display—Voice Terminal

If a multiappearance voice terminal user who is busy on a call receives a call on another appearance, the new call can redirect to coverage after the don't answer interval. However, when this occurs, the covering user (equipped with a display) to whom the call redirects will receive a redirection notification of "b" (for busy) rather than "d" (for don't answer).

DCS (Distributed Communications System)

In a DCS environment, the covering user and the principal must be located on the same node.

However, if Call Forwarding—Follow Me is active for a group coverage point (an ACD split or a dummy ACD split), calls will cover to the local split's queue and then can ***interflow*** to a distant DCS node according to call forwarding. (This effective DCS coverage operation is primarily recommended to divert coverage calls to a centralized AUDIX or Message Center split).

EUCD (Enhanced Uniform Call Distribution)

Same as the ACD interaction.

Extension Number Portability

If an extension is part of Call Coverage, either as a principal or a covering extension, it must be removed from coverage before it can be ported to a new node.

Hold

If a covering user puts a coverage call on hold and the principal then bridges onto the held call, the covering user who originally put the call on hold drops off and cannot reenter the connection.

Hunting

Normally, Call Coverage takes precedence over Hunting. That is, if the called extension has both Call Coverage active and has a hunt path assigned, an incoming call will follow the coverage path. However, if the coverage point is a VDN (ACD split) and there are no agents staffed, the call will follow the Hunting path.

When a call routes to coverage, Hunting is allowed from the last coverage point. From this point, hunting is limited to the first nine members of the hunt group following the coverage point.

ISDN—BRI (Basic Rate Interface)

Call Coverage works with ISDN—BRI stations in the same way that it does for other types of stations. ISDN—BRI voice stations can participate in coverage groups either as a principal or as a coverage point ISDN—BRI data stations are specifically blocked (in software) from using Call Coverage in any way. This includes blocking ISDN—BRI data calls directed toward an extension with coverage active from going to coverage.

Intercom—Auto, Dial, and Manual

Intercom calls do not route to coverage.

LWC (Leave Word Calling)

LWC messages are sent to the principal originally called, even when calls redirect to coverage. The only exception is that when calls are redirected to an attendant, Leave Word Calling is not allowed.

Coverage Message Retrieval

Coverage Message Retrieval is denied to any coverage user not in the principal's coverage path, except for "Global Retrievers."

LWC on the Switch

Leave Word Calling can be provided on a switch that has no AP. Messages are stored on the switch.

Global Retrieval

Global retrievers can retrieve messages for anyone in the switch. This ability is useful for attendants and message center agents who may support users for whom they are not covering users. (Note that message retrieval cannot be assigned to an attendant console; a separate display voice terminal or data terminal must be used for this purpose). Global retrievers need only one COVERAGE MSG RETRIEVAL button. The same button works for global retrieval as well as for coverage-path retrieval. All messages accessible by Coverage Message Retrieval can be accessed globally.

Look-Ahead Interflow

Look-Ahead Interflow is compatible with the Call Coverage feature. At a sending switch a VDN with a Look-Ahead Interflow "route to" step can be assigned as the final point in a coverage path. Moreover, when the Call Vectoring subroutine screens these redirected calls (on a call-by-call basis) to limit undesirable vector treatment of the calls, the subroutine will find the Look-Ahead Interflow "route to" step (whenever the interflow applies to the call) and allow the redirected calls to cover to the VDN. At this point, the "route to" step is executed for the redirected calls as it would be for direct calls to the VDN.

Main/Satellite/Tributary

Call Coverage cannot redirect a call over a tie trunk. Therefore, a terminal user at a satellite location cannot provide coverage for a terminal user at the main.

Override

An Override call does not route to coverage.

Precedence Calling

An incoming precedence call will route to coverage if the established coverage criteria are met, except that an incoming Precedence Call will not route to a group coverage (Message Center, ACD Group, or AUDIX).

Priority Calling

A Priority Call does not redirect to coverage. Only nonpriority calls (calls that do not terminate with 3-burst ringing) are routed to covering users.

Queuing

A ***Queuing Callback Call*** is treated as a priority call and does not route to coverage. These callbacks ring at the terminal that originally placed the call. Callback calls do not cover, even if coverage is in effect.

Queuing in a Main/Satellite Arrangement

If the original call was placed between a Main / Satellite switching arrangement and is placed in a ringback queue, the callback call appears to the switch as a normal tandem tie trunk incoming call and coverage occurs.

Restriction—Attendant Control of Voice Terminals

If Controlled Termination Restriction is activated toward an extension with coverage active, the restriction overrides, and calls to that extension do not route to coverage.

Restriction—Voice Terminal Restrictions

If Termination Restriction is assigned to an extension with coverage active, the restriction overrides, and calls to that extension do not redirect to coverage.

Ringing—Abbreviated and Delayed Ringing

The Send All Calls and Cover All functions of the Call Coverage feature take precedence over Abbreviated and Delayed Ringing. These functions control the redirection of ringing.

The Cover Don't Answer function of the Call Coverage feature takes precedence over Abbreviated and Delayed Ringing when the amount of ringing cycles used to time both features are equal.

The details of the don't answer condition are as follows. If the timing interval for the coverage group ***is less than or equal to*** the timing interval for Abbreviated and Delayed Ringing, terminating calls redirect to coverage without ringing the images(s) assigned delayed ringing. However, if the timing interval for the coverage group ***is greater than*** the timing interval for Abbreviated and Delayed Ringing, terminating calls first ring the abbreviated ringing image(s). Then, ringing transfers to the delayed ringing image(s), and these images ring for the rest of the Cover Don't Answer timing interval. After the Cover Don't Answer interval elapses, the call redirects to coverage.

Since Call Coverage takes precedence over Abbreviated and Delayed Ringing ring ping, if assigned in R2 V4, is heard for immediately redirected calls at the called party's terminal, except if delayed ringing (encode 2) is assigned in Procedure 052, Word 1, Field 11.

Ringing Cutoff

When Send All Calls and Ringing Cutoff are both active at a called voice terminal, ring ping is not provided, as the call redirects to coverage.

Ringing Transfer

Call Coverage takes precedence over ringing transfer. Calls will redirect to coverage when the coverage interval expires, regardless of the ringing state. Ringing Transfer will occur only if the transfer interval is shorter than the coverage interval.

Tenant Services

There are no tests in Procedure 011, Word 1 to ensure that every point in a coverage path belongs to the same extension partition. It is the responsibility of the system manager to ensure that coverage paths do not cross partition boundaries.

There are also no tests in Procedure 000, Word 2 to ensure that a coverage group is only assigned to extensions residing in the same extension partition. It is the responsibility of the system manager to ensure that a coverage group pertaining to one extension partition is not assigned to an extension in another extension partition.

Transfer

Any attempt to activate the Transfer feature from a multiappearance voice terminal during the Caller Response Interval of a coverage call is denied. The switch ignores the button press.

After a call goes to coverage, if the covering user presses the TRANSFER button, the temporary bridged appearance is removed from the principal's voice terminal.

When a local voice terminal user transfers an incoming trunk call to a local extension where dual coverage paths apply, the Call Coverage feature redirects the incoming trunk call according to the assigned criteria and path for ***internal*** calls.

UCD (Uniform Call Distribution)

The extension number associated with a UCD or DDC cannot be assigned coverage. Individual group member extension number can be assigned coverage for calls addressed directly to those extensions.

To assign a UCD or DDC group as a coverage point, the group number (for example, four for group 4 or five for group 5) is used rather than the extension number. A UCD or DDC group can be assigned only as the final point on a coverage path. The switch cannot redirect a call to coverage after the UCD/DDC feature has distributed the call to a group member.

The UCD/DDC group is classified as unavailable to serve as a covering user if the UCD/DDC group has been made unavailable via a dial access code.

Unattended Console Service—Preselected Call Routing

When an attendant console is in the night mode, calls routed to the preselected voice terminal from the unattended console do not redirect to coverage. Only direct calls to the preselected voice terminal assigned coverage are redirected.

Hardware Requirements

There are no special hardware requirements for the Call Coverage feature as such. While any voice terminal will meet the requirements of Call Coverage, the following combinations can provide enhanced capabilities for covering users:

- The 7205H Voice Terminal with the C201A Call Coverage Module

  This combination provides 20 additional 2-lamp appearances identified to specific principals to allow one voice terminal to cover a larger group of terminals. The 7205H is no longer available as new equipment.

- The 7404D VDS (Voice/Data Station) with the Z300B Messaging Cartridge and an associated EIA type data terminal such as the 513 BCT

  This combination provides an EIA data terminal with the capability of accessing and displaying messages using the Display Voice Terminal feature.

- The 7405D Voice Terminal with the C401A or C401B Call Coverage Module

  This combination provides 20 additional 2-lamp appearances identified to specific principals to allow one voice terminal to cover a larger group of terminals. Call Coverage modules are generally used by covering users.

- The 7405D Voice Terminal with the D401A Display Module

  This combination provides the covering user with identification of the calling party (internal calls) or outside source and the principal. It further provides the covering user, when permissions are granted, with the ability to access messages for supported principals.

  NOTE: The Call Coverage Module and the Digital Display Module are mutually exclusive. That is, the same terminal cannot use both modules at the same time.

- The 7406D With Display

  This is a digital voice terminal, smaller than the 7407D, with an optional built-in display module. It provides the covering user with identification of the calling party (internal calls) or outside source and principal. It provides the covering user, when permissions are granted, with the ability to access messages for supported principals.

- The 7407D IDT (Integrated Display Terminal)

  This is a digital voice terminal, similar to the 7405D, with a built-in display module. It provides the covering user with identification of the calling party (internal calls) or outside source and the principal. It provides the covering user, when permissions are granted, with the ability to access messages for supported principals.

- The 7434D Voice Terminal

  This is a digital voice terminal that provides 34 2-lamp buttons for call appearances. When these appearances are identified to specific principals, this voice terminal can serve as a coverage point for a large group of terminals. The 7434D can also be equipped with a D401A display module to provide a covering user with display information.

- The 7506 ISDN MDT (Modular Display Telephone)

  This is a digital voice terminal that uses the ISDN—BRI protocol. It is smaller than the 7507, with a built-in 2-line, 20 character per line display. It provides the covering user with identification of the calling party (internal calls) or outside source and principal. In an ISDN network, specific identification of an outside calling party may also be possible. The 7506 provides the covering user, when permissions are granted, with the ability to access messages for supported principals.

- The 7507 ISDN IDT (Integrated Display Telephone)

  This is a digital BRI voice terminal, similar to the 7506, with a built-in 2-line, 40 character per line display. It provides the covering user with identification of the calling party (internal calls) or outside source and identification of the principal. In an ISDN network, specific identification of outside calling parties may also be possible. The 7507 provides the covering user, when permissions are granted, with the ability to access messages for supported principals.

- The AT&T Personal Terminal 510D or 515 BCT (Business Communications Terminal).

  These are data terminals with integrated voice terminal capabilities. Both of these terminals can access the Display—Voice Terminal feature and, therefore, provide the same functionality as the 7405D Voice Terminal with D401A Digital Display Module or the 7407D IDT.

Feature Administration

Assignment of the Call Coverage feature is defined by the coverage path(s) and criteria for the group and then assigning the group to a principal.

On System 85 switches, the Call Coverage feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — CALL COVERAGE

PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the voice terminal class of service to an extension number.	Yes
000	2	Assigns a coverage group to an extension (for group numbers between 2000 and 4095, use an even group number). Also, designates whether a covering user is authorized Coverage Message Retrieval.	
010	1	Assigns Send All Calls to a voice terminal's class of service. Beginning with R2 V4, this procedure also assigns ring-ping tone to the class of service.	Yes
011	1	Administers the coverage path(s) and criteria for a coverage group. **NOTE:** When the Call Vectoring feature is administered, an ACD split number cannot be assigned as the final point in a coverage path.	Yes
051	1	Assigns coverage/display capabilities to the plug-in modules used on designated H/D Series voice terminals.	Yes
052	1	Beginning with R2 V4, assigns appearances of extensions on a multiappearance terminal as members of the Send All Calls group (Field 14).	Yes
054	1	Assigns the buttons to a multiappearance voice terminal for Call Coverage. The applicable encodes include: 19 Send All Calls Button (R2 V3 and earlier) 19 Send All Calls — Group of Extensions Button (R2 V4 and Generic 2) 20 Consult Button 23 Coverage Callback Button 29 Send All Calls — Extension (R2 V4 and G2).	Yes
054	4	Assigns the display feature buttons to the 7405D, 510D, or 515 BCT voice terminal, or to the D401A display module for the 7405D.	Yes
063	1	Assigns an automatic message waiting lamp to a voice terminal.	Yes

ADMINISTRATION PROCEDURES — CALL COVERAGE (Contd)			
PROCEDURE	WORD	PURPOSE	SMT
075	1	Displays all voice terminals having the same coverage group assignment.	Yes
275	3	Specifies the Caller Response Interval and the Coverage Don't Answer Interval.	Yes
350	1	Assigns the first digit of the dial access codes used with Call Coverage (if required).	No
350	2	Assigns the dial access codes for the Call Coverage feature. Applicable encodes are as follows: 55 Activate Send All Calls 56 Cancel Send All Calls.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — CALL COVERAGE	
PATH NAME	PURPOSE
terminal-change class-of-service attributes	Assigns Send All Calls to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number. Use this screen to assign a coverage group(s) to an extension number (for group numbers between 2000 and 4095, use an even group number). Also, use this screen to designate whether or not the line permits Coverage Message Retrieval.
terminal-change group coverage	Administers the characteristics of a coverage group (or pair of coverage groups).
terminal-change system parameters (select Call-Coverage option)	Assigns the Caller Response Interval and the Coverage Don't Answer Interval.
terminal-change terminal buttons	Assigns the Send All Calls, Consult, or Coverage Callback button to a multiappearance voice terminal. Also, use this screen to assign the display feature buttons to the 7405D, 510D, 515 BCT voice terminal, or the display module for the 7405D. This screen is also used to assign the Automatic Message Waiting lamp to a voice terminal.

Notes:

Call Detail Recording

Description

The CDR (Call Detail Recording) feature monitors and collects call information about selected off-premises voice and data calls (calls that use reportable trunk facilities). The CDR feature does not record call information on local (station-to-station) calls or calls placed over trunks that have not been specifically selected for recording (in Procedure 101, Word 1).

The recorded information is collected and processed by the switch for each call that uses a reportable trunk. When a reportable call is completed, the switch organizes the call information into a call record, also known as a CDR record. The CDR records are then sent to a recording and/or processing adjunct that is external to the switch, to be compiled and processed for management use.

The CDR records contain valuable data that management can use to:

- Computing call expenses
- Detecting malfunctions in the system
- Detecting abuse of expensive facilities
- Optimizing network facilities.

In addition to the information recorded on completed calls, the CDR feature can be administered to record ineffective call attempts. An ineffective call attempt record is generated for incoming and outgoing network (AAR, ARS, or WCR) calls that are not completed. See "Ineffective Attempt Call Record" in the "CDR Record Types" section. For more detailed information about record formats and peripheral devices, see the **Call Detail Acquisition and Processing Reference Manual,** 555-006-202.

Feature History and Development

During its evolution, CDR has used several names including:

- SMDR (Station Message Detail Recoding) introduced with System 85, Release 1
- CMDR (Centralized Message Detail Recording) introduce on a limited scale with System 85, Release 2, Version 1. The CMDR version is also referred to as CSMDR (Centralized Station Message Detail Recording).
- VFCDR (Variable Format Call Detail Recording) introduced with System 85, Release 2, Version 3.

Initially, these various forms of call detail recording were each listed as separate features. Because all three of these forms use the same the separate feature descriptions were combined under the single feature name CDR (Call Detail Recording), beginning with the DEFINITY Generic 2.1.

Even though every CDR record is generated by one software system inside the switch, the old feature names (SMDR, CMDR, and VFCRD) are still used to distinguish the types of CDR configurations. Because each of these three separate forms of CDR is still available, this feature description addresses each of the three configurations separately. The separate CDR configurations can be distinguished based on the following:

- Call Record Format (organization of the call information as sent from the switch)
- Output port(s) used (on the switch)
- Peripherals (local storage units and pollers).

Configuration Availability

- SMDR (Station Message Detail Recording) Configuration

  The SMDR configuration is available on all versions of the System 85 or DEFINITY Generic 2 switch. With the SMDR configuration, the switch is administered with a *15-word default format* and sends records out across the *SMDR port* to an *SMDR adjunct*. SMDR is the oldest and least efficient configuration of CDR. *See* the "SMDR (Station Message Detail Recording)" section for details.

- CMDR (Centralized Message Detail Recording) Configuration

  The CMDR configuration is available on all versions of the System 85, Release 2 or the DEFINITY Generic 2 switch. With the CMDR configuration, the switch is administered with an *18-word default format* and sends records out across the *CMDR/NCOSS ports* to a set of *94A LSUs* (Local Storage Unit) being polled by a *93B poller*. A 3B2 CDRP (Call Detail Record Poller) can also be configured to poll the 94A LSUs. *See* the "CMDR (Centralized Message Detail Recording)" section for details.

- VFCDR (VariabIe Format Call Detail Recording) Configuration

  The VFCDR configuration is available on System 85, Release 2, Version 4, and DEFINITY Generic 2 switches. When the terminology "VFCDR" is used, it typically means that the switch is administered with a *custom format* and sends records out across the *PCC (Processor Communications Circuit) port* to a *CDRU* (Call Detail Recording Unit) or a CDRU/S (CDRU/Small).

  A 3B2 CDRP may be added to poll the CDRUs. This method of centralized CDR is more efficient than the CMDR configuration. *See* the "VFCDR (Variable Format Call Detail Recording)" section for details.

DEFINITY Generic 2 Enhancements

The *ISDN MA UUI (Message Associated User-to-User Information) Counter* and the *ISDN Cause Value* data items first became available with the DEFINITY Generic 2.1 switch These data items record information specifically for ISDN—PRI trunks. *See* the "Recordable Data Items" section for details.

The Processor Communications Circuit, also first available with Generic 2.1, is capable of time stamping call records before they are sent to the peripheral.

The CDRP (Call Detail Record Poller) software package, used on the 3B2 series of computers, was also first available with Generic 2.1. This package supports polling of the CDRU or 94A LSU.

The CDR feature itself is not changed on the DEFINITY Generic 2.2 switch; however, the replacement of the networking features AAR (Automatic Alternate Routing) and ARS (Automatic Route Selection) by the new WCR (World Class Routing) feature, has a significant impact on the CDR feature. This impact is described as appropriate throughout this chapter and is summarized in the Interactions With Other Features section.

How This Feature Description is Organized

The CDR (Call Detail Recording) feature description is organized into these major sections:

- Description
  - Feature History and Development
  - Definitions and Acronyms
  - Recordable Data Items
  - CDR Record Generation
  - CDR Ports and Data Presentations From the Switch
  - CDR Record Types
  - CDR Record Formats
- SMDR (Station Message Detail Recording)
- CMDR (Call Message Detail Recording)
- VFCDR (Variable Format Call Detail Recording)
- User Operations
- General CDR Considerations
- Considerations That Determine the CDR Configuration
- Interactions With Other Features
- Hardware Requirements
  - SMDR Hardware Requirements
  - CMDR Hardware Requirements
  - VFCDR Hardware Requirements
- Feature Administration
  - FEAC (Forced Entry of Account Codes) Administration
  - CDR Configuration Administration.

Definitions and Acronyms

The following list can be used as a quick reference to some of the terms and acronyms used with CDR. See the Glossary at the back of Volume 2 for a more complete list of terms, abbreviations, and acronyms.

BCD (Binary Coded Decimal) — The data presentation used with the CDR records to represent four-bits as a decimal number.

BTC Protocol — A two-way communications language used by the PCC (Processor Communications Circuit) and the CDRU to ensure data integrity. The PCC using the BTC protocol is capable of sending BCD or ASCII characters.

CDAP (Call Detail Aquisition and Processing) — The process of obtaining CDR records from a switch, and sending them to a peripheral device for storage and processing.

CDR (Call Detail Recording) — The process of generating call records that contain information about trunk calls monitored by a System 85 or DEFINITY Generic 2 switch.

CDR Configuration — VFCDR, CMDR, and SMDR are the three CDR configurations. Each configuration consists of a CDR record format (administered on the switch), one or more CDR ports, and the peripherals capable of receiving the CDR records.

CDR Data Presentation — The form in which a CDR record is output from a device. Characteristics such as the character representation (BCD or ASCII), and envelope protocols (BTC) or direct output are data presentation characteristics.

CDR Record Format — The following characteristics determine the CDR record format for a particular switch:

- The number of data items in the records
- Where the data items are positioned in the call records
- The number of cells (four bits) each data item occupies in the call records.

CDR Record — The collected data (data items) which describes the characteristics of a trunk call or an attempt at a trunk call (if ineffective attempts are recorded).

CDRP (Call Detail Record Poller) — A 3B2 series computer running CDRP software. The CDRP software collects CDR records from one or more CDRUs or 94A LSUs. Typically, this configuration exists in a multiswitch network where the results are collated. A CDRP system provides multilocation customers with a centrally located device that collects call record data.

CDR Port — A physical point on the switch where CDR records are passed from the switch to a CDR link leading to a peripheral device.

CDRU (Call Detail Recording Utility) — A software package that runs on a 3B2 computer or 6386 PC for call detail recording applications.

CDRU/ S (Call Detail Recording Unit/ Small) — A device used for low-volume CDR applications.

Cell — A 4-bit partition (a half-byte, sometimes called a nibble) used in formatting CDR records. Each cell contains one alphanumeric digit. There are four cells in a CDR word.

CMDR (Centralized Message Detail Redording) — Sometimes called CSMDR (Centralized Station Message Detail Recording). CMDR is a CDR configuration in which the switch sends records across the CMDR/NCOSS ports to one or more 94A LSUs. The LSUs are then polled by a 93B. Typically this configuration is used in a multiswitch network where the results are collated. CMDR only supports 18-word formats with opcodes.

A CMDR system or the NCOSS (Network Control Operations Support System) provides multilocation customers with a centrally located device that collects call record data. With NCOSS the LSU is installed on the customer premises temporarily to collect sufficient data for their traffic analysis, network system analysis, and other studies to improve customer systems.

CAS (Call Accounting System) — Referred to as AT&T CAS, to distinguish it from the CAS (Centralized Attendant Service) feature. AT&T CAS is a PC based software diskette system that provides computerized accounting of business telephone costs. The CAS records and stores call records, assigns costs to calls and generates reports. CAS supports systems with CDR record volumes up to 150,000 per run, and up to 5,000 stations.

Custom Format — A CDR record format that has been administered through VFCDR (Procedure 288 Word 2).

Default Format — One of two fixed formats (15- and 18-word) that are administrable with a single operation in Procedure 288, Word 1, rather than being custom formatted through VFCDR.

DEFINITY Manager IV — An AT&T product that allows customers to administer, maintain, and monitor the switches and trunks in their networks. Of particular interest is the Cost Management application which provides customers with cost-accounting procedures similar to CDR.

Direct Output — An asynchronous data transmission. Direct output can be sent as formatted or unformatted ASCII or BCD data. Direct output can be sent from the switch to a CDRU, 94A LSU, or SMDR adjunct.

Encodes — Release 2, Version 4 introduced encodes to provide flexible formatting of data items for VFCDR. Each data item has a unique encode associated with it *(See* Table 27-C for a listing).

Fixed Format — See Default Format.

FEAC (Forced Entry of Account Codes) — FEAC is a function of the switch that forces the terminal user to enter an account code when making certain types of calls. *See* "FEAC" in the Administration section.

Ineffective Attempts — A condition where an incoming or outgoing call is not completed. Some ineffective attempts may be recorded by CDR.

LSU Local Storage Unit — Devices that store CDR records. The CDRU (Call Detail Recording Utility) and 94A LSU are examples of LSUs.

NCOSS (Network Control Operations Support System) — An AT&T service, used as required for system network sanity checking, that requires the use of buffered data collection with an LSU, usually the 94A LSU. *See* the "CMDR (centralized message detail recording)" section for details.

Opcode — Operation code. A 4-bit code placed in the left-most cell position of each word in the 15- and 18-word default formats. Opcodes are used for hardware control and to delimit format words. They do not appear on printed reports. The SMDR adjunct and the 94A LSU— equipment require the use of opcodes.

PCC (Processor Communications Circuit) — A CDR port circuit pack (TN474B) that provides data integrity-checking as well as data output. The PCC provides flexibility in data communications between the switch and the peripheral device. The PCC is required with the VFCDR configuration, but can be used with any CDR configuration.

Polled Output — Packed BCD (Binary Coded Decimal) record output that uses the DDCMP (Digital Data Communications Message Protocol). The NCOSS, Manager IV, and the 93B poller use the polled output option as a method of collecting records from LSUs through a dial up connection.

Poller — A device such as the 3B2 CDRP (Call Detail Record Poller) or the 93B poller that is capable of calling up the LSUs. The poller then collects and processes CDR records from one or more LSUs.

Recordable Data Items — Pieces of information (also known as call details) that can be recorded on trunk calls. The calling and dialed numbers are two examples of data items. Not all the data items can be recorded at the same time. A set of recordable data items must be chosen for any CDR application. The "Recordable Data Items" section lists all of the data items available.

SMDR (Station Message Detail Recording) — A CDR configuration that provides detailed call information on incoming and outgoing calls for selected trunk groups. The configuration consists of a switch sending 15-word default format call records across the SMDR port (TN403) to an SMDR adjunct, and possibly other peripheral equipment used for storage or processing.

Standard Format — The standard 18-word ISDN and standard 24-word CDR formats supported by standard AT&T processes and products. These formats are not default, and must be administered as a custom format. These formats are shown in the VFCDR section.

TELESEER® — A switch peripheral that processes CDR records for financial management of telephone expenses.

Unformatted (Direct Output BCD or ASCII) — A data presentation from the PCC that is sent with no spaces between each data item.

VFCDR (Variable Format Call Detail Recording) — A specific CDR configuration that provides detailed call information on incoming and outgoing calls for selected trunk groups. The configuration consists of a switch sending custom designed records (from 15 to 24 words) to a CDRU or other compatible peripheral. VFCDR allows customizing (through administration) of CDR record formats. VFCDR must be used with care when either the SMDR adjunct or the 94A LSU is used. These peripherals assume specific data formats.

Recordable Data Items

A data item is a piece of information about a trunk call or attempted trunk call. For example: the dialed number, the trunk DAC (dial access code), and the node number are some of the data items that the switch can record. These data items are organized into a format and sent out as a call record.

The VFCDR configuration is the only CDR configuration capable of recording any of the data items in the list. While the VFCDR configuration can record any data item on the recordable data items list, it cannot record all of the recordable data items at any given time. There are 75 recordable data items on the list. The VFCDR format can be administered to provide a maximum of 24 words. Each "word" consists of four 4-bit cells. A single data item may occupy one 4-bit cell (as a minimum) or could require as many as 15 or more cells (with a maximum of 31). The number of data items that can be contained in a VFCDR record is a function of the size of the specific data items selected for the record. The SMDR and CMDR configurations use fixed subsets of the list of recordable data items.

The following is an alphabetical list of the recordable data items and their descriptions.

- ***Access Codes***

  See Dial Access Codes or Trunk Access Codes.

- ***Account Code (Encode 9)***

  This data item contains a user-entered account number that further identifies the caller. Account codes are used primarily on outgoing calls, but may be used for incoming calls routed through a tandem. On System 85, Release 2, Version 2 and earlier switches, account codes were limited to 5 digits; therefore, on switches that use SMDR or CMDR requiring one of the default formats (either 15- or 18-word), the account code field is limited to 5 digits. The ability to use flexible account codes, of up to 15 digits, was introduced in System 85, Release 2, Version 3. If an account code longer than five digits is used with either of the default formats, it will override subsequent data items (authorization code, time-in-queue, and/or Facility Restriction Level) in the record, as needed.

  If an account code is not entered or is not used, this data item (if present) is blank on the call record.

  FEAC (Forced Entry of Account Codes): The administrator also has the ability to assign FEAC. The FEAC function requires the caller to enter an account code on specific calls. FEAC can be administered on an extension class of service or trunk group basis on all switches. For System 85 and Generic 2.1 switches, FEAC can also be assigned on a system class of service basis. For Generic 2.2 switches, assignment on a system class of service basis is replaced by assignment on a network basis. *See* "FEAC (Forced Entry of Account Codes)" in the Administration section.

- ***Agent Login (Encode 33)***

  If more than one agent is using a station during the course of a day, the VFCDR option can be administered to record an agent's login. The login number could be as many as four digits long. The agent login data item is typically used to record the login of an ACD agent. The agent login is recorded on both incoming and outgoing calls.

- ***Attendant Console Number (Encode 27)***

  This data item contains the attendant-console number if the call is handled by an attendant. If no attendant is involved in the call, this data item is blank.

- ***Authorization Code (Encode 10)***

  This 4- to 7-digit data item further identifies the caller and indicates caller assigned outgoing calling priveleges. If authorization codes are not used, this data item is blank.

- ***Call Duration-Hours (Encode 1)***

  The call duration-hours data item shows the number of full hours that the call was connected (range 0 through 9). This data item, used along with the *call duration-minutes* (encode 2) and the *call duration-tenths of minutes* (encode 3) data items, shows the amount of time a call was connected to within 6 seconds of accuracy.

  The call duration also indicates if a call was successfully completed. If call duration is zero, the call did not complete.

  The call-duration timing in the CDR record varies with the type of call as follows:

  — System 85, (Release 2 Version 3 and later) and DEFINITY Generic 2 switches have a variable call completion threshold capability. This can set a delay time between end of dialing and start of charging on either a per-trunk group or per-system basis. The administered time can be from 1 to 99 seconds in 1-second increments. If neither the trunk-group nor system time is administered, a 6-second default time is used automatically. If answer supervision is received (and the feature flag bit for answer supervision is set), the timing of the call is started or restarted with answer supervision rather than using a timer. Answer supervision timing overrides the administered timers.

  — Calls can be queued if the call-processing software is unable to find an available trunk. Queued calls do not start timing until they come out of queue. As a result, the call-duration time does not include the time in queue.

- ***Call Duration-Minutes (Encode 2)***

  The call duration-minutes data item shows the number of minutes that the call was connected (range 0 through 59). This data item, used along with the *call duration-hours* (encode 1) and the *call duration-tenths of minutes* (encode 3) data items, shows the amount of time a call was connected to within 6 seconds of accuracy (unless answer supervision is present). See *Call Duration-Hours (Encode 1)* for further details on call timing.

- ***Call Duration-Tenths of Minutes (Encode 3)***

  The call duration-tenths of minutes data item shows the number of 6 second increments (less than a full minute) that the call was connected (range 0 through 9). This data item, used along with the *call duration-hours* (encode 1) and the *call duration-minutes (encode 2)* data items, shows the amount of time a call was connected to within 6 seconds of accuracy (unless answer supervision is present). See *Call Duration-Hours (Encode 1)* for further details on call timing.

- ***Calling Number (Encode 8)***

  For fixed format configurations (SMDR, CMDR and the Recommended Standard Formats), this data item is the 4-digit extension number of the station that originates the call. For switches with 5-digit extension numbers, it is the last four digits of the 5-digit extension number that originates the call. For switches with 5-digit extension numbers, this data item is used in conjunction with the *calling number, ten-thousands digit* (encode 13), to provide the full 5-digit extension number. With VFCDR (custom format) this field can be made 5-digits long and the full extension number can be recorded in a single field.

  On switches prior to System 85, R2 V4, the calling number data item is used to record the trunk-group DAC *for incoming calls.* Beginning with R2 V4, the ISDN—PRI feature can provide the calling number on incoming ISDN calls. For incoming ISDN—PRI calls, the calling number filed records the actual calling number and a new data item is provided to record the incoming trunk-group DAC. The encodes for the incoming trunk-group DAC and the calling number data items are different. However, for non-ISDN applications, the incoming trunk-group DAC and the calling number data items are typically assigned to use the same CDR field. If the calling number data item is available, it overwrites the incoming trunk-group DAC.

- ***Calling Number, Ten-thousands Digit (Encode 13)***

  This data item is the leading or first digit of a 5-digit number. When 5-digit extension numbers are used, this data item is used along with a 4-digit *calling number* data item (encode 8) to record the full 5-digit extension number. This data item (if reported) is blank when a 4-digit dial plan is administered.

- ***Condition Code (Encode 4)***

  The condition code data item is a single digit alpha or numeric character that identifies specific information or records that appear on the record. Table 27-A contains descriptions of condition codes used with TN474B and TN403 output.

- ***Date-Day (Encode 23)***

  The date-day data item is a 2-digit data item that occupies two cells in the call record. It is used, in conjunction with the da *date-month* data item (encode 22) and the *date-year* data item (encode 24) to record the date of a call record. These data elements can be administered in any order (such as "dd mm yy", or "mm dd yy").

  Before Release 2, Version 4, peripheral hardware inserted the date into the CDR record. Beginning with R2 V4, switch software can generate the date.

TABLE 27-A. Call Detail Recording Condition Codes

Condition Code		
ASCII Form of Hexadecimal Output from Switch	**SMDR Unit Direct Output**	**Description**
1	A	Attendant-handled calls (except attendant conference calls).
2	B	Adding to the list of CDR monitored trunk groups.
3	C	Removing from the list of CDR monitored trunk groups.
4	D	A long hold time call (10 hours or more). Calls with this condition code and a duration of 9 hours, 59 and 9/10 minutes are reported each 10-hours. When these calls end, a final record with the remaining call duration is also reported.
5	E	Maintenance Test data record. Ignore in call data compilations.
6	F	System reload. Ignore in call data compilations.
7	G	Calls handled by a networking feature.
8	H	Calls that have been handled by the Queuing feature.
9	I	Incoming or tandem trunk calls.
A*	Blank	Outgoing calls.
C*	L	Conference Call—Attendant 6-Party. A separate call record is produced for each outgoing conference connection.
D*	M	Time-of-day plan change or control mode change data.
E*	N	Unsuccessful call, either incoming or an outgoing network (AAR, ARS, or WCR) call. Includes incoming call failures for any reason and outgoing calls routed to reorder tone or intercept.
F*	O	Unsuccessful outgoing network call (FRL too low).

* If more than one cell is administered to the condition code field, (VFCDR only) the decimal equivalent of the Hexadecimal value is output from the switch (for example, F = 15).

- ***Date-Month (Encode 22)***

The date-month data item is a 2-digit data item that occupies two cells in the call record. It is used, in conjunction with the *date-day* data item (encode 23) and the *date-year* data item (encode 24) to record the date of a call record. These data elements can be administered in any order (such as "dd mm yy", or "mm dd yy").

Before Release 2, Version 4, peripheral hardware inserted the date. Beginning with R2 V4, switch software can generate the date.

- ***Date-Year (Encode 24)***

  The date-year data item is a 2-digit data item that occupies two cells in the call record. It is used, in conjunction with the *date-day* data item (encode 23) and the *date-month* data item (encode 22) to record the date of a call record. These data elements can be administered to appear in any order (such as "dd mm yy", or "mm dd yy").

  Before Release 2, Version 4, peripheral hardware inserted the date into the CDR record. Beginning with R2 V4, switch software can generate the date.

- ***Dial Access Codes (1st through 24th) (Encodes 52 through 75)***

  Data elements used to produce the Reportable Trunk Groups Record which lists the DACs for trunk groups that are reportable under CDR. These data elements are not used in a normal call record.

- ***Dialed Number (Routed Number for WCR) (Encode 7)***

  This data element contains the number dialed by the caller or processed by the WCR feature. In Generic 2.2, the WCR feature may modify the dialed digits *during digit analysis*, the converted digits are stored in this data item. Other modifications that may be made by the WCR feature after digit analysis will appear in the "Digits Sent" data item. The "dialed number" can be recorded for incoming and outgoing calls.

  If the * or # are part of the dialed number, CDR records these characters as a hexadecimal C and E respectively. If the " # " is used as the end-of-dialing character, it is ignored by CDR data processing and will not show upon the record

- ***Extension Partition Number (Encode 29)***

  On partitioned switches (where the Tenant Services feature is active) the extension partition number data item records the extension partition of the station originating the call. The extension partition number can be as long as three digits. This data item is available only with VFCDR.

  For incoming calls:

  — If a call is originated from Remote Access facilities, the extension partition corresponding to the dialed authorization code (if any) is recorded in this data item.

  — If the call is DID (Direct Inward Dialing), this data item contains the extension partition of the extension or LDN (Listed Directory Number) to which the call terminates.

  — If the call tandems to another outgoing trunk or if the extension partition is not determined due to attendant assistance or other reasons, this data item is set to an "A" (hexadecimal).

  In all cases, the extension partition data item agrees with the extension number in the calling number or dialed number data items.

- ***FRL (Facilities Restriction Level) Used (Encode 12)***

  This data element is a 1-digit number that indicates the level of service associated with the calling privileges for the call. The FRL may be based on the extension used to originate the call (default FRL) or, if an authorization code is dialed, it may be the FRL associated with the authorization code used. FRLs are used to determine calling privileges for network route selection.

- ***Feature Flags (Encode 15)***

  The feature flags data item is a 1-digit hexadecimal code that represents the active and idle features that may be recorded by CDR. Feature flag codes and their meanings are shown in Table 27-B. Note that more than one feature flag can be active at a time. If none of the feature flags are active for a particular call, a zero will appear in the feature flags data item of the call record.

TABLE 27-B. Hexadecimal Code for Active and Inactive Feature Flags

Hexadecimal Code	Feature Flags			
	Invalidity Flag	Answer Supervision	Queuing Type	Data Call
0	Valid	Inactive	Off-Hook	Voice
1	Valid	Inactive	Off-Hook	Data
2	Valid	Inactive	Ringback	Voice
3	Valid	Inactive	Ringback	Data
4	Valid	Active	Off-Hook	Voice
5	Valid	Active	Off-Hook	Data
6	Valid	Active	Ringback	Voice
7	Valid	Active	Ringback	Data
A	Invalid	N/A	N/A	N/A

The following is an explanation of the meaning of the columns under Feature Flags in Table 27-B:

— Invalidity Flag: If set, this flag indicates that the feature flag information is invalid. It is the result of filling empty fields in the CDR with hexadecimal "A" characters.

— Answer Supervision Flag: This is a Feature Flag bit that was added in System 85, Release 2, Version 4. Answer Supervision is a condition that identifies when the called party answers a call. Answer supervision is available on outgoing, or incoming and outgoing trunks with E&M or ISDN signaling.

  Answer Supervision is also used to calculate call duration accurately rather than estimating call duration as with the administrable default timer. The answer supervision flag is set if (and when) answer supervision is returned for the call being recorded.

— Queuing Type Flag: This bit is reportable on incoming and outgoing calls. The "Time-in-Queue" data items shows if a call was queued or not. If a call was queued, the queuing type identifies type of queuing used.

— Voice/Data Call Bit: An active condition (set)= data call. Inactive condition (not set) = voice call. This bit is reportable on incoming as well as outgoing calls.

- ***Incoming Circuit ID (Encode 14)***

  This data item is used in conjunction with the incoming circuit ID (100ths digit) data item (encode 18) to identify the trunk within a trunk group that is used to carry incoming and tandem calls.

  The trunk number was expanded from 2-digits to 3-digits in System 85, Release 2, Version 3. Because of this change, there are two separate data items for the incoming circuit ID. For R2V3 and later System 85 switches and for Generic 2 switches, both data items must be recorded to identify the trunk used. With VFCDR, this field can be made 3-digits to record the full circuit ID in one field.

  For outgoing calls, the incoming circuit ID data item is blank.

- ***Incoming Circuit ID (100ths Digit) (Encode 18)***

  This data item is used in conjunction with the *incoming circuit ID* data item (encode 14) to identify the trunk within a trunk group that is used to carry incoming and tandem calls.

  The trunk number was expanded from two digits to three digits in System 85, Release 2, Version 3. Because of this change, there are two separate data items for the incoming circuit ID. For R2V3 and later System 85 switches and for Generic 2 switches, both data items must be recorded to identify the trunk used.

  For outgoing calls, the incoming circuit ID data item is blank.

- ***Incoming Trunk-Group Dial Access Code (Encode 25)***

  This data item contains the DAC of the trunk group used by an incoming call. It can be up to four digits long, depending on the administered data item length. If the data item length is administered for less than the DAC length, only the right-most (least significant) digits are recorded. This data item is available only with VFCDR on Release 2, Version 4 and later switches.

- ***IXC (Interexchange Carrier) Code/ ISDN Network Identifier (Encode 19).***

  This data item identifies the long-distance carrier (such as AT&T Communications, MCI Communications, or US Sprint Communications) used for a call.

  For System 85 and Generic 2.1, aliasing is typically used. For example, it is administered so that a "1" in the CDR record represents the vendor for that particular switch. Other vendors would be recorded as a 2, or 3, and so forth.

  For Generic 2.2, this data item may need to be expanded to three digits. The CICs (Carrier Identification Codes) that can be dialed are three digits. If the data item field width is less than that, truncation takes place leaving the least significant digit in the record and probably rendering the data useless.

If no interexchange carrier is used, the IXC code data item is blank. The IXC is recorded on outgoing calls only.

The ISDN Network Identifier is the same as the IXC code only expressed in ISDN message form for transmission over ISDN—PRI trunks.

- ***ISDN Bearer Capability (Encode 31)***

  The BCC is an ISDN concept that, among other things, identifies call types as voice or various types of data calls. The bearer capability concept, as such, is not used on System 85, R2 V3 and earlier switches. For non-ISDN facilities on System 85, R2 V3, the BCC is assumed by the switch. For non-ISDN facilities on DEFINITY Generic 2 switches, the BCC is determined by the BCCOS (Bearer Capability Class of Service). For ISDN calls and facilities, regardless of version, the BCC is obtained from the ISDN call setup message. For more detailed information on ISDN Bearer Capability see the BCCOS feature description.

- ***ISDN Cause Value (Encode 34)***

  The ISDN cause value is an information element of ISDN that describes what happened to a given ISDN call. This data item is used for diagnostics. Typically this data item will contain the number 16 meaning "Normal" call completion. *See* Appendix G for a list of ISDN cause values and their definitions.

- ***ISDN Network Service Value (Encode 28)***

  In System 85 Release 2, Version 4 the ISDN NS identified the network service [such as SDN (Software Defined Network), WATS, or MEGACOM WATS Service] used to complete an outgoing trunk call. This ISDN IE (Information Element) is also referred to as NSF (Network Specific Facility).

  For DEFINITY Generic 2, the ISDN NS is recorded for incoming as well as outgoing calls. The ISDN NS data item applies only to incoming and outgoing calls that use ISDN trunks.

- ***Node Number (Encode 30)***

  This data item contains the DCS node number of the switch generating the record. It is administered in Procedure 275, Word 3. This data item is used for networked switches.

- ***Outgoing Circuit ID (Encode 16)***

  This data item identifies the last two digits (the unit and 10s digits) of the trunk number that is used to carry outgoing calls. In System 85, Release 2, Version 3, trunk numbers were changed from two digits to three digits. Because of this change, a new data item was added (Outgoing Circuit ID, 100ths digit). For switches that use three digit trunk numbers, both these data items must be recorded to identify the actual trunk used for a call. With VFCDR, this field can be expanded to 3-digits to accommodate the full trunk number in one field. If there is no outgoing circuit ID associated with the call (for example, an incoming call), this data item is blank.

- ***Outgoing Circuit ID (100ths Digit) (Encode 17)***

  Beginning with System 85, R2V3, trunk numbers were increased from 2- to 3-digits. This data item was added to accommodate the additional digit. For switches that use three digit trunk numbers (System 85, R2V3 and later and Generic 2), both this data item and the previous 2-digit circuit ID data item (encode 16) must be recorded to identify the actual trunk used for a call. If there is no outgoing circuit ID associated with the call (for example, an incoming call), this data item is blank.

- ***Pattern Used (Encode 38)***

  This data item applies to Generic 2.2 only. It shows the WCR routing pattern used for an outgoing call.

- ***Precedence Level Digit (Encode 26)***

  The AUTOVON network is a government network that provides preemption capabilities for calls related to national defense. Five precedence levels are used to determine which calls have precedence over other calls in progress. Precedence level is recorded on outgoing calls only.

- ***Preference Used (Encode 39)***

  This data item applies to Generic 2.2 only. It shows the WCR routing preference used for an outgoing call.

- ***QDN/ VDN (Queue Directory Number/ Vector Directory Number) (Encode 32)***

  QDNs are used with ACD splits, and VDNS are used with Call Vectoring. The switch records either the QDN or the VDN based on the following criteria:

  — If ACD is being used without Call Vectoring, CDR records the QDN.

  — If Call Vectoring is used without an ACD environment, CDR records the VDN.

  — If Call Vectoring is used in an ACD environment, CDR records the VDN.

  If Call Vectoring is used with or without ACD, the VDN is recorded by CDR as the dialed number when the call is abandoned before an agent answers the call.

- ***Time-in-Queue (Encode 11)***

  This 2-digit data item indicates the time that the outgoing call has spent in a trunk group's queue. It represents minutes for ringback queue or tenths of minutes for off-hook queue. When ringback and off-hook queuing are both provided on the switch, the queuing type feature flag identifies the type of queuing used.

- ***Time-of-Day Control Mode (Encode 50)***

  The time-of-day plan control mode determines the way the switch routing time-of-day plan is changed. There are two basic modes of control:

  1. Automatic (System clock control) — When the system clock reaches an administered time, the time-of-day plan changes.

  2. Manual control — When the attendant or switch administrator takes specific action to change the time-of-day plan. The manual mode can be selected in one of three ways:

- Explicitly administered — The switch administrator sets the manual mode and time-of-day plan in administration.
- Clocked manual override — The switch administrator sets a start and end time for a specific time-of-day plan which overrides, but does not replace the automatic time-of-day change pattern.
- Dial access from attendant console — The attendant uses a DAC to cancel the current time-of-day plan and to replace it with one selected from the attendant console.

The records that contain control modes are special non-call records that are sent across the CDR ports after a control mode change or a time-of-day plan change.

a. A 0000 in the control mode data item of the CDR record indicates a time-of-day plan under automatic control by the system clock.

b. A 0001 indicates a time-of-day plan under manual control (attendant control).

c. A 0002 indicates a time-of-day plan under clocked manual override (administered override control).

It is possible to output the control mode data item in two different types of special call detail records. *See* the CDR Record Types section.

- ***Time-of-day Plan Set (Encode 51)***

  This 1-character data item appears in the non-call record when a time-of-day plan change is initiated. It appears in special reports dealing with the time-of-day plan used with the networking features ARS or WCR. This data item identifies the time-of-day plan in effect prior to and after a change.

  For System 85 and Generic 2.1, two data items, ARS Control Mode (encode 50) and Time-of-day Plan Set (encode 51) referred to the time-of-day functions of the ARS feature. Effective with Generic 2.2, all data items pertaining to ARS Plan have been changed to Time-of-Day plan. With the World Class Routing feature, the time-of-day plan is no longer limited to public network calls (ARS).

- ***Time-of-Day Hours (Encode 20)***

  This data item, is used in conjunction with time-of-day minutes (encode 21) to record the time that a call is disconnected.

  Before Release 2, Version 4, the time of day was inserted into the record by peripheral hardware. For R2V4 and Generic 2 switches, this is done by switch software using the 24-hour system. That is, 2:00 a.m. is recorded as 0200 and 2:00 p.m. is shown as 1400. For more information see the discussion on "PCC Time Stamping" in the "General Considerations" section.

- ***Time-of-Day Minutes (Encode 21)***

  This data item records the minute (during the hour recoded by encode 20) in which the call disconnected.

Before Release 2, Version 4, the time of day was inserted into the record by peripheral hardware. For R2V4 and Generic 2 switches, this is done by switch software using the 24-hour system. That is, 2:00 a.m. is recorded as 0200 and 2:00 p.m. is shown as 1400. For more information see the discussion on "PCC Time Stamping" in the "General Considerations" section.

- ***Trunk (or WCR Network) Access Code Dialed (Encode 5)***

  This can be a 4-digit data item for System 85, Release 2, Version 4 and DEFINITY Generic 2 switches. It is a 3-digit data item for System 85, Release 2, Version 3 and earlier switches. It is used only for outgoing or tandem calls, and can be a network DAC or the DAC of a specific trunk group. If this code is the same as the trunk-group access code used, the trunk-group access code used data item is blank.

- ***Trunk Access Code Used (Encode 6)***

  This can be a 4-digit data item for System 85, Release 2, Version 4 and DEFINTY Generic 2 switches. It is a three digit data item for System 85, Release 2, Version 3 and earlier switches. It is used only for outgoing or tandem calls, and contains the DAC for the trunk group that the call actually uses. If the access code dialed data item and the access code used data item have the same value, the DAC-used data item is blank.

- ***UUI (User-to-User Information) Counter (Encode 35)***

  This data item represents the count of UUI elements transmitted with an ISDN call. Message Associated (MA) UUI is an ISDN concept in which information can be passed between users, utilizing excess capity present on signaling links (D-channels). The public network is expected to charge for ISDN MA UUI service. Therefore, CDR will record this information in support of billing on ISDN calls. A counter is used to keep track of ISDN user-to-user data messages. Each time a user-to-user data message is sent, the MA UUI counter is incremented. The final count is provided in the CDR record.

  User-to-user data information can be contained in ISDN Call Setup, Alert, Connect, Disconnect, Release, or Release Complete messages. The following outgoing IEs (Information Elements) from codeset 0 and codeset 6 (or 7 depending on switch version) increment the MA UUI counter:

  — Codeset 0:

    High Layer Compatibility

    Calling Party Subaddress

    Connected Number

    Redirected Number

    User-to-User Information

    Locking Shift to Codeset 6 (or 7)

— Codeset 6 (or 7)

Traveling Class Mark

Display

AT&T Standard Facility (ASF)

- ***WCR Sent Number (Encode 37)***

  This data item applies to Generic 2.2 only. It shows the digits sent on a WCR call. If digit modification takes place after digit analysis (for example, digit modification due to preference selection), or if an IXC code is added, this data item will contain the modified number. See also, the "Dialed Number" data item.

CDR Record Generation

Call records are generated based on the seizure or attempted seizure of incoming and outgoing recordable trunks. Each time a call accesses a recordable trunk group, a CDR record is generated and stored in a buffer (switch memory). This record contains valuable information about the call.

When the call ends, the switch processor finishes collecting the call information or *"data items;"* and arranges them into a defined format. The defined format is either one of two default formats or an administratively defined format. Formatting ensures that the switch records and reports data items in a consistent order. This in turn, insures that the peripheral device can correctly identify the data items contained in the record. The CDR records can also be sent unformatted to custom developed equipment (not provided by AT&T). The various formats available are described in the "Record Formats" section.

The format used is determined to some extent by the vintage of switch used and the output port and peripheral configuration. Three different output port types are available as shown in Figure 27-1. Usually only one of the port types is used for a particular switch installation. The PCC port (used with the VFCDR configuration) offers the greatest flexibility of the three. The SMDR and CMDR/NCOSS ports are supported primarily for backward compatibility with older adjuncts.

CDR Ports and Data Presentations From the Switch

System 85, Release 2 and DEFINITY Generic 2 switches support three types of CDR ports:

- SMDR Port (TN403) available on all Release 2 versions
- CMDR/NCOSS Port (TN403) available on all Release 2 versions and Generic 2
- PCC Port (TN474B) available on R2V3 and later versions and Generic 2.

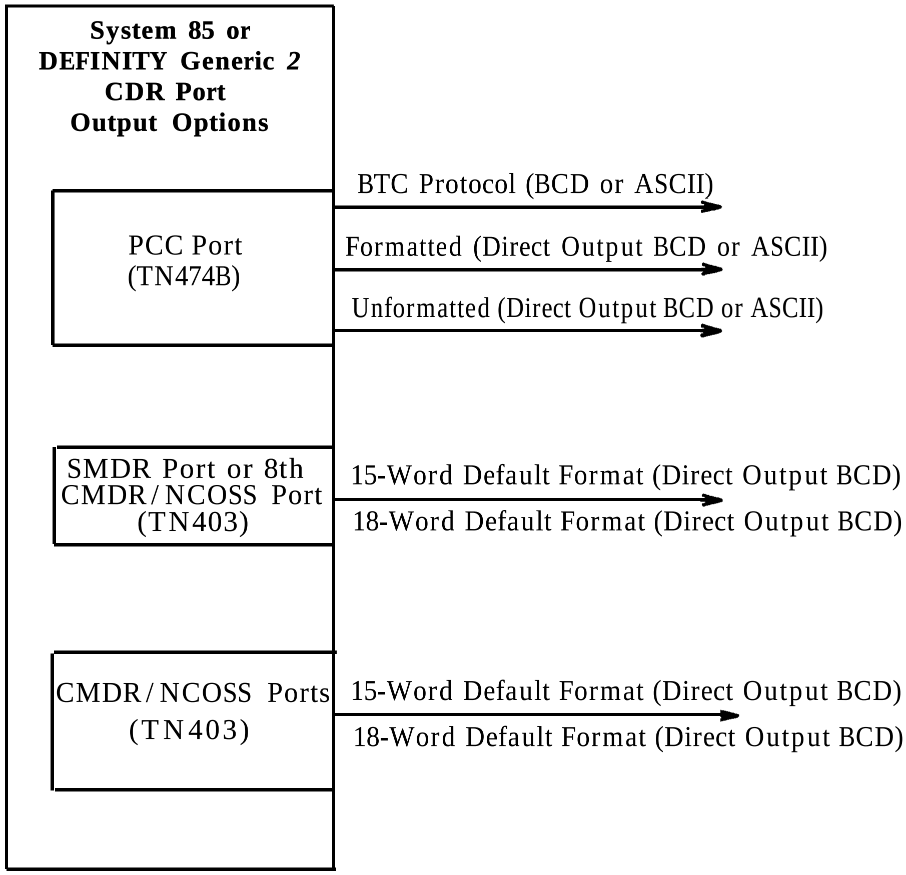


Figure 27-1. CDR Port Output Options

On any given switch, the call record is established in only one format. This same call record information is sent to all three port types (SMDR, CMDR/NCOSS, and PCC). The difference in port output is the number of words and the data format that the port and its associated peripheral devices can handle.

While the SMDR and the CMDR/NCOSS ports can receive VFCDR records from the switch, they may not correctly process these records. These ports and their associated adjuncts expect a certain format. The use of VFCDR may produce unexpected results if subsequent processing in not changed to handle a different format. The SMDR and CMDR/NCOSS ports can handle VFCDR records that are formatted the same as the 15- and 18-word default formats. However, they cannot handle VFCDR records that use a 24-word format or a 15- or 18-word format that differs from the default formats. That is, an SMDR port cannot handle a variable format record unless the first 15 data items of the variable format are the same as, in the same order as, and with the same length as in the 15-word default format. If the switch is administered to use a recommended standard format, the only port that can handle this data in a valid format is the PCC port.

CDR Record Types

This section describes the different types of CDR records available. These records are available to all CDR configurations.

Call Record

Call records are generated when a caller (or networking software) seizes a trunk to place a call. The record generated contains information based on the data items required by the format administered and the facilities used to place the call.

The CDR feature sends call records until a special event such as a time-or-day plan change or time-of-day control mode change occurs, a software reload occurs (due to a power failure or restart), or an attendant dials the SMDR-Stop DAC. When one of these special events occur, the switch sends a set of special records, the Reportable Trunk Group and Non-Reportable Trunk Group Records before resuming standard call records.

Ineffective Attempt Call Record

Incoming calls and outgoing AAR, ARS, or WCR calls that are not completed, can be administered to be recorded by CDR. These records are identified by a condition code (see the "Condition Codes" in the "Recordable Data Items" section for definitions). An ineffective attempt call record can result from any of the following conditions:

- All AAR, ARS, or WCR trunks are busy and no queuing exists, or all queue slots are busy.
- The call queues but never completes because the caller either cancels the call (hangs up), or the call remains in queue for the maximum time allowed and is released by the system.
- The FRL is not high enough to allow access to an available route in the routing pattern.
- The caller dials the toll-restricted access code, and only toll routes are available.
- An incoming call is placed to an unavailable (busy or unassigned) terminal.

The Switch Reload and Reportable Trunk Group Records

Usually the first record sent over the CDR ports is a System Reload Record. This record is automatically sent every time there is a power hit, switch failure, or software load. The switch sends a Condition Code Record, indicating that the following records are to be ignored until the Reportable Trunk Group Records appear. Upon reloading the software, the switch sends a set of Reportable Trunk Group Records.

For the default formats, the Reportable Trunk Group Records are sent automatically. That is, no special administration is required. However, for variable format, the DAC fields must be administered to a record format before the switch will send the Reportable Trunk Group Records. This record is administered by assigning the DACs (encodes 52 through 75) to specific record positions, just like the call record data items. This is done separately from the call record format and apprears to be overwriting the basic call record. The switch recognizes this DAC assignment as being specifically for the Reprotable Trunk

Group Records and does not confuse this format with the call record format. Unless the DAC fields are specifically assigned to a record position using Procedure 288, Word 2, the Reportable Trunk Group Record is not generated.

Reportable and non-reportable trunk group records (condition code 2 and 3 records) are sent by the switch to the CDR ports at the following times:

- When CDR is activated or deactivated by the attendant dialing the SMDR-Start or SMDR-Stop dial access code
- When the switch's software is reloaded
- When a time-of-day plan change or time-of-day control mode change occurs.

All trunk groups reporting to CDR must be assigned a DAC. Encode 52 (1st dial access code of the CDR record) must be administered in the CDR format for the trunk-group records to be sent.

Reportable Trunk Group Record

The first record sent by the switch is a Condition Code 2 (or B) Record. The "2" means that the following list of records with Condition Code 2 consist of DACs assigned to trunk groups that are reporting (or providing call data) to CDR (up to 999 trunk groups in DEFINITY Generic 2).

Non-Reportable Trunk Group Record

Following the Condition Code 2 records is a list of Condition Code 3 (or C) records. These records contain the DACs of all the unmonitored trunk groups.

> **NOTE:** CDR will send as many records as necessary to output all the reportable and non-reportable trunk groups with DACs assigned.

Time-of-Day Plan Control Mode Record

When a control mode change is administered, either one or two special records are sent across the PCC. The first record contains the status of the Condition Code ("D" if administered) and the new status of the control mode after the change was made. A second record is sent only if the change in control mode caused a time-or-day plan change.

If the time-of-day plan changes due to the control mode change, a second record is sent that contains the condition code, control mode, and the time-of-day plan in effect.

After the time-of-day plan control mode record, a set of reportable and non-reportable trunk group records (condition codes 2 and 3) is sent.

> **NOTE:** Even though the it is administered as part of the variable format, the control mode data item is sent to the CDR port only when the control mode or time-of-day plan has changed.

Time-of-Day Plan Change Record

When a time-of-day plan change is initiated (by the attendant or automatically), a special record is sent across the PCC. This record contains the status of the changed-to time-of-day plan, condition code ("D" if administered) and control mode (if the control mode data item is administered).

After the time-of-day plan change record, a set of reportable and non-reportable trunk group records (condition codes 2 and 3) is sent.

> **NOTE:** Even though the time-of-day plan change data item can be administered as part of the call record format, the time-of-day plan change data item and control mode data items are recorded only when the control mode or time-of-day plan is changed and only in the special non-call related records.

Long Call Duration Record

The long call duration record identifies extremely long duration calls (10 hours or longer). On such calls, the switch produces a call record with Condition Code 4 (if Condition Codes are administered), and a duration entry of 9 hours, 59 minutes and 9 tenths of a minute after the first 10-hour period. The switch produces another call record for the same call after each succeeding 10-hour period. When the call does end, the switch produces a final call record for the same call, with a different condition code identifying the call type.

Record Formats (Default, Recommended Standard, and Variable)

Call information (data items) is collected by the switch and stored in a buffer. Through administration, the layout of data items for all the CDR records on the switch is organized into a specific format. Formats are usually described in terms of ***words*** (for example, 15-word default format or 18-word recommended standard format). It is important to note the difference between a ***word*** and a data item. Within the context of CDR, a word is 16 binary bits, broken down into four 4-bit cells or nibbles. A data item, while assigned to specific positions within the word structure, is not specifically associated with a word. For example, an opcode is a data item that occupies 4 bits or 1 cell, while an account code is a data item that can consist of from 5 to 15 digits, with each digit requiring 4 bits or 1 cell. As a result, there is no direct relationship between the number of words in a format and the number of data items that format can contain. Two default formats and a variable format are available. The variable format provides several options in available word structures depending on the adjuncts and peripheral devices used.

The available formats and the data items each can use, are shown in Table 27-C. Table 27-D shows the compatibility between the available formats and standard peripheral equipment.

TABLE 27-C. Data Items Available for Default, Standard and Variable Formats

		FORMATS				
Encode	**CDR Data Item**	**15-Word Default**	**18-Word Default**	**18-word ISDN**	**Standard 24-word**	**Variable Format**
1	Call Duration Hours	x	x	x	x	x
2	Call Duration Minutes	x	x	x	x	x
3	Call Duration 10ths/Minute	x	x	x	x	x
4	Condition Code	x	x	x	x	x
5	Trunk Access Code Dialed	x	x	x	x	x
6	Trunk Access Code Used	x	x	x	x	x
7	Dialed Number/Routed Number	x	x	x	x	x
8	Calling Number	*	*	*	x	x
9	Account Code	x	x	x	x	x
10	Authorization Code	†	†	x	x	x
11	Time in Queue	†	†		x	x
12	Facilities Restriction Level	†	†	x	x	x
13	10,000s Digit of Calling Number		x	x	x	x
14	Incoming Circuit ID		x	x	x	x
15	Feature Flags		x	x	x	x
16	Outgoing Circuit ID		x	x	x	x
17	Outgoing Circuit ID (100s digit)		x	x	x	x
18	Incoming Circuit ID (100s digit)		x	x	x	x
19	Interexchange Carrier Code		x	x	x	x
20	Time of Day, Hours					x
21	Time of Day, Minutes					x
22	Date, Month					x
23	Date, Day					x
24	Date, Year					x
25	Incoming Trunk Dial Access Code*	*	*	*	x	x
26	Precedence Level Digit				x	x
27	Attendant Console Number				x	x
28	ISDN Network Service Value			x	x	x
29	Extension Partition Number					x
30	Node Number				x	x
31	Bearer Capability Class				x	x
32	QDN/VDN					x
33	Agent Login					x
34	ISDN Cause Value					x
35	UUI Counter				x	x
37	WCR Number Sent					x
38	WCR Pattern					x
39	WCR Preference					x
50	ARS/WCR Control Mode				x	x
51	ARS/WCR Time-of-Day Plan Set					x
52-59	Dial Access Codes 1-8	x	x			x
60-75	Dial Access Codes 9-24					x

* For all default 15- and 18-word record formats, the Incoming Trunk Dial Access Code is located in the same field as the Calling Number.

† The default 15- and 18-word formats may or may not have the Authorization Code, Time-in-Queue, and FRL fields, depending on the length of the Account Code.

TABLE 27-D. CDR Formats and Compatible Peripherals

	CDR Device *(See* **Note)**						
Format	Manager IV	93B	NCOSS	TELESEER SMDR	CAS	CDRU & CDRP	94A LSU
15-word default	X	X	X	X	X	X	X
18-word default	X	X	X	X	X	X	X
18-word expanded*	X	X	X	X	X	X	X
18-word ISDN*	X	X	X			X	X
24-word standard*	X		X			X	
Variable Format*						X	
NOTE: An "X" in a box means the **CDR Device** and the **Format** are compatible.							

* Implemented using VFCDR.

Default Formats

Two predefined or default formats are available:

- The 15-word default format is available on all System 85 and DEFINITY Generic 2 switch over the SMDR, CMDR, or PCC ports.
- The 18-word default format is available on System 85, Release 2, Version 1 and later and on DEFINITY Generic 2 switches over the CMDR/NCOSS, or PCC ports.

Opcodes

Opcodes are a standard part of both the 15- and 18-word default formats. They occupy the first cell of each word in these formats.

Opcodes are used for hardware control and on-line error checking (identify the end of a record). Opcodes are sent to the storage devices, such as the SMDR Adjunct or LSU (Local Storage Unit). However, opcodes are not included in the output received by the records collection devices such as the 93B Polling Device.

Account Code of More Than Five Digits with Default Formats

Either the traditional 1- to 5-digit or an expanded (6- to 15-digit) Account Code can be used on System 85, Release 2, Version 3 or later, and on DEFINITY Generic 2 switches. When the 15- or 18-word default format is used with expanded account codes, the subsequent data fields [authorization code, time-in-queue, and FRL (Facilities Restriction Level)] will not be formatted properly in the CDR record. The data fields that are disrupted depend on the length of the account code used.

The default formats assume the account code will be 5-digits or less. When an account code longer than 5-digits is used, the account code (which is left justified), must be administered to overwrite the following data fields:

- If the account code is between 6- and 12-digits, the Authorization Code field is overwritten (in part or as a whole).
- If the account code is between 13- and 14-digits, the Time-in-Queue field is also overwritten.
- If the account code is 15-digits, the FRL field is also overwritten.

Variable or Customized Formats

VFCDR (Variable Format Call Detail Recording) was introduced in System 85, Release 2 Version 4 and is available only over the PCC port. VFCDR provides the ability to customize CDR record reports to meet the need for additional or different data items and new services. VFCDR allows customers to define record contents by choosing from a set of 75 available data items. The data items can be arranged in any order and combination, to provide record-formatting flexibility. Table 27-C lists the data items available for use with the VFCDR configuration. See the VFCDR (Variable Format Call Detail Recording) section for more details.

Account Codes of More than 5-Digits with Variable Format CDR

When a custom format is administered, there may be room in the record for additional data elements. These could include the following:

- Expanded account code (up to 15-digits)
- Authorization code
- Time-in-Queue
- FRL (Facility Restriction Level).

The customer is responsible for administering the CDR format to prevent overlap of these fields. This is one example of the crucial nature of planning the record layout *(See* **Feature Administration).**

Recommended Standard Formats

While formats used with the VFCDR configuration can be customized to meet customer needs, specific *recommended standard* formats have been developed. The primary advantage of using these recommended standard formats as opposed to a customized format is that a wider variety of peripheral devices can accept output when the recommended standard formats are used (see Table 27–D).

Like customized formats, the recommended standard formats must be specifically administered using Procedure 288, Word 2. The following are recommended standard formats for existing AT&T peripherals that can be obtained through variable format administration:

- Recommended Standard 18-word ISDN
- Recommended Standard 24-ISDN
- Recommended Standard 24-word ISDN.

SMDR (Station Message Detail Recording)

Description

The SMDR (Station Message Detail Recording) configuration is the original, and simplest CDR output configuration. With this configuration, 15-word default format (direct output BCD code) records are sent through the SMDR port to an SMDR adjunct. The SMDR adjunct provides minimal record storage (temporary) and processing before sending call records to other devices for long term storage and output.

Later CDR enhancements, specifically CMDR (Centralized Message Detail Recording) and VFCDR (Variable Format Call Detail Recording), surpass the capabilities of SMDR. Both of these enhanced configurations are described in their own sections later in this chapter.

The SMDR port (used with the SMDR Configuration) is one of the three different port type shown in Figure 27-1. The SMDR configuration with possible adjuncts and peripherals for the SMDR port is shown in Figure 27-2.

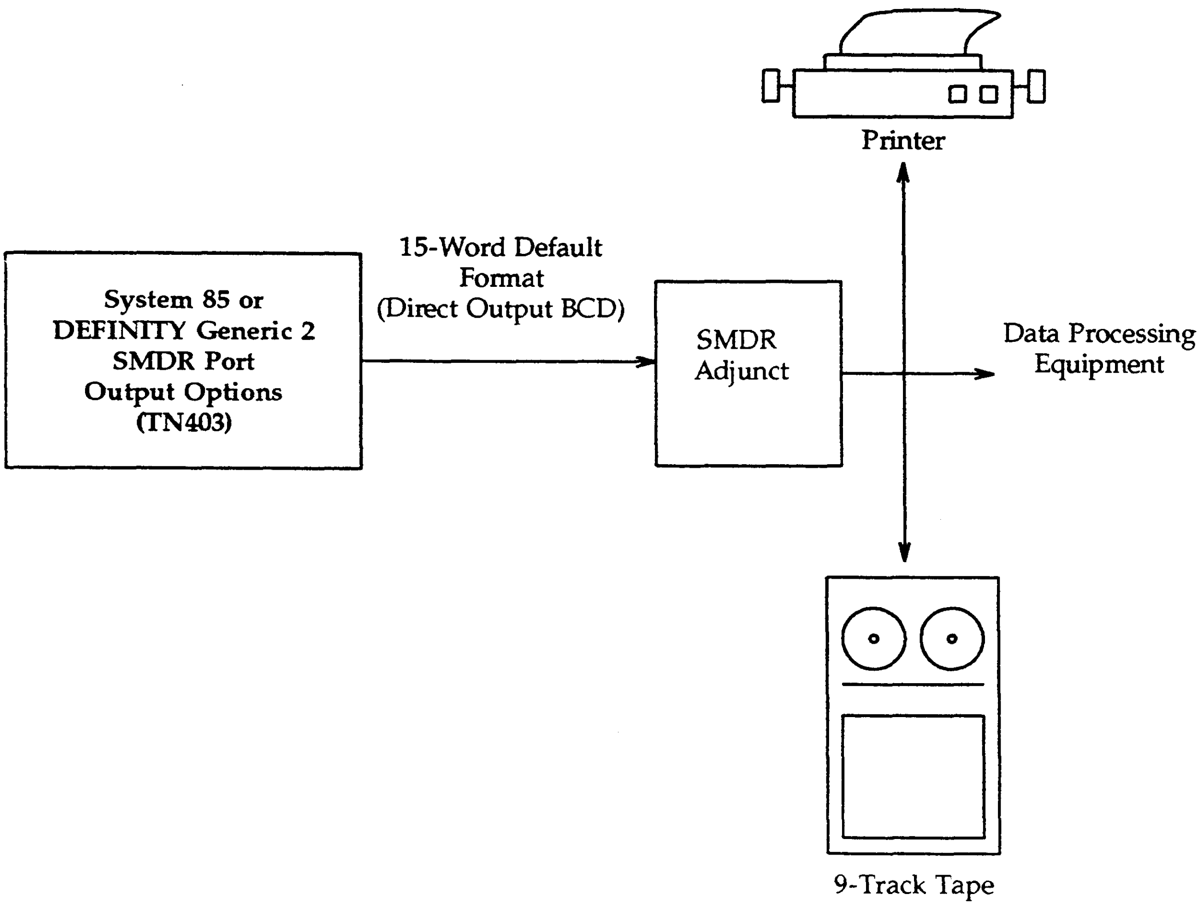


Figure 27-2. SMDR Port Output Options

Figure 27-3 is an example of call information from SMDR that has been processed for easy readability. With the SMDR configuration, direct output BCD (15-word default format) records are sent from the switch without processing. Any processing of this data must be done by peripheral equipment. Figure 27-3 shows calls in chronological order that have been formatted by a peripheral device.gp The form and order can be changed by the peripheral equipmentgp to meet the user's needs. See the notes on the following page for call type descriptions associated with each record in this table.

Date: 5/5/87

Time of Day	Duration	Cond Code 9-Trk	Trunk Group Dialed	Trunk Group Used	Dialed Number	Calling Number	Account Code	Authorization Code	Time in Queue	FRL	See Notes*
10:43	0:09.9		9		1-317-265-7894	46113					1
10:44	0:05.1		81	82	1-317-357-7192	46181					2
10:44	0:11.5	1	82		303-451-1857	46374	16113				3
10:54	0:15.6		9		765-5751	43711					4
10:58	0:05.3		83		319	46113					5
11:05	0:02.4	9			44259	7					6
11:15	0:14.7	9	9		1-317-357-7192	7					7
11:30	0:06.5	9	9		1-213-357-1212	44456			01		8
12:11	959.9	4	9		1-303-452-3057	356	152				9
15:09		2	81	83	7						10
15:09		3	9	41 57 58 59	73	74					11
09:03	0:03.2	7	9	83	1-303-424-6112	43634	23541	7365810		2	12
09:30	0:16.2	C	9		723-1401						13
14:22	0:18.1	7	8	734	321-7866	44638	58732	3347721		1	14
16:30	0:24.5	7	8	735	638-4034	18				4	15
11:38		D				40000				2	16
09:51	0:08.2	9			3222	16	303				17
09:01	0:06.8	7	8	736	536-2487		201	36852		2	18
13:19	0:14.3	7	8		434-3226	732	315	52168		1	19
12:08	0:09.5	1	8	737	776-8621	733	212	34523		2	20
14:26	0:00.1	E	8		341-7894	44864	36185	28734		3	21
15:03	0:00.1	F	8		1-315-342-2368	43715		46213		1	22
15:04	0:04.5	9	2		2015	7					23
15:04	0:10.5		9		415-4196	42015					23
15:10	3:00.0	4	9		1-303-452-3057	43561	152				24
15:12	0:00.1		9		451-1419	42016					25
15:14	0:05.4	C			2015	73					26
15:14	0:04.9	C	9		451-4110	73					26
15:20	2:17.0		2		345	43222					27
15:22	2:54.1		71		2345	43222					28

Figure 27-3. Example of Custom Formatted SMDR Call Information Report

* See Notes on following page.

Notes:

1. Outgoing call over long distance
2. Outgoing call with Route Advance over long distance
3. Outgoing call placed by attendant and charged to account code
4. Local outgoing terminal to terminal call
5. Outgoing tie trunk call requiring 3-digit dialing of terminal of distant end
6. Incoming call completed to terminal
7. Incoming call completed to outgoing trunk
8. Outgoing call served by queuing feature
9. Long duration outgoing call (see Note 24)
10. Begin or continue monitoring of these trunk groups
11. Discontinue monitoring of these trunk groups
12. Call Record example of outgoing trunk call with AAR or WCR selection
13. Call record example of attendant conference call
14. Call record example of outgoing trunk call with AAR or WCR
15. Call record example of incoming trunk completed to outgoing trunk with AAR or WCR
16. Call Record example of Automatic Route Selection pattern change to pattern 2
17. Incoming remote access call to terminal
18. Incoming remote access call completed to outgoing trunk with AAR or WCR
19. Incoming call on WATS trunk to attendant and extended to the terminal with AAR or WCR
20. Incoming call on WATS trunk to attendant and completed to outgoing trunk with AAR or WCR
21. Ineffective attempt of outgoing call
22. Ineffective attempt of outgoing call
23. Incoming call to extension 2015, extension 2015 flashes and adds CO Trunk 451-4196 on 3-way conference
24. Second part of long duration call (see Note 9)
25. Partially dialed outgoing call from extension 2016
26. Incoming call to attendant requesting attendant conference
27. Main/satellite call (without multidigit steering)
28. Main Satellite call (with multidigit steering)

SMDR Data Items

Call records reported over the SMDR configurationgp must be arranged in the 15-word default format shown in Figure 27-4. The following list shows the set of data items used with the 15-word default format. These data items are arranged in only one standard order and length that is automatically set up by the switch.

Call Duration Hours (Encode 1)

Call Duration Minutes (Encode 2)

Call Duration 10ths / Minute (Encode 3)

Condition Code (Encode 4)

Access Code Dialed (Trunk-Group) (Encode 5)

Access Code Used (Trunk-Group) (Encode 6)

Dialed Number (Encode 7)

Calling Number* (Encode 8)

Account Code (limited to 5 digits)* (Encode 9)

Authorization Code (Encode 10)

Time in Queue (Encode 11)

Facilities Restriction Level (Encode 12).

SMDR Record Format

The SMDR Record Format shown in Figure 27-4 is the 15-word default format. Any other format sent to the SMDR adjunct will not be interpreted correctly.

SMDR Port

The SMDR configuration has been in use since the DIMENSION PBX. SMDR can be used on any System 85 or DEFINITY Generic 2 switch. The SMDR port (TN403) can only handle the 15- and 18-word default formats.

The 18-word default format is used if the SMDR port is administered as an 8th CMDR port. The adjuncts that can be connectd to the SMDR port are not designed for high traffic-volume switches.

---

* Data entry in CDR record is left justified

Word	Bit 15–12	Bit 11–08	Bit 07–04	Bit 03–00
01	Opcode	Call Duration Hours	Call Duration Minute (Xx)	Call Duration Minute (xX)
02	Opcode	10th Minute	Condition Code	Access Code Dialed 1
03	Opcode	Access Code Dialed 2	Access Code Dialed 3	Access Code Used 1
04	Opcode	Access Code Used 2	Access Code Used 3	Dialed Number 1
05	Opcode	Dialed Number 2	Dialed Number 3	Dialed Number 4
06	Opcode	Dialed Number 5	Dialed Number 6	Dialed Number 7
07	Opcode	Dialed Number 8	Dialed Number 9	Dialed Number 10
08	Opcode	Dialed Number 11	Dialed Number 12	Dialed Number 13
09	Opcode	Dialed Number 14	Dialed Number 15	Calling Number 1
10	Opcode	Calling Number 2	Calling Number 3	Calling Number 4
11	Opcode	Account Code 1	Account Code 2	Account Code 3
12	Opcode	Account Code 4	Account Code 5	Authorization Code 1
13	Opcode	Authorization Code 2	Authorization Code 3	Authorization Code 4
14	Opcode	Authorization Code 5	Authorization Code 6	Authorization Code 7
15	Opcode	Time in Queue 1	Time in Queue 2	FRL

Words 11–15: Variable-Length Account Code†

* The Account Code can vary in length from 1 to 15 digits. The length is administered in Procedure 275, Word 1. If the Account Code exceeds five digits, the Authorization Code is lost. If it exceeds 12 digits, the Time-in-Queue data is lost If it is 15 digits long, the FRL data is lost.

Figure 27-4. 15-Word Default Format

SMDR Adjuncts

Direct Output SMDR Adjunct:

The SMDR Direct Output adjunct is connected directly to the SMDR port. The switch outputs BCD (Binary Coded Decimal) to the SMDR adjunct. The adjunct converts this output to ASCII (American Standard Code for information exchange) which it outputs to a peripheral printer. The capacity of the buffer allows 16 call records to be stored. If 16 call records are stored and another call comes in, the incoming call causes a memory overflow and forces the SMDR to overwrite the oldest call record in the buffer. Call records are sent out on a FIFO (first-in first-out) basis.

9-Track Tape Unit:

The 9-track tape unit provides a storage medium for call records that can be used when the output rate is higher than the direct output configuration can handle.

- Memory Capacity
  - The 9-track unit itself has buffering capacity that can store call data for about 31 calls.
  - A single 9-track tape can store call data information for about 330,000 calls.
- Reports From 9-Track Tape

  When the SMDR application user desires a printout of call activity, the 9-track magnetic tape requires off-line processing (software decoding). This must be provided by the user. The output from the magnetic tape can be grouped according to user specifications. For example, output can be grouped by type of call, account number, or specific date.
- Tape Drive Maintenance

  The tape must be temporarily removed or changed once a month to allow maintenance of the tape drive.

  Call records cannot be written while the tape is being changed. Tapes should be changed during periods of low traffic or alternate tape drives provided for use during down time for maintenance.
- Tape Specifications

  High-quality magnetic tape is strongly recommended for use with the 9-track system. To assure reliability, the customer should use only tape that is full-width certified, and permanently error-free at 3200 flux changes per inch. This performance should be reached with the tape output voltage level at least 35 percent of the tape saturation level. The tape manufacturer should guarantee this performance level at final inspection.

TELESEER SMDR

The TELESEER SMDR is an adjunct that receives, stores and processes SMDR output. The TELESEER SMDR can store as many as 28,000 call records, 500 extension numbers, and 2,000 account codes. TELESEER SMDR can generate the following reports:

- Summary Report — This report provides a condensed listing of the number, duration, and cost of outgoing calls.
- Account Code Report — This report provides information on any call record that contains an account code. This information can be used to track calls from specific terminal users, cost allocation, or terminal user billing.
- Selection Report — This report allowsgp the system manager to specify information to be printed in the report. Date, time duration, and cost of each call can be specified by the manager.

For more information on TELESEER SMDR System, refer to the ***TELESEER SMDR Reference Manual*** (999-302-175)

COMM-Stor* II Communications Storage Unit

The COMM-Stor II is an adjunct that receives, stores and processes SMDR output. The COMM-Stor II can store between 4,000 and 10,000 call records depending on its configuration. The COMM-Stor II can be programmed to provide a variety of customized reports and output them to a standard printing device to meet the needs of users.

* Registered trademark of Sykes Datatronis, Inc.

Notes:

CMDR (Centralized Message Detail Recording)

Description

This option provides a means of collecting and reporting detailed call information on selected trunk groups for one or more switches.

CMDR collects the CDR records over the CMDR / NCOSS ports (1 through 7). The SMDR (Station Message Detail Recording) port can be used as an eighth CMDR port. With the CMDR configuration, call records from each port are collected in a 94A LSU (Local Storage Unit). Refer to Figure 27-5.

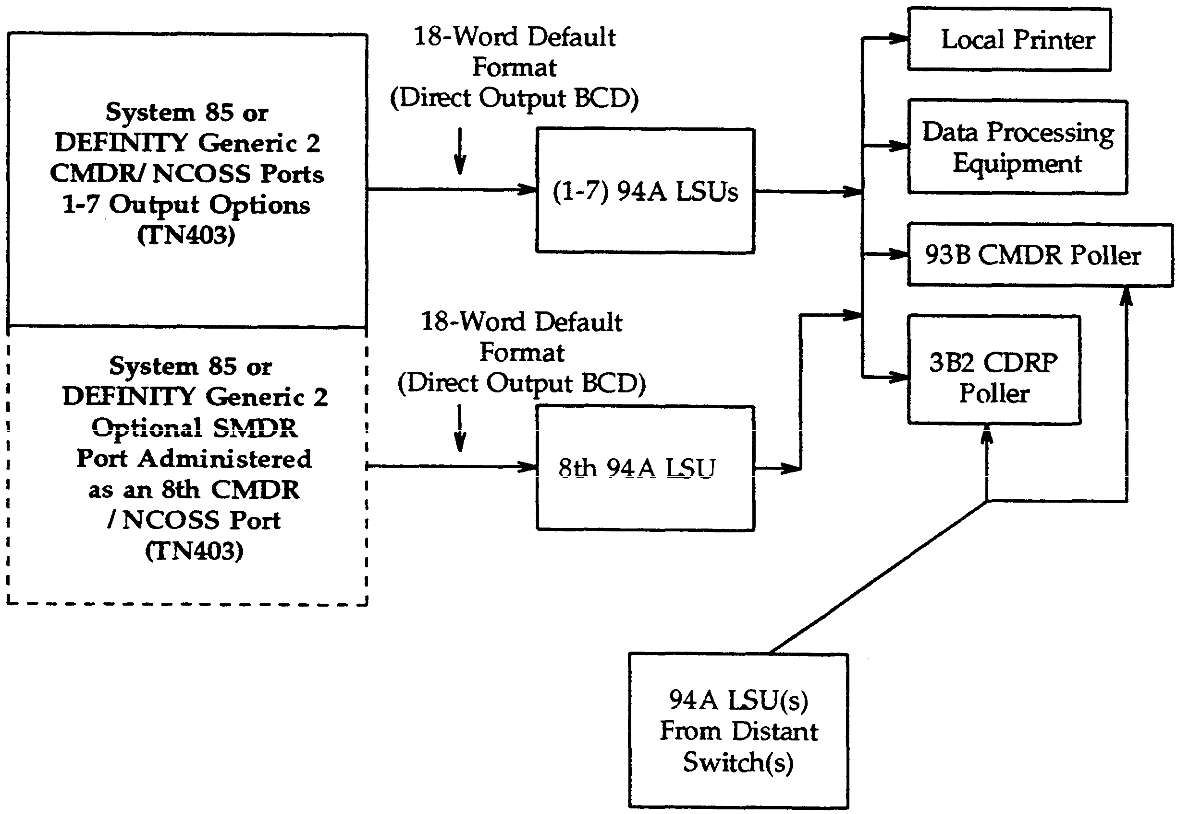


Figure 27-5. CMDR Port Output Options

Since the development of CMDR and SMDR (Station Message Detail Recording) another configuration for recording and reporting called VFCDR (Variable Format Call Detail Recording) has been provided for System 85 and DEFINITY Generic 2. VFCDR is discussed later in this chapter.

Data Items for CMDR

The following list is the set of data items used for the 18-word default format. These data items are selected automatically, along with 18-word default format, when the CMDR configuration is specified.

Call Duration Hours (Encode 1)

Call Duration Minutes (Encode 2)

Call Duration 10ths / Minute (Encode 3)

Condition Code (Encode 4)

Trunk Access Code Dialed (Encode 5)

Trunk Access Code Used (Encode 6)

Dialed Number (Encode 7)

Calling Number* (Encode 8)

Account Code* (5-digits) (Encode 9)

Authorization Code (Encode 10)

Time in Queue (Encode 11)

Facilities Restriction Level (Encode 12)

10,000s (5th) Digit of Calling Extension (Encode 13)

Incoming Circuit ID (Encode 14)

Feature Flags (Encode 15)

Outgoing Circuit ID (Encode 16)

IXC Code (Encode 19)

CMDR Record Format

The data items from a CDR record are sent across the CMDR / NCOSS Port in the 18-word default format. This format is format shown in Figure 27-6).

* Data entry in CDR record is left justified.

Word	Bit 15–12	Bit 11–08	Bit 07–04	Bit 03–00
01	Opcode	Call Duration: Hours	Call Duration: Minute (Xx)	Call Duration: Minute (xX)
02	Opcode	10th Minute	Condition Code	Access Code Dialed 1
03	Opcode	Access Code Dialed 2	Access Code Dialed 3	Access Code Used 1
04	Opcode	Access Code Used 2	Access Code Used 3	Dialed Number 1
05	Opcode	Dialed Number 2	Dialed Number 3	Dialed Number 4
06	Opcode	Dialed Number 5	Dialed Number 6	Dialed Number 7
07	Opcode	Dialed Number 8	Dialed Number 9	Dialed Number 10
08	Opcode	Dialed Number 11	Dialed Number 12	Dialed Number 13
09	Opcode	Dialed Number 14	Dialed Number 15	Calling Number 1
10	Opcode	Calling Number 2	Calling Number 3	Calling Number 4
11	Opcode	Account Code 1	Account Code 2	Account Code 3
12	Opcode	Account Code 4	Account Code 5	Authorization Code 1
13	Opcode	Authorization Code 2	Authorization Code 3	Authorization Code 4
14	Opcode	Authorization Code 5	Authorization Code 6	Authorization Code 7
15	Opcode	Time in Queue 1	Time in Queue 2	FRL
16	Opcode	Calling Number 5	Incoming Circuit ID 1	Incoming Circuit ID 2
17	Opcode	Feature Flags*	Outgoing Circuit ID 1	Outgoing Circuit ID 2
18	Opcode	Out Circuit ID 3	In Circuit ID 3	IXC Code

Words 11–15: Variable-Length Account Code †

* Feature Flags: This field contains a single hexadecimal character that indictes the status of specific features. For an explanation of these hexidecimal codes see Table 28-B.

† The account code can vary in length from 1 to 15 digits. The length is administered in Procedure 275, Word 1. If the account code exceeds five digits, the authorization code is lost. If the account code exceeds 12 digits, the time-in-queue data is lost. If the account code is 15 digits long, the FRL is lost.

Figure 27-6. 18-Word Default Format

CMDR Word Structure

Each CDR word consists of four 4-bit nibbles, or "cells", (16 bits to each word). The cells that contain the Condition Code, IXC Code, and Feature Flag can contain hexadecimal characters. All other cells contain binary coded decimal characters. With the CMDR configuration, the first cell of each word, contains an operation code (opcode).

CMDR/NCOSS (Network Control Operations Support System) Ports

When the CMDR confirmation is used, CDR records are sent out over seven ports (eight if the SMDR port is used), to 94A LSUs (Local Storage Units). Each LSU receives records in a "round robin" pattern. This allows the LSUs collectively, enough time to process the high speed output from the switch.

An external polling device such as the 93B CMDR or the 3B2 CDRP is used to collect and process records from LSUs on a periodic basis. The CMDR/NCOSS ports support both the 93B poller option and the network monitoring option NCOSS (Network Control Operations Support System).

With the CMDR configuration the SMDR direct output port can be administered as an eighth CMDR/NCOSS port. This is done by assigning unit type 14 to slot 23, circuit 15 in Procedure 253. When the SMDR port is administered as a CMDR port, 18-word records are sent over the SMDR port to the attached LSU.

CMDR Storage and Polling Units

The CMDR configuration consists of from one to eight LSUs and one (or more) polling device that collects CDR records. The polling device(s) can be centrally located. In a network arrangement, a central location gives CDR record uses all the CDR information about their network at a single site.

94A LSU (Local Storage Unit)

The 94A LSU is a call record storage and processing adjunct. A single 94A LSU can receive up to 14,600 CDR records per hour, store up to 64,000 records, and transmit up to 5,700 call records per hour over one or two 1200-bps asynchronous data links. Data integrity between the 94A LSU and the polling device is assured by using a data protocol that checks for errors in blocks of data.

The 94A LSU has a direct-output port that feeds into AT&T processing systems (such as Manager IV, TELESEER SMDR, CAS or other customer-provided equipment). The 94A also has two NCOSS/LSU ports. NCOSS can collect the CDR records by polling each LSU over these NCOSS/ISU ports on a regular basis. The NCOSS/ISU ports can be polled simultaneously by NCOSS and by a polling device such as the 93B poller.

Mulitple LSUs can be used beginning with System 85, Release 2, Version 3. With multiple LSUs, each LSU receives a portion of the CDR records distributed by the switch. This compensates for relatively slow polling devices in high volume applications. Multiple LSUs also provide at least partial CDR retrieval in the event of a LSU failure.

The 94A LSU supports the default 15- and 18-word, and 18-word ISDN record formats, but not the 24-word or variable format.

93B CDRP (Call Detail Record Poller)

The 93B CDRP is an external polling device used with the CMDR configuration. The 93B CDRP connects to a 94A LSU through a low speed modem (212A or equivalent) over the DDD (Direct Distance Dialing) network. For customers with multiple locations, the 94A LSUs in each location can be polled by a centralized 93B.

A single 93B can poll up to 31 LSUs but only 3 LSUs can be polled at one time. The 93B can collect up to 27,000 call records per hour at a rate of 9,000 call records per hour per LSU (in the maximum configuration).

There are two versions of the 93B. The first version, called CMDR-TO (Centralized Message Detail Recorder — Tape Output), provides CDR records to be stored on magnetic tapes. The second version, called CMDR-DO (Centralized Message Detail Recorder — Direct Output), provides CDR records directly to a customer host or to Manager IV.

Polling arrangements used for centralized data collection and reporting must be based on customer needs. The 93B CDRP is a standard system, that is compatible with the 94A LSU, System 85, and DEFINITY Generic 2; however, other systems could be used for this purpose.

3B2 CDRP (Call Detail Record Poller)

The 3B2 CDRP is an AT&T application software package that runs on AT&T 3B2 computers. The 3B2 CDRP provides an alternative polling arrangement, capable of polling several 94A LSUs. The 3B2 CDRP acquires, organizes, and stores call records sent from many record collection systems. It then transmits these records to a host processor or tape storage device. The CDRP has the following capabilities:

- Can collect 24,000 18-word call records per hour over a 9600 bps direct output ASCII link
- Supports BCD, ASCII, and EBCDIC character sets
- Can poll remote locations at data rate from 1200 to 9600 bps
- Services up to 128 total input devices
- Capable of simultaneously polling seven CDRUs (Call Detail Recording Units)*

* The various CDRUs available are described later under the VFCDR configuration.

- Supports 15-, 18-, or 24-word formats
- Maintains a complete polling and transmission history
- Is compatible with asynchronous and SCSI (Small Computer System Interface) tape output devices
- Allows remote maintenance and administration of input devices.

For more information see the ***Call Detail Record Poller Manual*** (555-006-213).

VFCDR (Variable Format Call Detail Recording)

Description

The VFCDR configuration is a major enhancement over the SMDR and CMDR configurations. The VFCDR configuration sends custom formatted CDR records over the PCC (Processor Communications Circuit) to a local storage unit or CDR record processing device (typically a CDRU).

VFCDR has the following advantages over SMDR and CMDR:

- VFCDR does not require opcodes.
- VFCDR can record many data items not recordable by SMDR or CMDR.
- VFCDR can record any data item in the "Recordable Data Items" list.
- VFCDR data items (call information) can be recorded in any order.
- VFCDR can use any default or standard format, or a custom format up to 24 words long.
- VFCDR uses the error checking protocol provided by the PCC.

Table 27-C (shown earlier), compares the data items available using VFCDR with the SMDR or CMDR configurations. Note that the "recommended standard" formats also require format administration like a custom format in VFCDR.

Using VFCDR administration, data items can be assigned to the call record in any order or combination using from 1 to 24 words. With VFCDR, the call record format is administered using Procedure 288, Words 1 and 2.

See the "Recordable Data Items" section for a detailed description of specific data items.

The PCC board provides three basic output options as shown in Figure 27-7. While any of these three options is available to a switch with VFCDR, only one of these options can be used at a time. Option 1 uses the full capabilities of VFCDR and is considered the basic option for VFCDR. Option 2 is basically a backward compatibility option that allows customers who are upgrading from a CMDR (Centralized Message Detail Recording) configuration to continue to use their on-hand peripheral equipment, connected to 94A LSU direct output ports. Option 3 is provided for those customer who wish to use non-AT&T peripheral equipment to store and process their CDR records. Once the CDR records are sent across the PCC port, the peripheral equipment receives the records for storage and further processing.

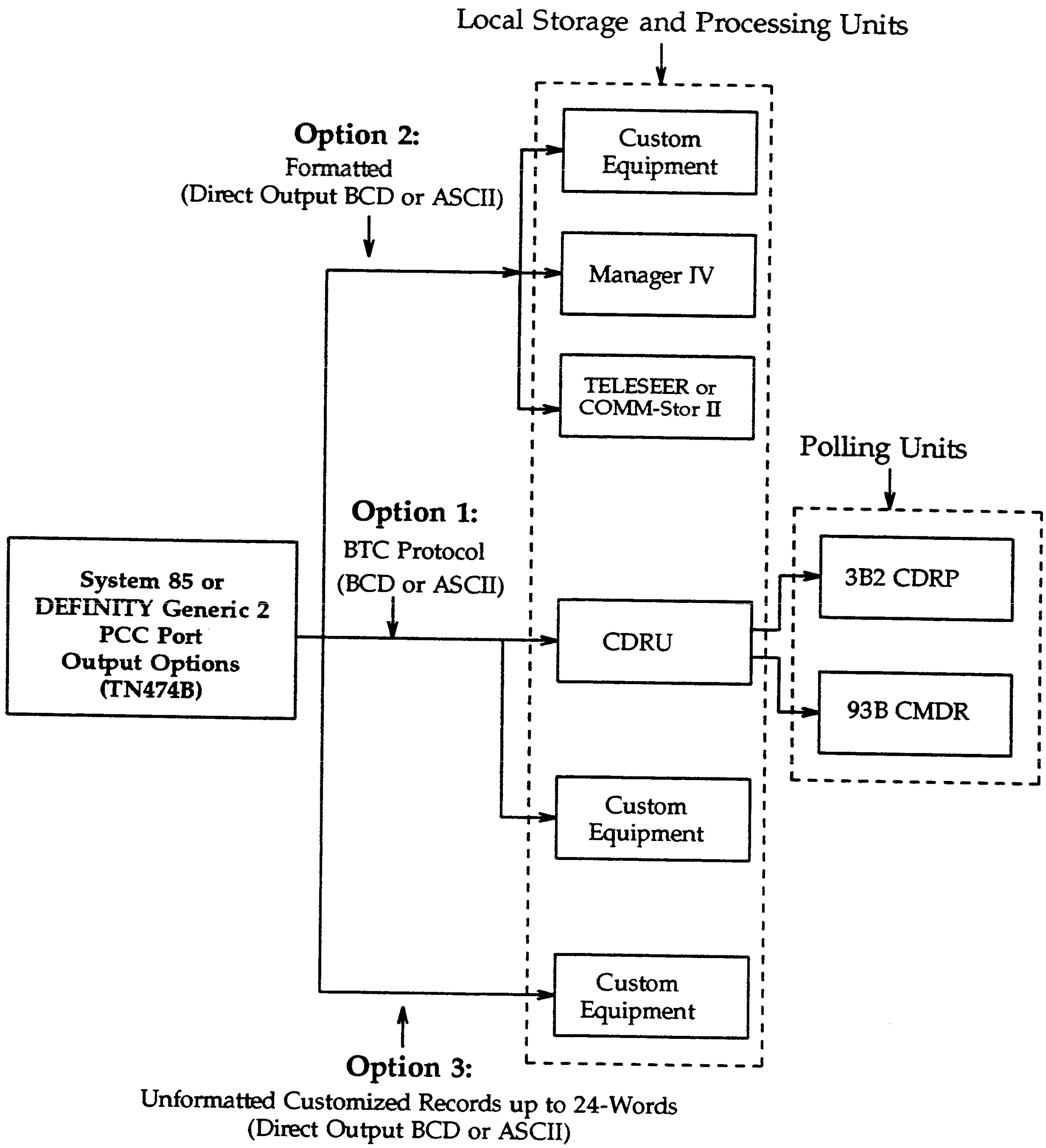


Figure 27-7. PCC Port Output Options

Data Items for VFCDR

While the *recommended standard* formats call for specific data items, this is to accommodate specific peripheral devices and is not a constraint of the VFCDR configuration. All of the data items available with the CDR feature can be accessed with the VFCDR configuration. The complete list of available data items is described under Recordable Data Items (earlier in this chapter) and shown in Table 27-C.

Call Record Formats With VFCDR

Table 27-E shows an example of data items that can be selected with variable format. Included in the table are some assorted calls to simulate customer formatted output.

TABLE 27-E. Example of Items in Formatted VFCDR Record for Various Call Types

Call Type	Data Items											
	VDN QDN	Cond. Code	Feat. Flags	Trunk DAC Used	IXC	Dialed Number	Calling Number	Node Number	Incoming Trunk DAC	Console Number	Call Duration	Agent Login
Outgoing ARS Call		7	4	621			86113	1			0105	
Outgoing CO Call		A	0	621	2	357-7192	86181	1			0315	
Outgoing Tie Trunk Call		A	4	183	1	1-303-451-1857	86374	1			0638	
Incoming Tie to ACD Call	82000	9	0			86374	439	1	439		0972	1234
Incoming CO to Attd to ACD Call	82000	1	0			1-602-341-7894	621	1	411	3	0333	1234
Incoming DCS Call		9	0			82600	111	1	111		0235	
Outgoing DCS Call (Main/Sat)		A	0	711	1	87766	86434	1			0237	
Incoming ISDN Call		9	0		2	357-2323	214	1	214		0875	
Outgoing ISDN Call		A	7	337	288	624-2007	82005	1			0138	

NOTE: All the possible data items are not included. Refer to the "Recordable Data Items" section for more information. The blank positions in the chart represent not applicable data. Special processing would be required to format data as shown in this table.

Table 27-F contains a different set of data items. Tables 27-E and 27-F are examples of *custom formatted* information that can be obtained from one switch. It may be possible to obtain all the data shown in these two examples in one format; however, the number of digits cannot exceed 96 (the number of cells available in a 24-word format). A record of this size and content can only be obtained with the variable format CDR configuration.

TABLE 27-F. Example 2 of Items in Formatted VFCDR Record for Various Call Types

Call Type	Data Items								
	Time of Day	Account Code	Auth. Code	Time in Queue	FRL	Incoming Circuit ID	Outgoing Circuit ID	ISDN Network Service	Extension Partition Number
Outgoing ARS Call	1321	0321	70721	12	3		217		? A ?
Outgoing CO Call	1322	0367	63210		4		311		213
Outgoing Tie Trunk Call	1340	0921	93210	02	2		7		562
Incoming Tie to ACD Call	1355				6	31			15
Incoming CO to Attd to ACD Call	1400			120	5	71			106
Incoming DCS Call	1423				7	55			45
Outgoing DCS Call	1424	0678			7		59		297
Incoming ISDN Call	1425				7	75		251	78
Outgoing ISDN Call	1427	5201			3	86		200	99

NOTE: All the possible data items are not included. Refer to the "Recordable Data Items" section for more information. The blank positions in the chart represent not applicable data. Special processing would be required to format data as shown in this table.

Recommended Standard Formats

Recommended Standard 18-Word ISDN Format

The storage and output options for the Recommended Standard 18-word ISDN record format are the same as for the 18-word default record, described previously under the CMDR Configuration. Specific differences between the two formats are:

- In the ISDN format the IXC code is expanded from 1- to 3-digits and moved from cell 4 of word 18 to cell 4 of word 3 and cells 2 and 3 of word 4.
- The time-in-queue and FRL data items in word 15 are replaced by the ISDN Network Service Value item.
- The FRL data item which was in cell 4 of word 15 is moved to cell 4 of word 18.

Processing of the recommended standard 18-word ISDN format is not supported by TELESEER SMDR. The CDRU and Manager IV provide processing support for this format.

For customized processing, customers can choose from host processors or other vendor equipment that might be modified to support the recommended standard 18-word ISDN format. Figure 27-8 shows this format.

Word \ Bit	15 14 13 12	11 10 09 08	07 06 05 04	03 02 01 00
01	Opcode	Call Duration Hours	Minute (Xx)	Minute (xX)
02	Opcode	10th Minute	Condition Code	IXC 1
03	Opcode	IXC Code 2	3	Access 1 or 2 1
04	Opcode	Access Code 1 or 2 2	3	Dialed Number 1
05	Opcode	Dialed Number 2	3	4
06	Opcode	Dialed Number 5	6	7
07	Opcode	Dialed Number 8	9	10
08	Opcode	Dialed Number 11	12	13
09	Opcode	Dialed Number		Calling Number
10	Opcode	Calling Number / Incoming TG DAC 2	3	4
11	Opcode	Account Code 1	2	3
12	Opcode	Account Code 4	5	Authorization Code 1
13	Opcode	Authorization Code 2	3	4
14	Opcode	Authorization Code 5	6	7
15	Opcode	ISDN Network Service 1	2	3
16	Opcode	Calling Number 5	Incoming Circuit ID 1	2
17	Opcode	Feature Flags*	Outgoing Circuit ID 1	2
18	Opcode	Out Circuit ID 3	In Circuit ID 3	FRL

***** Feature Flags: This field contains a single hexadecimal character that indicates the status of specific features. For an explination of these hexadecimal codes see Table 28-B.

Figure 27-8. Recommended Standard 18-Word ISDN Format

Recommended Standard 24-Word Format

This format contains the fields that are most commonly required by CDR applications beginning with System 85, Release 2 Version 4. The 24-word format has evolved from the R2 V4 version to the Generic 2 version which is shown in Figure 27-9. The only difference between the R2 V4 and Generic 2 formats is the new data item "ISDN User-to-User Information (UUI)" located in word 22 of the Generic 2 record format. This new format is a standard for both DEFINITY Generic 1 and Generic 2.

The 24-word recommended standard format is constructed on the switch by administration of the individual fields. The options available for the administration of the 24-word format are similar to those for the recommended standard 18-word ISDN format described previously.

CDRU, CDRU/S, and Manager IV support the processing of the 24-word recommended standard format. Customers can choose from host processors and vendor processing for customized processing of the data, provided that their systems have been designed to receive the administered format. Figure 27-9 illustrates the recommended standard 24-word format.

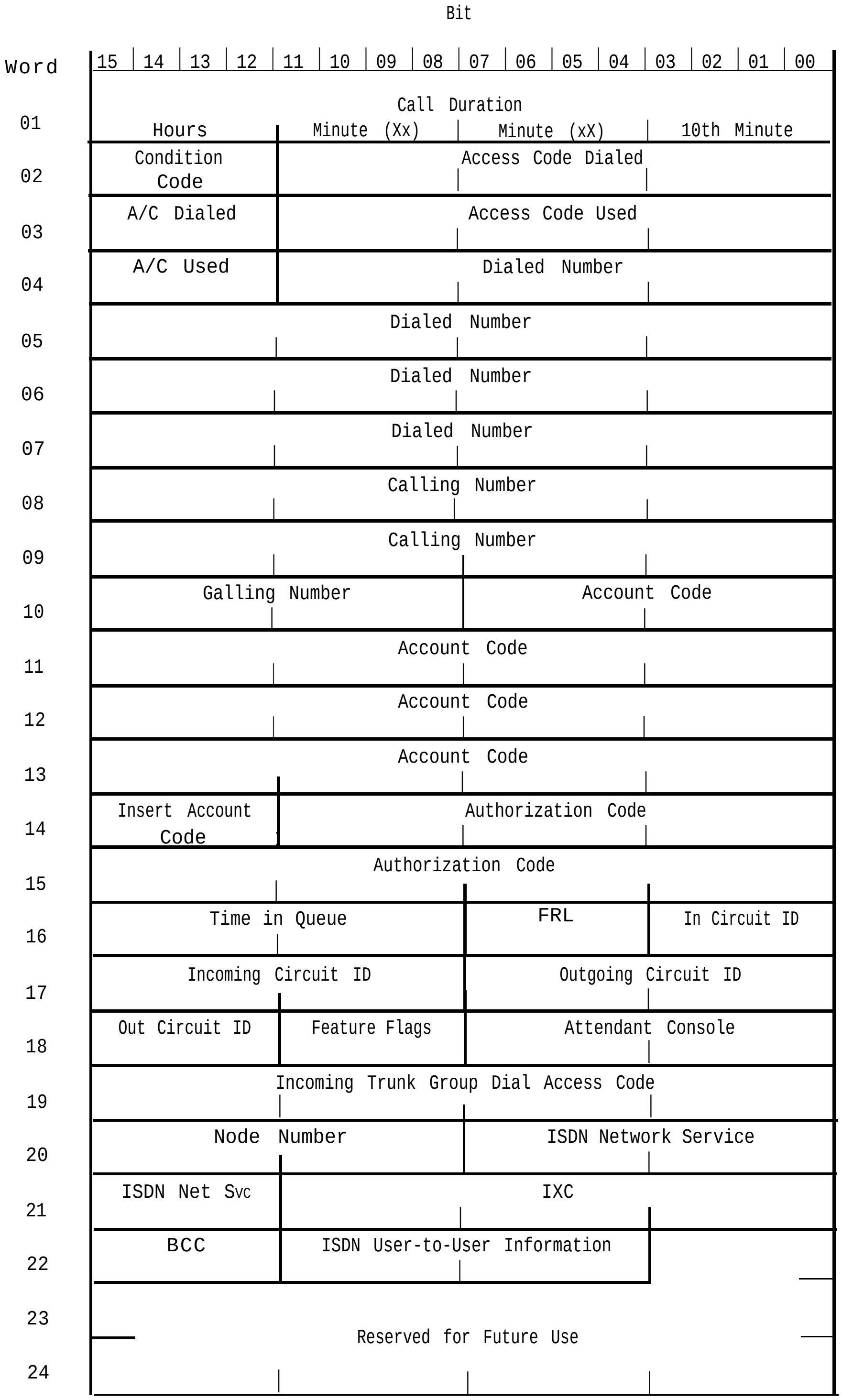
> **NOTE:** While the recommended standard format is satisfactory for most applications, local requirements and system evolution may dictate local customizing. For example, the recommended standard format provides for a single digit BCC (word 22). While this design will accommodate the nine predefined BCCOSs, with the BCCOS feature it is possible to have up to 255 BCCOSs. Where custom designed BCCOSs are used and need to be recorded, the 24-word format can be customized to meet this need.

PCC (Processor Communication Circuit) Port

The PCC Port is required for the VFCDR configuration. The PCC port offers the most flexibility for Call Detail Recording. The primary storage unit connected to the PCC port is the CDRU. A group of select call data items are recorded by the switch and sent out the PCC port to the CDRU. The data presentations available across the PCC port are:

- 15- to 24-word output with BTC (Better Transmission Checks) Protocol (BCD or ASCII)
- 18-word Formatted Direct Output (BCD or ASCII)
- Unformatted Direct Output (BCD or ASCII).

See the "CDR Ports And Data Presentations From The Switch" section for more information.



* Feature Flags: This field contains a single hexadecimal character that indicates the status of specific features. For an explanation of these hexadecimal codes see Table 28-B.

Figure 27-9. Recommended Standard 24-Word Format

VFCDR Adjuncts

Storage Units

Any of the following machines can be configured as a VFCDR storage unit.

- 3B2/300 (upgraded to UNIX® System 3.2)
- 3B2/310 (upgraded to UNIX System 3.2)
- 3B2/400
- AT&T 6386 WGS (Work Group Station) PC
- Call Detail Recording Unit/Small.

The 3B2 and 6386 CDRUs perform the functions of a 94A LSU with greater storage capacity and processing power. The CDRU is connected to the switch via the PCC and can receive CDR records in BCD over an RS-232 link at various baud rates. The BTC protocol is used to provide error detection and record retransmission. The CDRU can also send CDR records to direct output devices or remote polling devices (such as the 93B Poller or NCOSS) via the DDCMP protocol.

3B2 CDRU (Call Detail Recording Utility)

The 3B2 CDRU is an accounting program that runs on the AT&T 3B2 models 300, 310, or 400 computers. It collects and stores statistics about calls for the System 85, R2V4 or DEFINITY Generic 2 switches.

A 3B2 CDRU can receive up to 37,000 18-word CDR records per hour over a 9600 bps link to the switch and can store about 22,000 records per megabyte of disk capacity up to a maximum of 70 megabytes.

The 3B2 CDRU system can send records at data rates of from 1200 bps through 19.2K bps from its polled and direct output ports.

The 3B2 CDRU can operate with the expanded storage capabilities offered by the 3B2 Expansion Module [reference AT&T 3B2/XM Manual (305-538)] and the Remote Management Package that includes the AIC (Alarm Interface Circuit) for interface with remote alarm devices.

An optional UPS (Uninterruptible Power Supply) is available to help ensure that records are not lost in the event of power outages and surges. The AT&T 1KVA UPS power supply or equivalent is recommended. For more information on the CDRU Systems, refer to the ***Call Detail Recording Utility*** (555-006-211).

6386 CDRU (Call Detail Recording Utility)

The 6386 CDRU is an accounting program which collects and stores statistics about calls for System 85, R2V4 or DEFINITY Generic 2 switches. It runs on the AT&T 6386 WGS (Work Group Station) PC.

The 6386 CDRU system can collect 53,000 18-word records per hour over a 9600 bps link from a System 85 Release 2, Version 4 or Generic 2. The 6386 CDRU system can store approximately 22,000 18-word records per megabyte of disk capacity up to a maximum of 70 megabytes.

The 6386 CDRU system can send records at rates from 1200 bps through 19.2K bps from its polled and direct output ports. For more information on the 3B2 and 6386 CDRU Systems, refer to the ***Call Detail Recording Utility*** (555-006-211).

Call Detail Recording Unit/Small

The CDRU/S is a local storage unit designed for low volume call detail recording. The CDRU/S can receive 15-, 18- or 24-word records. The maximum storage capacity of the CDRU/S is 7008 records (15-word record). See the ***Call Detail Recording Unit/ Small User Guide*** (555-006-212) for detailed information.

The CDRUs can interface with the following machines (Refer to Table 27-E for format and device compatibility):

- ***3B2 CDRP (Call Detail Record Poller).*** The 3B2 CDRP is an application software package that runs on the AT&T 3B2/400, 500, and 600 computers. In a local configuration the CDRP can poll several CDRUs to obtain as many as 24,000 18-word call records per hour over 19.2K bps direct links. The CDRP supports output of BCD, ASCII, and EBCDIC character sets. In a remote configuration, the 3B2 CDRP can poll at rates of from 1200 to 9600 bps.
- ***DEFINITY Manager IV.*** The CDRU supports Manager IV either in a local configuration (the CDRU collocated with Manager IV) or in a remote configuration (Manager IV polls the CDRU using a 93B CMDR as a front end). Unlike the 3B2 CDRU, Manager IV cannot yet process the variable format.
- ***93B CMDR.*** The 93B CMDR in a remote location polls the CDRU over a dial-up 1200 bps connection, to obtain call records. The 93B does not support variable format CDR records. The 93B expects to receive 15- and 18-word default or recommended standard formats.
- ***NCOSS Network Control Operations Support System).*** The NCOSS system polls the CDRU to obtain call records over the dial-up network at 1200 bps.
- ***Customer-Owned Peripherals.*** Host processors, tape drives, printers, and polling equipment can accept variable and default formats.

VFCDR Polling Systems

In networking customized formats created through VFCDR require polling devices sophisticated enough to recognize the format being sent across the network by the CDRUs. The polling device must have the capability to recognize the record's code and protocol. This is necessary for record sanity.

- The 3B2 CDRP (Call Detail Record Poller) is an AT&T application software package that runs on the AT&T 3B2 line of computers. CDRP acquires, organizes, and stores call records sent from many record collection systems, then transmits the records to a host processor or tape storage device. The 3B2 CDRP has the following capabilities:

— Can obtain 80,000 18-word BCD call records per hour over a 19.2K bps direct link.

— Supports output of BCD, ASCII, and EBCDIC character sets.

— Can poll remote location at 1200 to 9600 bps.

— Services up to 128 total input devices.

— Capable of simultaneously polling seven CDRUs or LSUs.

— Supports 15- to 24-word formats (fixed and variable format).

— Maintains a complete polling and transmission history.

— Supports asynchronous and SCSI (Small Computer System Interface) tape output devices.

— Allows remote maintenance and administration of CDRUs and LSUs.

- The 93B CMDR can be interfaced with 3 CDRUs at a time to provide polled call detail information at a centralized location. The CDRUs must be administered with either a 15- or 18-word default format in order to send records to the 93B. The 93B can also be connected to 94A LSUs through 212A or equivalent data sets over the DDD network. This scheme only works with CMDR port output. For more information refer to the ***Call Detail Acquisition and Processing Reference*** (555-006-202) manual.

VFCDR Processors

Certain processing systems, retrieval, and storage devices can only handle some of the possible CDR formats provided by the switch. For example, the Manager IV and NCOSS processors can handle all the default and recommended standard formats, but cannot handle a variable format (other than recommended standard). See Table 27-D for possible combinations.

User Operations

The following are the user operating procedures for all three CDR options (VFCDR, SMDR, CMDR).

General

CDR reports on selected trunk groups. These trunk groups are identified in local switch translations described later in the "Feature Administration" section. Once a trunk group has been identified as reportable, actual reporting is controlled from an attendant console at the recording switch. The attendant can control (turn on or off) reporting on an individual trunk-group basis.

To Activate Trunk-Group Recording at an Attendant Console:

1. Press an idle loop button. [PA lamp goes out].
2. Press the **[START]** button. [Dial tone]

3. Dial the CDR—Start access code. [Second dial tone]
4. Dial the access code for the desired trunk group. [Recall dial tone]

   NOTE: Step 4 can be executed repeatedly for as many trunk groups as desired.

5. Press the **[RELEASE]** button. [PA lamp lights].

To Deactivate Trunk-Group Recording at an Attendant Console:

1. Press an idle loop button. [PA lamp goes out]
2. Press the **[START]** button. [Dial tone)
3. Dial the CDR—Stop Access Code. [Second dial tone]
4. Dial the access code for the trunk group you want to stop recording. [Recall dial tone]

   NOTE: The previous step can be repeated to deactivate recording on a series of trunk groups.

5. Press the **[RELEASE]** button. [PA lamp lights].

Entering an Account Code

On a switch where traditional FEAC is used:

1. Go off-hook. [Dial tone].
2. Enter the Account Code DAC. [Second dial tone].
3. Enter your assigned account code. [Third dial tone].
4. Finish dialing your number, including network DAC, IXC code, etc, as applicable. [Call progress tones as appropriate].

On a System 85 or Generic 2.1 switch where the account codes prefix is defined in AAR administration:

1. Go off-hook. [Dial tone].
2. Enter the AAR DAC. [Second dial tone].
3. Enter the account code prefix and your assigned account code followed by the number you wish to call (including IXC code, etc. as applicable). [Call progress tones as appropriate].

On a switch where World Class Routing uses string identifiers for account codes:

1. Go off-hook. [Dial tone].

2. Enter the Network DAC. [Second dial tone].
3. Enter your assigned account code followed by the number you wish to call (including IXC code, etc. as applicable). [Call progress tones as appropriate].

General Considerations

The "Considerations" section for this feature is divided into two major sections: "General Considerations" and "Considerations That Determine the CDR Configuration". The following are the General Considerations for the Call Detail Recording feature.

The 302 Processors

Do not stack 3B2 computers directly on top of one another. Use shelves to separate computers. The 3B2 computers require good air circulation for cooling. Also, use dedicated power feeders to maintain system integrity.

Answer Supervision

When Answer Supevision is provided on an outgoing trunk call, the call duration field accurately reflects the connect time. Call supervision restarts the call timer and overrides the default timer values. The pre-Release 2, Version 4 administrable default timer values are not as accurate as Answer Supervision. Answer Supervision is limited to a set of trunk signaling types and the answer supervision flag in the feature flag field shows if answer supervision was received.

Attendant Console Number

This field records the attendant console number of the attendant that assisted in call completion. If an attendant does not assist in completing a call, this field contains an "A" (hexadecimal). An attendant console number is associated with every attendant call. This is useful for Tenant Services applications.

Blocking

The CDR feature can be administered to block calls on recordable trunks when the record buffer becomes full. In fact, this is the default. This is to prevent users from making calls that do not appear on the call record. To avoid such blocking, a "1" must be administered in field 14 of Procedure 275, Word 3.

CDR Output

Call records can be output to all ports at once. Specific port types are restricted on record length. For example, the SMDR Port, when administered to the SMDR configuration, is limited to 15-word records. See the section entitled "CDR Ports and Data Presentations From the Switch" for specific details.

Customer-Owned Peripherals

Customers have the option of attaching host processors, tape drives, and polling equipment, rather than AT&T processing systems, to customize the call detail record collection process. In addition, many software firms currently offer CDR processing. Processing is offered either on a service-bureau basis, where the customer sends tapes to the service bureau for processing (generally, on a monthly basis), or through the lease or license of processing software.

Data Presentations

The CDR ports on the DEFINITY Generic 2 switch can be administered to output different types of output code. Refer to the "Data Presentations From The Switch" section for details. The data presentation administered must be compatible with the storage, processing, or output device receiving the data. Refer to the device's documentation for input data specifications.

Enhanced Trunk Signaling and Error Recovery

Enhanced Trunk Signaling and Error Recovery provides CDR records pertaining to ineffective call attempts.

Extension Number Steering

The call record shows calls to data ports (DCA or DCP) and calls to Main/Satellite locations over Main/Satellite trunks. When these calls are placed using Extension Number Steering, the call record shows both the dialed extension number and the dial access code of the associated trunk group.

Five-Digit Dialing

For systems using 5-digit dialing plans, the extension number is truncated by suppressing the leading digit on the SMDR output (either direct output or 9-track tape). For example if the extension number being recorded is 64221, the SMDR output shows 4221 and the leading digit "6" is not shown. For prefix type 5-digit dialing, this should not be a problem. SMDR was designed before a 5-digit dialing plan was used or needed. CMDR and VFCDR support expanded digit dialing plans. See the Extension Number Portability and the DCS features for more information on 5-digit dialing plans.

FEAC (Forced Entry of Account Codes)

Certain network calls can require the terminal user to enter an account code before (or during) dialing. (These can included toll calls, calls routed over selected trunk groups and calls attempting to use selected networks).

If an account code is required for a preference and one was not dialed, that preference is skipped for call routing purposes. With variable format, the account code field may or may not be administered in a CDR record. If it is not administered, Forced Entry of Account Codes is ignored. See "FEAC (Forced Entry of Account Codes) Administration" in the "Administration" section.

Attendant Participation With FEAC

If FEAC is in effect and a call is placed by an attendant, the FEAC requirement is not enforced. However, if a call is initiated by the attendant and then returned to station control (for example, a Through Dialing call), FEAC requirements are enforced and the attendant must enter an account code before returning control to the station, or the call will fail.

Multidigit Steering

The call record shows calls to DCA (Data Communications Access) ports and calls to Main/Satellite locations over Main/Satellite trunks. When these calls are placed using Multidigit Steering, the call record shows both the dialed extension number and the dial access code of the associated outgoing trunk group.

Multiple Records on Network Calls

In a network environment where the CDR feature is used on multiple nodes, the same call is recorded separately on each node where a recordable trunk circuit is used.

NCOSS (Network Control Operations Support System)

The CDR ports used for NCOSS are connected to the TN403 board. The CMDR/NCOSS application type must be administered in Procedure 253, Word 1.

Radio Frequency Shielding

3B2 computers and their cables are subject to radio frequency interference and thus may require shielding. Interference may be caused by radio or TV transmitters, radio frequency binding or sealing equipment, and X-ray equipment. Radar transmitters may also cause interference.

Reportable Trunk Groups

Trunk groups can be assigned in switch translations as reportable or nonreportable. Activity on nonreportable trunk groups cannot be recorded by the CDR feature.

SMDR Cabinet Separation

The SMDR cabinet must be located within 200 feet of the common control cabinet because of high-speed data transmission between the two systems. The SMDR cabinet should not be located in a building without lightning protection.

SMDR Direct Output

- Memory Capacity

  The direct output memory can store call data for up to 16 calls at a time.

- Effect of Heavy Traffic

  With the SMDR adjunct and printer combination, the printer can use excessive paper and miss call activity during heavy traffic periods. It is recommended that the 9-track tape, 94A LSU (with a 93B), AP-16, or 3B2 versions be used in systems with heavy traffic.

SMDR Power Failure Protection

The power reserve circuit pack (LC38), located in the SMDR cabinet, provides a limited power backup for memory circuits in the event of a power failure. This power reserve enables the SMDR memory to hold call data for about 7 hours and allows the clock and calendar, located on the front of the cabinet, to continue to run (without display).

Call data for calls made during a power failure is not recorded. If a power failure lasts less than 7 hours, the call data retained in memory should be valid. However, if a power failure lasts longer than 7 hours, the clock and calendar stops, and call data stored in the SMDR memory will either be lost or cannot be assumed to be valid.

SMDR Port

The SMDR port can be administered as an eighth CMDR/NCOSS port.

SMDR Testing

The call records are not recorded during the execution of the SMDR link-test procedure. This link test prevents the CDR network from outputting records to the SMDR port.

Time Stamping

The CDR records can be time stamped either by the switch (when VFCDR is used or when using the 18-word formatted direct output from the PCC), or by the storage/polling devices.

Time-of-Day Insertion Using VFCDR:

If unformatted is selected, the time-of-day data item can be administered using variable format, and inserted into the CDR record by the switch. In this case, the time-of-day data item must be assinged to an appropriate field in the record format.

Time-of-Day Insertion by the PCC:

The time of day is written into the PCC dual port RAM by the 501CC at 30-second intervals. If the 18-word formatted output is chosen, the PCC will use this time of day to time stamp the record. This is done by inserting the time into the proper location. Unformatted direct outuput and BTC protocols do not use PCC time stamping. If the BTC protocol is selected, then the peripheral (such as a 3B2 CDRU) is expected to time stamp the records.

Time-of-Day Insertion by the CDRU:

The CDRU is required to time stamp incoming CDR records when they are received from the switching system. The computers contain an internal clock which provides the time for the CDRU software.

When the CDRU is connected to a 93B, the 93B controls the time-of-day clock on the 3B2 computer or 6386 PC. When the CDRU is not connected to the 93B, the CDR records are time stamped by the computer clock (set locally).

Variable Call Completion Thresholds

The switch scans each CDR call once every 6 seconds. The call duration timer is started when an active CDR call is found (resulting in a timing accuracy limit of 6 seconds). This compensates for time spent waiting for the far end to answer the call. Users can increase the delay time by administering a start timer that delays starting the duration time by up to 99 seconds. This is done on a trunk-group basis and on a system basis. If used, the trunk-group setting takes precedence over the system setting. If the "Feature Flags" data item is recorded, timing is restarted when answer supervision is received. This action overrides the threshold timers.

Considerations That Determine the CDR Configuration

A CDR configuration is the arrangement required to meet customers' call management needs. It consists of the customers' switches and associated peripherals (including storage, polling, and processing devices). For most applications, the recommended standard 24-word format can provide sufficient data for records. Based on all possible options in the System 85 or DEFINITY Generic 2 switch and in the peripheral systems, various CDR configurations are possible in the DEFINITY Generic 2 time frame. This section describes the factory that affect customers' choices of CDR configurations, and highlights some specific configurations along with their strengths and constraints. The specific configuration a customer chooses will depend on several factors:

- CDR volume at the switch(es)
- Customers' switch capabilities (such as ISDN and Tenant Services) which in turn impact the CDR format chosen
- Cost management needs and the degree of versatility desired in the customers' processing systems
- Customers' CDR processing needs (local versus central).

CDR Volume at the Switch(es)

The amount of call traffic through the customers' switch(es) is an important factor in planning CDR configurations. Before installing a CDR configuration, a study of associated storage and processing devices should be performed. Detailed information can be obtained from the following documents:

- ***Call Detail Aquisition and Processing Reference*** (555-006-202) manual
- ***Call Detail Recording Utility System Description*** (555-006-210)
- ***Call Detail Recording Utility Installation, Administration, and Maintenance Guide*** (555-006-110).
- ***Call Detail Recording Unit/ Small User Guide (555-006-212)***

Customers' Switch Capabilities

The capabilities the CDR user chooses to administer on the switch determine the choice of CDR format, and hence the CDR configuration. For example, an ISDN customer has the choice of either a recommended standard 18-word ISDN format, a recommended standard 24-word format, or a variable (locally designed) format.

If the CDR user desires network CDR capability with a variable format, a 3B2 CDRP can be used to poll the remote CDRUs.

Cost Management Needs

The customers' Cost Management needs and the sophistication desired in the reports generated by their processing systems, affect customers' CDR configurations. Manager IV, TELESEER SMDR and AT&T CAS all support Cost Allocation and Call Record Analysis. In addition, Manager IV supports Internal Billing and History Search features. The following paragraphs describe some of the differences between these systems.

Cost Allocation

Cost Allocation is a reporting process that provides customers the ability to recover the cost (or cost-plus-profit) incurred in their network. In Manager IV, cost allocation is a normalized process in which the allocated costs are proportional to the vendor's bill which is the actual cost. TELESEER SMDR and AT&T CAS simply calculate the costs and do not normalize them over the actual cost.

Manager IV allows customers to allocate costs for calls, recurring charges (such as station equipment, common equipment, and customer-defined recurring costs), and nonrecurring charges (such as additions and moves). TELESEER SMDR and the AT&T CAS only provide cost information for calls.

In Manager IV, processed call records can be directed to magnetic tapes, or directly downloaded to the customer host processor for additional processing, high-speed printing, or integration into the customer accounting process.

Internal Billing

The Billing feature, supported by Manager IV, allows customers to produce account invoices, to post payment of and to make adjustments to those invoices, as well as to track outstanding balances internally.

History Search

The History Search feature, supported only in Manager IV, provides a selective search capability of the calls for call trace and pattern analysis. The 2-day search can be conducted over any field (such as data, time, or extension) or any combination of fields in the call record.

Customers' Processing Needs

The differences between customers' local and central processing needs can serve as criteria in determining the customers' CDR configuration.

Customers have the option of performing CDR processing on a local or central basis. For *local* processing, CDR data is collected at each switch location. For *central* processing, the CDR data from different switches is collected and accumulated at a central point. *Networked* processing is a special case of centraI processing in which data is generally collected at the network tandems. Calls made from the subtending locations behind each tandem can be cost allocated based on the data at that tandem. This data does not generally identify the calling extensions unless every call is made using authorization codes. Thus, call detail recording made at tandems has the drawback of allowing cost allocation by location only (and not by extensions). On the other hand, customers can choose to have call recording at each switch location, including tandems and subtending switches in the network. This would allow cost allocation on an extension basis even though it is not cost effective.

Customers in a network [such as ETN (Electronic Tandem Network) and DCS (Distributed Communications System)] have a definite need for a product that does CDR processing on an integrated (networked) basis. There are significant complexities in a network-oriented process. In a network environment, one has to deal with multiple switches and multiple switch types, which are not an issue in a stand-alone environment. The tariff databases become much more complex in a network environment. For a single location, only one set of tariff databases is needed for costing purposes. Multilocations have intralata and interlata specific to each location which may result in a wide variety of local, intralata, and interlata databases.

This is further complicated by multiple subtending switches behind each monitoring location which may result in an even greater number of tariff databases. Another complexity with a network environment is that a single call may result in multiple records being generated (for example, at an intermediate tandem), in contrast to a single record/call in a stand-alone environment.

Of the current AT&T processing products, only Manager IV provides centralized and integrated system management capabilities across multiple product types in the customer's network. Although it is targeted for a networked environment, Manager IV can also support a single switch as well as multiple stand-alone switches. CDRU, TELESEER SMDR, and AT&T CAS (Call Accounting System) can only support a single-site switch for CDR.

Local Storage and Processing

Customers with local processing needs can choose from customized processing or AT&T provided processing based on their needs. Specific peripherals may have limitations on which formats that can support. For example, Manager IV does not support the variable (customized) format. However, customers can still administer the variable format record structure from Manager IV if Manager IV is the administration vehicle for their switch. The AT&T peripherals and the formats they can support are shown in Table 27-D (shown earlier).

Local Storage and Processing Systems

the choice of a local processing system is determined by the customers' cost management needs and the extent of CDR recording made on their switches.

- The 3B2 and 6386 CDRUs (Call Deatil Recording Utilities) can perform all the functions of a 94A LSU and have greater storage capacity and processing power.

  the variable CDR format can be sent via the PCC board directly to a customer host for further processing, or it can be directed to a CDRU. From the CDRU, a direct output option can interface to a customer host for further processing or to a tape drive for storage.

  Of the two output options, each has its pros and cons. Although it is definitely cheaper to use the direct output options on the switch, there are certain advantages to using the CDRU.

  The CDRUs can use theBTC protocol which provides error checking of data and therefore, is more reliable. The CDRU also buffers the CDR data during periods of high call activity. In addition the CDRU has a filtering capability that allows the customers to discard calls meeting certain customer-specific crtieria. However, for the filtering capability to work for the variable CDR format, certain fields (such as call duration and condition code) should stay in specific positions in the record (as specified in the recommended standard formats).

  For more information refer to the ***Call Detail Recording Utility*** (555-006-211) Manual.

- The CDRU/S (small) is a local storage unit used for low volume storage applications. For more information refer to the ***Call Detail Recording Unit/Small User Guide*** (555-006-212) for information.

- Manager IV offers the following features: In addition to Cost Allocation and CDR Analysis (also supported by TELESEER SMDR and AT&T), Manager IV performs billing/accounts receivable and history search. The Cost Management application in Manager IV shares a common database with its TCM (Terminal Change Management) and FM (Facilities Management) applications. The integration of applications in Manager IV reduces duplication of tasks. Customers with larger CDR volumes than Manager IV can normally support have the option of negotiating Manager IV custom development. Refer to the ***Manager IV Cost Management User's Guide*** (585-220-704).

- AT&T CAS is an option for customers with CDR record volumes up to 150,000 per run and up to 5000 stations, and for whom cost allocation and CDR analysis are important considerations. Refer to the ***AT&T Call Accounting System User's Guide*** (555-006-201).

- TELESEER SMDR is an option for CDR record volumes that do not exceed 28,000 per run, and only cost allocation and CDR analysis are important considerations. Refer to the ***TELESEER SMDR Reference Manual*** (999-302-175 IS).

- Host processing is an option if the variable format option is used with local (PCC) connectivity.

Centralized Processing

Multiple switches in a single location or multiple switches in multiple locations have a need for central processing of their CDR data. Networked customers also fall in this category.

In central processing, the CDR data from different switches is collected (one switch at a time) and accumulated at a central point, until the central facility, a poller, can poll the storage device for the local CDR data. Similar to customers who perform local processing, customers with centralized processing needs have the option of choosing customized processing or AT&T provided processing. In the customized processing option, customers can choose from the variable record format or one of four fixed formats. For centralized AT&T provided processing, customers can choose from only four fixed CDR formats.

For processing provided by AT&T, ISDN customers interested in central processing can choose data items to administer for the 18-word ISDN format. It is recommended that the 18-word Recommended Standard ISDN format is used with Manager IV processing.

Centralized Processing Systems

The choice of a centralized processing system depends on the record volume and the application for which the information is being used.

- Manager IV is the only AT&T processing system that supports centralized processing of the 24-word recommended standard format. Refer to Manager IV documentation or the ***Call Detail Acquisition and Processing Reference*** (555-006-202) manual.
- The 93B can be used to poll call records (15- and 18-word) from 94As and CDRUs, but cannot provide the processing power of Manager IV. Refer to the ***CMDR Descriptive Information 93B CPS (Customer Premises System)*** (190-403-100).
- The 3B2 CDRP (Call Detail Record Poller) can be used to poll call records (15-, 18-, and 24-word from the 94A LSUs and CDRUs. Refer to the ***Call Detail Record Poller*** (555-006-213) manual.
- NCOSS (Network Control Operations Support System) is part of AT&T services provided to ETN customers. NCOSS collects CDR data from an ETN being monitored. NCOSS provides reports, and searches on specific data to accomplish network operations support tasks, particularly network trouble localization and network performance assessment.

  NCOSS collects data by periodically polling a 94A LSU / CDRU at a customer switch. NCOSS polling is done over an independent port on the 94A LSU / CDRU, and thus does not disrupt the customer's data collection process. NCOSS support for any ETN site is limited to the format the customer has chosen to administer for that site. For example, NCOSS cannot design an ISDN network explicitly, unless a customer has chosen to administer the recommended standard 18-word ISDN format from the switch (refer to NCOSS documentation for details).

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of the CDR feature.

Abbreviated Dialing

Outgoing calls are recorded by this feature as though they were manually dialed.

Advanced Private Line Termination (APLT)

- Incoming Tie Trunk Calls

  If a calling party dials an account code and then calls across a tie trunk to an attendant on another network node, the account code is recorded only at the originating node. The attendant at the terminating node can record the account code as requested by the caller.

- Outgoing Tie Trunk Calls

  If a calling party dials a CDR account code and then calls the attendant, the account code is dropped. However, the attendant can reenter the account code before extending the call across the tie trunk.

Attendant Call Waiting

The CDR record generator records the called extension number and account code for attendant-assisted incoming calls using the Attendant Call Waiting feature.

AAR (Automatic Alternate Routing)

Incoming: A Private Network Access caller cannot input a CDR account code. If a call to the attendant is extended and the attendant dials an account code, the account number is recorded.

Outgoing: If the calling party dials an account code and then calls the attendant, the account code is lost. The attendant can, however, input the account code on such a call, and the number is recorded.

On System 85 and DEFINITY Generic 2.1 switches, a terminal user can dial the account code before dialing the AAR access code.

When FEAC (Forced Entry of Account Codes) is active on a trunk group basis, account codes may be required for specific trunk groups within AAR patterns. If a FEAC trunk group appears in a routing pattern and the account code has not been dialed, AAR will skip that trunk group.

ACD (Automatic Call Distribution)

On an ACD call, the CDR feature records the answering agent's extension number as the called number, rather than the split's associated extension number.

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, if terminal class of service or system class of service requires an account code [FEAC (Forced Entry of Account Codes) is active], and if a code is not dialed, the switch returns intercept treatment. If a trunk group administered with FEAC appears in a routing pattern and the account code has not been dialed, ARS will skip that trunk group. See the "Forced Entry of Account Codes" section in the "Administration" section.

If an ARS (time-of-day) plan is being changed from one plan to another (three possible plans), the CDR (Call Detail Recording) unit will send a special record to show this. The newly entered ARS plan (1, 2, or 3) is recorded in the Calling Number field to show that the ARS plan has been changed. The appropriate control mode is recorded in the Control Mode field for 24-word and variable format (in the FRL field for 15- and 18-word formats). The rest of the CDR fields are empty, because an actual trunk was never seized.

Call Coverage

When a call is redirected and answered by a covering user, the extension number of the covering user, rather than the principal, is recorded in the Dialed Number field.

If a covering terminal user answers a redirected call on a soft number, the *primary extension number* of the covering user's voice terminal (not the soft number or the principal's number) is recorded in the call record.

CDRR (Call Detail Recording and Reporting)

CDRR is not supported on Release 2, Version 4 or DEFINITY Generic 2 switches.

Call Forwarding—Busy and Don't Answer

The call record shows the forwarded-to extension number as the called extension for the Call Forwarding—Busy and Don't Answer feature.

If a user establishes call forwarding outside, other users may attempt to call that extension and be routed over facilities which they might not have permission to access without an account code (because of FEAC). The callers would dial the extension number and be denied without a clear indication of why.

A terminal whose class of service requires an account code (administered through FEAC) entry is denied the ability to set up call forwarding off the switch in other than a DCS (Distributed Communications System) environment (only to terminal extensions within the DCS).

If the system class of service requires an account code (administered through FEAC), forwarding off-net will not be permitted through the networking features (AAR, ARS, or WCR).

Call Forwarding—Don't Answer

The call record shows the forwarded-to extension number as the called number for the Call Forwarding—Don't Answer feature.

Call Forwarding—Follow Me

The call record shows the forwarded-to extension number as the called number for the Call Forwarding—Follow Me feature.

Call Park

The extension number of the last voice terminal in a Call Park connection is recorded as the called number.

Call Pickup

The extension number of the terminal using Call Pickup (the terminal that actually answered the call) is recorded as the called number.

Call Vectoring

Call detail records show the extension number of the answering destination (rather than the VDN) as the called number when an internal or external call is processed by a vector.

CAS (Centralized Attendant Service)

The CDR feature is provided for calls extended by a CAS attendant or backup voice terminal. However, an attendant cannot activate CDR trunk-group recordings of RLT (Release Link Trunk) trunk groups.

Code Calling Access—Universal

An attendant with an incoming tie trunk call can dial the CDR access code and account code and then use code calling. The call record shows the call, not the page.

Conference—Attendant Five Party

The attendant can dial the CDR access code and account code before adding a voice terminal or a trunk to a conference. The CDR feature records the trunk or extension number.

Conference—Attendant Six Party

The attendant can dial the CDR access code and account code before adding a voice terminal or a trunk to a conference. When the codes are dialed, CDR records the trunk or voice terminal number of the added parties (whatever is available).

Conference—Three Party

If an account code had already been dialed to establish the original connection, the dialing of the new account code to add the second party will create a second call detail record for the added party.

If an established call is extended to a trunk using the conference or transfer features, the following Forced Entry of Account Codes rule applies: If an account code had already been dialed to establish the original connection, the dialing of a new account code to add the second party will result in two CDR records.

Data Call Setup

Forced Entry of Account Codes may be assigned for data terminal to data trunk calls. The same rules and restrictions apply as for voice terminals.

DCA (Data Communications Access)

The System 85 or DEFINITY Generic 2 CDR feature can provide statistical information for DCA calls.

DDC (Direct Department Calling)

The extension number of the answering agent in a DDC call is recorded by CDR as the called number, rather than the split's LDN (Listed Directory Number).

DID (Direct Inward Dialing)

The DID calling party cannot dial an account code. If the call is to the attendant, the attendant can dial the account code before extending the call. All DID calls will be recorded if recording is activated for the DID trunk group.

DOD (Direct Outward Dialing)

An account code may be required for direct access to trunk groups with Forced Entry of Account Codes administered.

DCS (Distributed Communication System)

Each switch in a DCS arrangement is assigned a "node number". A data item is provided which contains this node number. If needed, the accumulation of records from numerous switches can carry an identifying field showing the DCS node for a multinode call record. This field contains the value of the node number administered in Procedure 275, Word 3.

Extension Number Portability

Each switch in a portability network is given a "node number". A new data item is provided which contains this node number so that, if necessary, subsequent accumulation of records from switches carry an identifying field from which a multinode call can be reconstructed. This field contains the value of the node number administered in Procedure 275, Word 3. Note that this is the same as for the DCS feature. Where these two features are used together, the node number assignments in Procedure 275, Word 3 must match. That is the DCS node number for a given switch must be the same for both the DCS and the Portability Subnetwork.

FRL (Facilities Restriction Level)

A call record stores the FRL when one is used. If the FRL is raised to complete the call, the raised to FRL is reported in the CDR record.

Host Computer Access

The CDR feature provides information on and specific identification of data calls, including Host Computer Access calls. This enables managers to allocate resources to meet dynamic needs based on use.

Hunting

The call record shows the hunted-to extension number (the extension that actually answers the call) as the called number.

ISN (Information Systems Network) Interface

The CDR feature provides information on and specific identification of data calls including ISN Interface calls.

ISDN—PRI (Primary Rate Interface)

ISDN trunks may be dedicated for use as specific trunk types such as FX, WATS, CO, or tie trunks. Standard type records are satisfactory for their use. However, ISDN trunks may also be specified as *Dynamic Trunk Type*. Dynamic trunks are used for the call-by-call service selection capability provided with the ISDN—PRI feature. If so configured, the *ISDN NS* data item for the specific outgoing trunk is provided in the CDR record which can be used for billing purposes.

The *ISDN NS* data item specifies the network service used for an outgoing trunk (such as WATS).

The calling number field for incoming ISDN calls can record from 7 to 10 digits for incoming calls. The calling number contains the full ten digits of the incoming calling number, unless the calling number field is administered for less than ten digits.

The BCC (Bearer Capability Class) of ISDN is supported in Release 2 Version 4 and supplies voice/data characteristics of the call.

Intercept Treatment

If an incoming trunk call receives Intercept Treatment and is routed to the attendant, the call record looks as if the attendant had been dialed directly. This interaction occurs regardless of whether recording of ineffective attempts is active.

Look-Ahead Interflow

The sending switch only generates one CDR record for a Look-Ahead Interflow call. If the VDN call originated outside the switch, the switch records the incoming trunk-group dial access code as the calling number, and the digits sent from the Call Vectoring group list as the dialed number. If this incoming call is answered at the sending switch before the receiving switch accepts the call, the sending switch records the extension of the local answering destination as the dialed number.

Unless recording of ineffective attempts is assigned, the receiving switch only generates a CDR record for an accepted Look-Ahead Interflow call. When a call is accepted, the receiving switch records the incoming trunk-group dial access code as the calling number and the extension of the local answering destination as the dialed number.

Look-Ahead Interflow calls work like other calls when FEAC is assigned.

Loudspeaker Paging Access

With an incoming call waiting, the attendant can dial the CDR access and account code, and then use Loudspeaker Paging Access. The call record shows the call — not the page.

Main/Satellite

Main / Satellite trunk groups should not require the entry of an account code, since they are used for terminal-to-terminal calls. However, there is no requirement that restricts assignment of the option.

Personal Central Office Line

CDR does not record calls made on Personal CO (Central Office) Lines. Personal CO lines are for direct access to the Central Office and, therefore, not provided all the features of a PBX terminal extension. Forced Entry of Account Codes is not applicable to Personal CO lines.

Remote Access

If a Remote Access call terminates in the system or tandems through the system, the call record stores the attendant ID number (DAC), if assigned, for the remote access group. The caller can dial a CDR account code after the remote access code. Remote access may require a dialed barrier code or authorization code. The call record stores an authorization code if dialed.

Class of service 31 is used for remote access calls to set FEAC (Forced Entry of Account Codes) on an extension class of service basis.

Route Advance

FEAC (Forced Entry of Account Codes) applies to trunk groups in a route advance pattern, as well as to the primary trunk group. If the caller does not dial an account code, trunk groups that require an account code are skipped.

Serial Calls

The call record shows each call in a serial call, including call duration, separately.

Tenant Services

A data item is provided to indicate the extension partition of the extension originating a call. If the caller is transferred to one or more additional extensions, it is the final, or terminating extension partition which is recorded in this field. For an unpartitioned switch, this field contains an "A" (hexadecimal).

Through Dialing

Whenever the attendant allows Through Dialing, the call record shows the call as an attendant-assisted call. If an account code is needed, the attendant must dial the account code before dialing the trunk-group access code.

Transfer

On a transferred call, the CDR shows the last extension connected as the called number.

Trunk-to-Trunk Connections

Incoming calls from a tie trunk that are routed through the switch to an outgoing trunk do not require an account code under any circumstances.

Trunk Verification

The use of trunk verification by either the attendant or the maintenance terminal(s) never requires an account code. That is, Trunk Verification is exempt from FEAC.

Unattended Console Service—Call Answer Any Voice Terminal

While CAAVT (Call Answer From Any Voice Terminal) is active, calls answered by dialing the CAAVT access code are extended using the 3-Party Conference and Transfer features. Therefore, Forced Entry of Account Codes applies to these calls according to the options assigned to the answering "night" terminal.

Unattended Console Service—Preselected Call Routing

While Preselected Call Routing is active, calls routed to a preselected voice terminal are extended using the 3-Party Conference and Transfer features. Therefore, Forced Entry of Account Codes applies to these calls according to the options assigned to the preselected voice terminal.

WCR (World Class Routing)

The CDR feature and the WCR feature are compatible with each other; however, some functions work differently than with the earlier networking features (AAR and ARS).

FEAC (Forced Entry of Account Codes)

The FEAC function was originally designed for use with the earlier networking feature, ARS. With ARS, the account code is entered before the network dial access code, and only the length of the account code entered is checked. With the WCR feature, string identifiers can be administered for account code prefixes in the digit analysis module. Also with WCR, account code prefixes are not necessarily limited to the first digit. This provides the ability to verify not only the account code length, but also the specific account code prefix digits entered.

With DEFINITY Generic 2.2, FEAC is supported on a network basis and is assigned using Procedure 312, Word 1. Three options are provided:

- Encode 0 = account code not required.
- Encode 1 = account code is required and must be entered using the CDR DAC, prior to dialing the WCR network DAC.
- Encode 2 = account code is required and is entered either before or after dialing the WCR network DAC.

Of the above options, encodes 0 and 2 can be used when the WCR feature verifies account code prefixes. With encode 0, FEAC is not active and has no bearing on operations in the WCR digit analysis module. With encode 2, FEAC is active and an account code is entered either before or after the WCR network DAC. If the account code is entered before the WCR network DAC, it is not checked against string identifiers administered for the network.

On calls using the IXC (Interexchange Carrier) Access feature, the CIC (Carrier Identification Code), the digits following the 10- or 101- prefix, is recorded in the CDR according to the following rules:

- If an IXC code is sent, the corresponding CIC is recorded.
- If an IXC code is not sent, the CIC value administered for the preference (in fields 8 through 11 of Procedure 321, Word 1) is recorded.

Calls routed by the WCR feature undergo digit analysis. The WCR feature can change the dialed address digits during digit analysis. If this happens, the changed digits appear in the *Dialed Number/Routed Number (Encode 7)* data item, rather than the originally dialed digits. The WCR feature can also change the routed number digits after digit analysis during digit sending. If this occurs, these changed digits appear in the *WCR Number Sent (Encode 37)* data item. If the WCR feature changes the network to route an Extension Number Portability call, the DAC of the "changed to network" is recorded in the *trunk or WCR network access code dialed* data element (encode 5).

Hardware Requirements

The three CDR configurations (VFCDR, SMDR, and CMDR) all require unique hardware. The following three sections outline the necessary hardware for each configuration.

SMDR Hardware Requirements

The following two sections describe the internal (required) and external hardware available for an SMDR configuration.

Switch Hardware Required for SMDR Interface

- Data Channel Circuit Pack, TN403. Circuit 15, slot 23, carrier 0 is specifically designated as the direct output port for SMDR

External Hardware Used With SMDR Interface

- Direct Output—J59209A
  - — Direct output earner and required circuit packs
  - — Direct output cabinet
  - — 207B power supply
- 9-Track Tape Unit—J58886H
  - — 9-track tape carrier and required circuit packs
  - — 9-track cabinet
  - — 207B power supply
  - — KS-22077 9-track tape drive
  - — KS-22078 tape formatter unit
- Supplementary Equipment
  - — COMM-Stor II Communications Storage Unit
  - — TELESEER SMDR
  - — TELETYPE® Model 4310 AAC teleprinter or equivalent.

A more detailed description of the hardware needed for the SMDR configuration is provided by the ***Call Detail Acquisition and Processing Reference*** (555-006-202) manual.

CMDR Hardware Requirements

The following two sections describe the internal and external hardware required for a minimum CMDR configuration.

Switch Hardware Required for CMDR Interface

- Data Channel Circuit Pack, TN403. Up to eight circuits (seven if SMDR is also used). Circuits 14 and 15, Carrier 0, Slots 23 to 26 can be used for CMDR Interface connections. If SMDR is also used on the switch, Circuit 15, Slot 23 is reserved for that purpose.

External Hardware used for CMDR Interface

- 94A Local Storage Unit
- 93B or 3B2 CDRP

A more detailed description of the hardware needed for the CMDR Interface is provided by ***Centralized Message Detail Recording (CMDR), Descriptive Information, 93B Customer Premises System (CPS),*** (190-403-100).

VFCDR Hardware Requirements

The VFCDR interface requires external as well as internal hardware to complete a VFCDR configuration.

Switch Hardware Required for VFCDR Interface

- The TN474B PCC circuit pack supports VFCDR.

External Hardware Used for VFCDR Interface

The 3B2 CDRU requires different hardware than the 6386 CDRU. See the ***Call Detail Recording Utility*** (555-006-211) manual for details.

Feature Administration

The administration procedures used for CDR configurations can vary based on customer needs. The three CDR configurations (VFCDR, SMDR, and CMDR) available on the switch are administered differently. Refer to the "CDR Configuration Administration" section for the details.

FEAC (Forced Entry of Account Codes) Administration

The FEAC function, as originally developed in Release 2, Version 3, forces callers to enter an account code for selected calls (or to use selected calling routes). FEAC is useful for billing communications services to accounts (for example, a law firm that wants to allocate phone services to a specific client's account). FEAC can also be used to control access to selected calling routes.

The FEAC function can work somewhat differently, and is administered differently, depending on the switch and network routing feature being used.

System 85 and DEFINITY Generic 2.1 Switches

For System 85 and DEFINITY Generic 2.1 switches, use the AAR and ARS features for network routing. With these features, the FEAC call accounting function is administrable on three bases.

Extension Class of Service

When FEAC is assigned on an extension class of service (Procedure 010 Word 2) basis, the terminal user is required to enter an account code directly after any of the following toll call conditions:

- The first digit dialed is a one
- An area code is dialed
- An office code that administered as a toll office code is dialed.

The toll and ARS toll restrictions for extension class of service are assigned in Procedure 010 Word 3. These restrictions set up the distinction between access to ARS toll and ARS nontoll routes (encodes 32 and 33 in Procedure 350 Word 2).

Trunk Group Basis

If a trunk group is administered with FEAC (Procedure 101, Word 1, field 8), and the terminal user dials the trunk-group DAC without an account code being dialed first, the switch returns intercept.

If a trunk group administered with FEAC is in a routing pattern and the caller does not dial an account code, the switch will skip that trunk group as if it wasn't in the pattern. If the caller dials an account code, then dials the networking feature access code, the switch will allow access to that trunk group.

System Basis for ARS Access

If the System Class of Service for ARS Access is administered with FEAC, and the ARS feature access code is dialed without an account code being dialed first, the switch returns intercept. If the user's terminal class of service is assigned for FEAC, and the FEAC is not assigned for ARS on a system basis, and the terminal user places a toll call, the switch returns intercept.

DEFINITY Generic 2.2 Switches

On DEFINITY Generic 2.2 switches, three forms of FEAC are still available. However, with Generic 2.2, FEAC can be assigned on a network basis rather than on a system class of service basis.

Also on Generic 2.2 switches, the WCR feature can use string identifiers to check account codes on a per network basis (to accommodate the earlier AAR account code prefix capability). If string identifiers are assigned for account codes with a specific network, FEAC should either be disabled, or assigned using encode 2 in field 3 of Procedure 312, Word 1, for that network. The earlier form of FEAC (encode 1), requires that the account code be dialed prior to the WCR network DAC.

The Account Code

The Account Code is adjustable from 1 to 15 digits. When the standard FEAC function is in effect, the switch checks to see that the correct number of digits has been entered and then passes them to CDR. If the World Class Routing feature uses string identifiers for account codes, one or more specific digits dialed can be checked. See the User Operations section for specific procedures required to enter the account code.

CDR Configuration Administration

The CDR feature can be administered in three general configurations: SMDR, CMDR, or VFCDR. The following three sections discuss the procedures required to administer each configuration.

SMDR Configuration Administration

This section includes a list of procedures that must be administered to set up the 15-word default format. With the SMDR configuration, call records are sent over an SMDR port (TN403) to an SMDR adjunct.

SMDR is assigned on a per-system, and on a per-trunk group basis within the system.

Procedure Administration

The customer can partially administer SMDR using the SMT (System Management Terminal) or the TCM (Terminal Change Management) when available. Both the SMT and the TCM are older administration vehicles and are not available with newer switches. SMDR can also be administered using the MAAP (Maintenance and Administration Panel), Manager II, and Manager IV.

SMDR requires the administration of the following procedures: 000, Word 1, 101, Word 1, 253, 275, Word 1, 275, Word 3, 288, Word 1, 350 Word 1, 350 Word 2. Optional procedures are 010, Word 2, 285, and 286, Word 1. See the administration tables at the end of this chapter for general information on each of these procedures.

CMDR Configuration Administration

This section includes a list of procedures that must be administered to set up the 18-word default format. The records are sent over the CMDR port(s) (TN403) to a 94A LSU(s). See the administration table at the end of this feature.

The CMDR confirmation is assigned on a per-system basis.

Procedure Administration

On System 85 switches, the CMDR configuration is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer CMDR using the SMT (System Management Terminal) or the TCM (Terminal Change Management) when available.

On DEFINITY Generic 2 switches, CMDR is administered using the DEFINITY Manager II.

This configuration can also be administered using the Manager IV.

The CMDR configuration requires the administration of the following procedures: 000, Word 1, 101, Word 1, 253, 275, Word 1, 275, Word 3, 288, Word 1, 350, Word 1, 350, Word 2. The optional procedures are 010, Word 2, 285, and 286, Word 1. See the administration tables at the end of this feature for general information on each of these procedures.

VFCDR Configuration Administration

This section includes a list of procedures that must be administered to set up a variable or "custom" format. The records are sent over the PCC port to a CDRU or other storage device. The VFCDR configuration is assigned on a per-trunk group basis.

Procedure Administration

The VFCDR configuration requires the administration of the following procedures: 000, Word 1, 101, Word 1, 255, Word 1, 255, Word 2, 275, Word 1, 275, Word 3, 288, Word 1, 350, Word 1, and 350, Word 2. The optional procedures are: 010, Word 2, 285, and 286, Word 1. See the administration tables at the end of this feature for general information on each of these procedures.

> **NOTE:** The two default formats are administered in Procedure 288, Word 1. All other formats are administered item-by-item in Procedure 288, Word 2.

Planning the Layout

The first step in administering a variable format record is to carefully plan the layout of the format.

In planning the layout, first decide whether or not you will be using opcodes. Opcodes are necessary if the SMDR or CMDR/NCOSS is in use, or if a peripheral requires their use. Note that both default formats and the recommended standard 18-word ISDN format include opcodes. This keeps these formats compatible with pre-Release 2, Version 4 hardware configurations using SMDR or NCOSS ports. The variable format and the recommended standard 24-word format (both without opcodes) can only be used with the PCC port.

The most crucial step in planning the layout is to create a complete list of data items with the encode number (data item identifier), start position, and number of cells for each data item (see the example in Table 27-H). Also see Table 27-G and Figure 27–10 for information on the word and cell structure layouts.

Format Word Structure

All CDR formats use the 4-cell, 16-bit word structure shown in Table 27-G.

TABLE 27-G. Call Detail Record Word Structure

Field ID	**Byte 4**	**Byte 3**	**Byte 2**	**Byte 1**
Bits Used	15 14 13 12	11 10 09 08	07 06 05 04	03 02 01 00
Contents	Opcode/Call Data	Call Data	Call Data	Call Data

> **NOTE:** The data items must always be administered the same way if a peripheral (such as a 94A LSU) expects them in a certain position. These are opcodes, call duration, and condition codes. Opcodes, when used, always

occupy the left-most cell position of each word in the record. CDR peripherals, such as the SMDR adjunct and the 94A LSU, require opcodes to identify the start and end of a record, whereas the CDRU does not. The call duration field is four cells in length and occupies the first four available cells in the record (the left-most cell of each word is not available if opcodes are used). The condition code occupies the cell immediately following the call duration field. See Figure 27-5 for information on the (24-word recommended standard format) cell structure layout.

Opcodes

The VFCDR option does not require opcodes. If opcodes are not used, call data can be placed in the opcode field (the first cell of each word). Opcodes are used for hardware control and on-line error checking (to identify the end of a record). Opcodes are not included in the outputs sent to polling devices. The last 12 bits, bits 11 through 0, are arranged in 4-bit cells that contain coded data representing CDR data items, or "fields". The 24-word format and the variable (customized) format, do not require any opcodes, and in this way differ from the other CDR records.

Data Item Encodes

Beginning with System 85, Release 2, Version 4, encodes are used as data item identifiers with VFCDR. This allows the flexibility available through the VFCDR configuration. Each encode identifies a specific data item (for example, encode 30 = node number). The encodes are used for internal (switch) processing and are not transmitted to the CDRU.

It is possible to administer 96 data items to a variable format. The problem in doing this is that it limits the length of each data item to one character (four bits). There are 96 cells (at one character per cell) available in a 24-word variable format. The more data items that are administered in a CDR format, the shorter each data item must be.

For example purposes, assume you are going to format the recommended 18-word ISDN record. You would compile a list of desired data items like that shown in Table 27-H.

To create a VFCDR data item list:

- See Table 27-C for a complete listing of available data items.
- Figures 27-4, 27-6, 27-8, and 27-9 can provide ideas on structuring your own variable format record.
- Use Figure 27-10 to locate the starting and ending cell numbers for each data item.

TABLE 27-H. Sample List of Data Items to Plan a CDR Record Layout With Encodes

Encode	CDR Data Items	Number of Cells	Starting Cell Number
1	Call Duration Hours	1	2
2	Call Duration Minutes	2	3
3	Call Duration 10ths/Minute	1	6
4	Condition Code	1	7
19	Interexchange Carrier (IXC)	3	8
6	Trunk-Group Access Code Dialed/Used	3	12
7	Dialed Number	15	16
8	Calling Number*	4	36*
9	Account Code	5	42
10	Authorization Code	7	48
25	Incoming Dial Access Code*	4	36*
28	ISDN Network Service	3	58
13	Calling Extension 5th Digit	1	62
14	Incoming Circuit ID	2	63
15	Feature Flags	1	66
16	Outgoing Circuit ID	2	67
17	Outgoing Circuit ID, 3rd Digit	1	70
18	Incoming Circuit ID, 3rd Digit	1	71
12	Facilities Restriction Level	1	72

* Typically in 15- and 18-word formats, the Calling Number and the Incoming Dial Access Code are in the same position (cells 36-39).

4 Bits per Cell, 4 Cells per Word

Words	15 14 13 12	11 10 09 08	07 06 05 04	03 02 01 00
1	1	2	3	4
2	5	6	7	8
3	9	10	11	12
4	13	14	15	16
5	17	18	19	20
6	21	22	23	24
7	25	26	27	28
8	29	30	31	32
9	33	34	35	36
10	37	38	39	40
11	41	42	43	44
12	45	46	47	48
13	49	50	51	52
14	53	54	55	56
15	57	58	59	60
16	61	62	63	64
17	65	66	67	68
18	69	70	71	72
19	73	74	75	76
20	77	78	79	80
21	81	82	83	84
22	85	86	87	88
23	89	90	91	92
24	93	94	95	96

NOTE: To compose an 18-word format, disregard the lower portion of the figure, starting with Word 19 (Cell 73). For a 15-word format, disregard Words 16 through 24.

Figure 27-10. Cell Structure for a 24-Word Format

The following are the applicable MAAP procedures.

ADMINISTRATION PROCEDURES CALL DETAIL RECORDING			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service and equipment location to an extension number.	Yes
010	2	Assigns Forced Entry of Account Codes to an extension class of service.	Yes
101	1	Activates CDR and FEAC on a trunk-group basis (field 8). The applicable encodes are as follows: 0 = Not active. 1 = CDR active, 2 = Both CDR and FEAC active (account code required). Also sets variable timer, 1-99.	No
253	1	Administers unit type and equipment location for each data channel assignment (except the PCC). This procedure is display only for the PCC.	No
255	1	Assigns the CDR application to a PCC equipment location. Also administers attributes for the assigned application, such as data rate (baud), parity check, data type, etc.	No
255	2	Assigns additional attributes to applications (such as, CDR) and displays mismatches, if any, between switch translation and circuit (TN474B) translation.	No
275	1	Activates and deactivates the CDR feature. Sets FEAC on a system basis. Field 13 sets the account code length (1 through 15). Activates CDR on outgoing calls only (Field 14 = 0), and active on both incoming and outgoing calls (Field 14 = 1).	Yes
275	3	Sets the system variable timer, node number, and CDR Blockage.	Yes
285	1	Sets the Account Code prefix. Valid encodes for the account prefix and reserved digit fields (6 and 7), other than zero, cannot be the same. Also enables Authorization Codes.	No
286	1	Activates CDR for ineffective attempts at accessing a trunk. There are some limitations. 0 = Ineffective Attempts are not recorded by CDR 1 = Ineffective Attempts are recorded by CDR.	Yes

(Continued)

ADMINISTRATION PROCEDURES CALL DETAIL RECORDING *(Continued)*			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
288	1	Performs the *Remove-Execute* function to change the CDR record length, the opcode indicator, and the field indicating variable or standard format. Also specifies record length for CDR ports. The fields are administered as follows: 1. A value from 15 to 24. 2. A value of 0 or 1. A 1 indicates opcodes in the left-most 4 bits of each word. A 0 indicates these bit positions are available for data items. A 0 is valid only when variable format is being administered in the next field. 3. A value of 0 or 1. A 0 indicates a standard 15- or 18-word format (valid only if Field 1 is administered as 15 or 18). A 1 indicates variable format (valid any time). 4. A value of 15 or 18 replaces the CDR port record length indicator.	Yes
288	2	Allows the formatting of CDR records (to a maximum length of 24 words) by administering the starting position and field length for a data item encode. Only administrable if the CDR feature is deactivated in Procedure 275, Word 1. The fields are administered as follows: 1. Specifies the encode of the data item to be administered (cannot be 0). Must be administered before Field 2 or 3 can be administered. 2. Starting cell number of the data item encode specified in Field 1. 3. A value up to 31. Indicates the field length for the data item encode specified in Field 1.	Yes
312	1	For Generic 2.2 switches, assigns FEAC on a network basis (field 3). The following encodes apply: 0 = Account code not required 1 = Account code required, and must be dialed before the WCR network DAC 2 = Account code required, and can be dialed after the WCR network DAC.	N/A

(Continued)

ADMINISTRATION PROCEDURES CALL DETAIL RECORDING *(Continued)*			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
350	1	Assigns the first digit of the feature dial access codes.	No
350	2	Assigns the CDR dial access codes. The applicable encodes are as follows: 36 CDR Account Number 37 CDR Start 38 CDR Stop.	No

Notes:

Call Forwarding — Busy and Don't Answer

Description

The Call Forwarding—Busy and Don't Answer feature redirects calls to one extension number and sends them to another specified extension number or to the attendant queue.

When this feature is active, calls are forwarded whenever the called terminal is busy or the user does not answer the call within a specified period (number of ringing cycles). The user selects the forwarding destination when activating the feature. The Call Forwarding—Busy and Don't Answer feature is activated or canceled by either the voice terminal user or an attendant.

Call Forwarding—Busy and Don't Answer provides a simple form of local call redirection for calls that might otherwise go unanswered.

The don't answer interval is flexible and may be chosen by the customer. To suit the needs of an individual switch, the don't answer interval can be set from one to eight ring cycles. The designated interval will be the same for every terminal within the switch.

Feature History and Development

The Call Forwarding—Busy and Don't Answer feature was first available on System 85 in Release 1. There have been no changes to his feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Activate Call Forwarding to a Voice Terminal

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Busy and Don't Answer access code,

   or

   Press the **[CALL FORWARD BUSY/DON'T ANS]** button. [Second dial tone]
3. Dial the destination extension number. [Confirmation tone]
4. Go on-hook.

From an attendant console for a voice terminal:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights].
2. Press **[START]** . [Dial tone]
3. Dial the Call Forwarding—Busy and Don't Answer access code. [Second dial tone]
4. Dial the extension number of the forwarding terminal,

   or

   Press the appropriate DXS button. [Third dial tone]
5. Dial the forwarding destination extension number,

   or

   Press the appropriate DXS button. [Confirmation tone]
6. Press **[RELEASE].** [ATND lamp goes out, and PA lamp lights].

To Activate Call Forwarding to the Attendant Queue

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Busy and Don't Answer access code,

   or

   Press the **[CALL FORWARD BUSY/DON'T ANS]** button. [Second dial tone]
3. Dial the attendant access code, or dial an LDN (listed directory number). [Confirmation tone]
4. Go on-hook.

To Cancel Call Forwarding

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Cancel access code,

   or

   Press the **[CALL FORWARD BUSY/DON'T ANS]** button. [Confirmation tone].
3. Go on-hook.

From an attendant console for a voice terminal:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights].
2. Press **[START]** . [Dial tone]

3. Dial the Call Forwarding—Cancel access code. [Second dial tone]
4. Dial the extension number of the forwarding terminal. [Confirmation tone]
5. Press **[RELEASE]** . [ATND lamp goes out, and PA lamp lights].

To Activate Call Forwarding While on a 2-Way Connection

Using a single-appearance terminal:

1. Momentarily press the switchhook,

   or

   Press **[RECALL]** . [Recall dial tone is heard, and the second party put on soft hold].
2. Dial the Call Forwarding—Busy and Don't Answer access code. [Second dial tone]
3. Dial the forwarding destination extension number. [Confirmation tone]
4. Momentarily press the switchhook,

   or

   Press **[RECALL]** . [Reconnected to the second party].

Using a multiappearance terminal:

1. Press **[CONFERENCE]** or **[TRANSFER]** . [Dial tone is heard, and the second party put on hold].
2. Dial the Call Forwarding—Busy and Don't Answer access code,

   or

   Press the **[CALL FORWARD BUSY/DON'T ANS]** button. [Second dial tone]
3. Dial the forwarding destination extension number. [Confirmation tone]
4. Select the held appearance. [Held party is reconnected].

To Cancel Call Forwarding While on a 2-Way Connection

Using a single-appearance terminal:

1. Momentarily press the switchhook,

   or

   Press **[RECALL]** . [Recall dial tone is heard, and second party put on soft hold].
2. Dial the Call Forwarding—Cancel access code. [Confirmation tone]
3. Momentarily press the switchhook,

   or

   Press **[RECALL]** . [Reconnected to the second party].

Using a multiappearance terminal:

1. Press the **[CONFERENCE]** or **[TRANSFER]** button. [Dial tone is heard, and second party is put on hold].
2. Dial the Call Forwarding—Cancel access code,

   or

   Press the **[CALL FORWARD BUSY/DON'T ANS]** button. [Confirmation tone]
3. Select the held appearance. [Held party is reconnected].

Considerations

Limitations

Call Forwarding—Busy and Don't Answer can only be used within the local switch. Even in a DSC environment, this feature cannot be used to forward calls to a distant network node.

Forwarding to an Attendant

Call Forwarding—Busy and Don't Answer can be used to forward calls to the attendant queue, but forwarding to a ***selected attendant*** is not allowed.

Two Types of Forwarding Not Allowed

Call Forwarding—Busy and Don't Answer and Call Forwarding—Follow Me cannot be activated at the same time on the same voice terminal. The switch returns intercept tone when this is attempted.

Two Destinations Not Allowed

The terminal user must always cancel the previous forwarding destination before establishing a new destination for forwarding. Without previous cancellation, the switch returns intercept tone.

Double Forwarding Not Allowed

Terminal A may activate forwarding to terminal B, and terminal B may activate forwarding to terminal C. However, this does not imply that terminal A's calls will forward to terminal C. Voice terminal calls forward only once.

Mutually Exclusive Features

Call Forwarding—Busy and Don't Answer and Call Forwarding—Don't Answer cannot be assigned to the same class of service. Therefore, a voice terminal user can have access to either one of these two features, but not both. Both features are activated using the same dial access code (Encode 2) and using the same feature button (Encode 4). When either method of activation is used, the feature accessed depends on the class-of-service entry in

Procedure 010, Word 1, Field 4: "1" defines Call Forwarding—Busy and Don't Answer, and "2" defines Call Forwarding—Don't Answer.

Hard and Soft Processor Swaps

Call Forwarding relationships are stored in a translation portion of switch memory. Therefore, if Call Forwarding is activated to forward calls to a local extension number and then a hard processor swap occurs, the forwarding relationship will endure after the hard swap is finished.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

When Call Forwarding—Busy and Don't Answer is active at the called voice terminal, the forwarding operation occurs before Attendant Call Waiting is allowed. There are four possible operations. If the originally called voice terminal is busy and the forwarded-to voice terminal is idle, attendant calls forward to and ring at the forwarded-to voice terminal. If the originally called voice terminal is busy and the forwarded-to voice terminal is busy, Attendant Call Waiting is denied and busy tone is returned. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is idle, attendant calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, attendant calls continue ringing at the originally called voice terminal.

AUDIX (Audio Information Exchange)

The Call Forwarding—Busy and Don't Answer feature cannot be used to forward calls to the AUDIX extension number. When this is attempted, the switch returns intercept tone.

ACD (Automatic Call Distribution)

When an ACD split supervisor activates Call Forwarding—Busy and Don't Answer (Intraflow—Threshold), calls are forwarded to a local destination in an overflow condition. The Don't Answer portion of this feature does not apply.

The split supervisor cannot use call forwarding to forward calls to the supervisor's individual extension number. Activation of either feature only forwards calls which are directed to the split.

For ACD calls, there can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destination is an attendant or an ACD split (without the inflow level specified for the split), the remaining priority destinations would not be checked. Instead, a forwarded call would unconditionally enter the attendant's or the split's queue.

When an ACD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split.

Automatic Callback

Call Forwarding—Busy and Don't Answer has no effect on an Automatic Callback call origination. Callback always directs to the originating terminal, not to the forwarded-to terminal.

Busy Verification of Lines

Busy verification is allowed toward an extension even if the extension has Call Fowarding—Busy and Don't Answer active.

Call Coverage

Call Forwarding—Busy and Don't Answer takes precedence over Call Coverage in the following situations.

The following list shows the alternate treatments for cases where a covering user has Call Forwarding—Busy and Don't Answer active. These specified treatments are based on the assumption that the alternate voice terminal has at least one idle appearance.

- If Call Forwarding—Busy and Don't Answer is active for a principal who has coverage, calls to that principal's extension redirect according to call forwarding.
- If a coverage point (other than an ACD split) has Call Forwarding—Busy and Don't Answer active, the point is not eligible to receive a coverage call.
  - If there is only one coverage point, the principal's voice terminal rings.
  - If the coverage point is the final (not the only) point, the previous coverage point rings.
  - If there is a subsequent coverage point in the path, the forwarded coverage point is skipped.

If a situation arises where the alternate voice terminal(s) does not have an idle appearance, the switch returns busy tone to the calling party.

When a call is forwarded to a principal with coverage active, the forwarded call does not direct to coverage. If the principal doesn't answer, the forwarded call will ring (until abandoned) at the principal's voice terminal. However, if every appearance of the principal's voice terminal is busy, the switch either continues ringing at the forwarding terminal in a don't answer condition or returns busy tone to the calling party if the forwarding terminal is busy.

Also, if Call Forwarding—Busy and Don't Answer is active for a group coverage point (ACD split), calls will cover to the split's queue and then intraflow according to call forwarding.

CDR (Call Detail Recording)

The extension that a call forwards to is the extension number that CDR records in the called number field.

Call Vectoring

The Call Forwarding—Busy and Don't Answer feature cannot be used to forward calls to a VDN (Vector Directory Number).

With Call Vectoring assigned, an ACD split supervisor can use this feature to forward calls for the supervisor's individual extension. Without Call Vectoring assigned, Call Forwarding—Busy and Don't Answer is instead used to forward calls which are directed to the split's queue. The destination for a "route to extension" step in a vector can be an extension with Call Forwarding—Busy and Don't Answer activated. When a VDN call routes to a forwarded extension, the VDN call will forward to and ring the forwarded-to extension if the forwarding extension is ***busy.*** When there is ***no answer*** at the forwarding extension, the VDN call will continue to ring the forwarding extension and will not forward.

Call Waiting

When Call Forwarding—Busy and Don't Answer is active at the called voice terminal, the forwarding operation occurs before Call Waiting is allowed. There are four possible operations. If the originally called voice terminal is busy and the forwarded-to voice terminal is idle, calls forward to and ring at the forwarded-to voice terminal. If the originally called voice terminal is busy and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is idle, calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal.

Data Call Setup

Call Forwarding—Busy and Don't Answer can be used by voice terminal and data terminal users. For data terminals, this type of forwarding can serve as a substitute for Call Coverage.

DCS (Distributed Communications System)

The Call Forwarding—Busy and Don't Answer feature is not transparent in the DSC environment. The forwarded-to extension in a call forwarding relationship cannot reside in a different DSC node.

EUCD (Enhanced Uniform Call Distribution)

When an EUCD split supervisor activates Call Forwarding—Busy and Don't Answer (Intraflow—Threshold), calls are forwarded to a local destination in an overflow condition. The Don't Answer portion of this feature does not apply.

The split supervisor cannot use call forwarding to forward calls to the supervisor's individual extension number. Activation of either feature only forwards calls which are directed to the split.

For EUCD calls, there can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destination is an attendant or an EUCD split (without the inflow level specified for the split), the remaining priority destinations would not be checked. Instead, a forwarded call would unconditionally enter the attendant's or the split's queue.

When an EUCD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split.

Extension Number Portability

Call Forwarding—Busy and Don't Answer cannot be used to forward calls outside the switch. Thus, an extension number that has been ported to a new node cannot be a forwarded-to extension.

Hold

Hard hold ringbacks for single-appearance voice terminals are not forwarded. Instead, the held call rings back the terminal that put the call on hold when the user goes on-hook.

Hunting

When a call forwards to a terminal in a hunt group, another terminal in the hunt group may receive the call rather than the designated terminal (hunting occurs).

When an incoming call reaches a terminal in a hunt group with Call Forwarding—Busy and Don't Answer active, the call forwards to the designated terminal (call forwarding takes precedence).

IPA (Interpartition Access)

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition group or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition group.

Last Extension Dialed

Call Fowarding can be used in conjunction with Last Extension Dialed. This is useful when activation of the Call Forwarding—Busy and Don't Answer feature toward the last extension dialed is desired. (Dial the Call Forwarding—Busy and Don't Answer DAC or press the feature button before pressing the LXD feature button).

Last Number Dialed

Call Forwarding can be used in conjunction with Last Number Dialed. This is useful when activation of the Call Forwarding—Busy and Don't Answer feature toward the last number dialed is desired. (Dial the Call Forwarding—Busy and Don't Answer DAC or press the feature button before pressing the LND feature button).

Leave Word Calling

Within the local switch, Leave Word Calling messages direct to the principal originally called even when the calls redirect via Call Forwarding—Busy and Don't Answer. The only exception to this is when calls are redirected to the attendant. Leave Word Calling is not allowed when a call is redirected to the attendant.

Line Lockout

A call directed toward a locked-out voice terminal with Call Forwarding active will forward to the designated terminal.

Look-Ahead Interflow

At either a sending or receiving switch, the Call Forwarding—Busy and Don't Answer feature cannot be used to forward calls to a VDN.

Override

When a called terminal has activated Call Forwarding—Busy and Don't Answer, an override call does not forward. Three-burst ringing is provided for an idle forwarding terminal, and the override call enters the conversation of a busy forwarding terminal.

Priority Calling

When Call Forwarding is in effect, the forwarding operation occurs before Priority Calling is allowed. There are four possible operations. If the originally called voice terminal is busy and the forwarded-to voice terminal is idle, calls forward to and ring at the forwarded-to voice terminal. If the originally called voice terminal is busy and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is idle, calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal.

Restriction—Attendant Control of Voice Terminals

Call Forwarding—Busy and Don't Answer cannot be activated to forward calls to a terminal that is ***already restricted*** by Controlled Terminal-to-Terminal, Controlled Termination, or Controlled Total restriction. When this is attempted the switch returns intercept tone.

If calls are ***already being forwarded*** to a voice terminal before an attendant activates a termination restriction against the forwarded-to terminal, these forwarded calls are allowed to terminate to the restricted terminal.

If a voice terminal has forwarding activated, and then an attendant activates a termination restriction against the forwarding terminal, calls to the forwarding terminal do not terminate or forward. The switch returns intercept tone to the calling party.

Restriction—Voice Terminal Restrictions

Calls may not forward to a voice terminal with Restriction—Voice Terminal Restrictions (Termination, Manual Terminating Line, or Terminal-to-Terminal Calling) activated.

Incoming calls on public or private network trunks may not forward to an Inward restricted terminal.

Call Forwarding—Busy and Don't Answer functions normally for an Origination restricted voice terminal when the attendant or a single-line voice terminal user activates the Call Forwarding—Busy and Don't Answer feature from a hold or recall dial tone state. The Origination restricted terminal is not allowed to activate Call Forwarding—Busy and Don't Answer from an idle state.

If an urestricted voice terminal is assigned as the forwarded-to voice terminal and then Restriction—Voice Terminal Restrictions (Inward, Termination, Terminal-to-Terminal Only, or Manual Terminating Line) restriction is activated in its extension class of service, calls will still forward to the voice terminal.

Ringing—Abbreviated and Delayed

Call Forwarding—Busy and Don't Answer takes precedence over Abbreviated and Delayed Ringing when the amount of ringing cycles used to time both features are equal.

The details of the don't answer condition are as follows. If the timing interval for call forwarding ***is less than or equal to*** the timing internal for Abbreviated and Delayed Ringing, terminating calls forward to the destination extension without ringing the image(s) assigned delayed ringing. However, if the timing interval for call forwarding ***is greater than*** the timing interval for Abbreviated and Delayed Ringing, terminating calls first ring the abbreviated ringing image(s). Then ringing transfers to the delayed ringing image(s), and this image rings for the rest of the call forwarding timing interval. After the call forwarding timing interval elapses, the call forwards to the destination extension.

Tenant Services

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition.

Timed Reminder

If the attendant presses the STA ID button after the extended call forwards, the attendant display indicates the extension number that the call forwarded to rather than the originally called extension number.

Restricting Feature Use

Voice Terminal Restrictions

The voice terminal restrictions that restrict call forwarding are:

- Termination restriction
- Manual Terminating Line restriction
- Terminal-to-Terminal Only Calling restriction.

Attendant Control of Voice Terminal Restrictions

The attendant can restrict terminals from being used as a destination for forwarding. The restrictions are:

- Termination restriction
- Total restriction
- Terminal-to-Terminal restriction.

The application of Attendant Control of Voice Terminal restrictions takes precedence over the Call Forwarding—Busy and Don't Answer feature only when applied before the Call Forwarding—Busy and Don't Answer feature is activated.

Hardware Requirements

None.

Feature Administration

Assignment of the Call Forwarding—Busy and Don't Answer feature is on a per-terminal class-of-service basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES CALL FORWARDING—BUSY AND DON'T ANSWER			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the voice terminal class of service to an extension number.	Yes
010	1	Assigns Call Forwarding—Busy and Don't Answer to a voice terminal class of service. (Enter "1" in Field 4).	Yes
054	2	Administers the Call Forwarding—Busy and Don't Answer button to a multiappearance voice terminal. The applicable encode is: 4 Call Forwarding—Busy and Don't Answer	Yes
200	1	Specifies the don't answer timing interval (one to eight cycles) for call forwarding.	No
204	1	Designates the desired alphanumeric display for calls forwarded to an attendant. The applicable encode is as follows: R2 V1 to R2 V3: 290 Call Forwarding R2 V4 and later: 2290 Call Forwarding.	No
350	1	Assigns the first digit of the feature dial access codes (if required).	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 2 Call Forwarding—Busy and Don't Answer 3 Call Forwarding—Cancel.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — CALL FORWARDING—BUSY AND DON'T ANSWER	
PATH NAME	**PURPOSE**
terminal-change class-of-service attribute	Assigns Call Forwarding—Busy and Don't Answer to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change terminal buttons	Assigns the Call Forwarding—Busy and Don't Answer button to a multiappearance voice terminal.

Notes:

Call Forwarding — Don't Answer

Description

This feature forwards calls at an extension number to another selected extension number or to the attendant queue. When this feature is active, calls forward whenever the user does not answer the call. The user designates the forwarding destination when activating the feature. Either a voice terminal user or an attendant can activate or cancel the feature.

The feature provides a simple form of local coverage for calls that might otherwise go unanswered.

The don't answer internal is flexible and may be chosen by the customer. To suit the needs of an individual switch, the don't answer interval can be set from one to eight ring cycles. The designated interval will be the same for every terminal within the switch.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Activate Call Forwarding to a Voice Terminal

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Don't Answer access code,

   or

   Press the **[CALL FORWARD DON'T ANSWER]** button. [Second dial tone]
3. Dial the destination extension number. [Confirmation tone]
4. Go on-hook.

From an attendant console for a voice terminal:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights].
2. Press **[START]** . [Dial tone]
3. Dial the Call Forwarding—Don't Answer access code. [Second dial tone]

4. Dial the extension number of the forwarding terminal,

   or

   Press the appropriate DXS button. [Third dial tone]

5. Dial the forwarding destination extension number,

   or

   Press the appropriate DXS button. [Confirmation tone]

6. Press **[RELEASE]** . [ATND lamp goes out, and PA lamp lights].

To Activate Call Forwarding to the Attendant Queue

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Don't Answer access code,

   or

   Press the **[CALL FORWARD DON'T ANSWER]** button. [Second dial tone]

3. Dial the attendant access code,

   or

   Dial an LDN (listed directory number). [Confirmation tone]

4. Go on-hook.

To Cancel Call Forwarding

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Cancel access code,

   or

   Press the **[CALL FORWARD DON'T ANSWER]** button. [Confirmation tone]

3. Go on-hook.

From an attendant console for a voice terminal:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights].
2. Press **[START]** . [Dial tone]
3. Dial the Call Forwarding—Cancel access code. [Second dial tone]

4. Dial the extension number of the forwarding terminal. [Confirmation tone]
5. Press **[RELEASE]** . [ATND lamp goes out, and PA lamp lights].

To Activate Call Forwarding While on a 2-Way Connection

Using a single-appearance terminal:

1. Momentarily press the switchhook,

   or

   Press **[RECALL]** . [Recall dial tone is heard, and the second party put on soft hold].
2. Dial the Call Forwarding—Don't Answer access code. [Second dial tone]
3. Dial the forwarding destination extension number. [Confirmation tone]
4. Momentarily press the switchhook,

   or

   Press **[RECALL]** . [Reconnected to the second party].

Using a multiappearance terminal:

1. Press **[CONFERENCE]** or **[TRANSFER]** . [Dial tone is heard, and the second party put on hold].
2. Dial the Call Forwarding—Don't Answer access code,

   or

   Press the **[CALL FORWARD DON'T ANSWER]** button. [Second dial tone]
3. Dial the forwarding destination extension number. [Confirmation tone]
4. Select the held appearance. [Held party is reconnected].

To Cancel Call Forwarding While on a 2-Way Connection

Using a single-appearance terminal:

1. Momentarily press the switchhook,

   or

   Press **[RECALL]** . [Recall dial tone is heard, and second party placed on soft hold].
2. Dial the Call Forwarding—Cancel access code. [Confirmation tone]
3. Momentarily press the switchhook,

   or

   Press **[RECALL]** . [Reconnected to the second party].

Using a multiappearance terminal:

1. Press the **[CONFERENCE]** or **[TRANSFER]** button. [Dial tone is heard, and second party is put on hold].
2. Dial the Call Forwarding—Cancel access code,

   or

   Press the **[CALL FORWARD DON'T ANSWER]** button. [Confirmation tone]
3. Select the held appearance. [Held party is reconnected].

Considerations

Limitations

Call Forwarding—Don't Answer can only be used within the local switch. Even in a DCS environment, this feature cannot be used to forward calls to a distant network node.

Forwarding to an Attendant

Call Forwarding—Don't Answer can be used to forward calls to the attendant queue, but forwarding to a ***selected attendant*** is not allowed.

Two Types of Forwarding Not Allowed

Call Forwarding—Don't Answer and Call Forwarding—Follow Me cannot be activated at the same time on the same voice terminal. The switch returns intercept.

Two Destinations Not Allowed

The terminal user must always cancel the previous forwarding destination before establishing a new destination for forwarding. Without previous cancellation, the switch returns intercept tone.

Double Forwarding Not Allowed

Terminal A may activate forwarding to terminal B, and terminal B may activate forwarding to terminal C. However, this does not imply that terminal A's calls will forward to terminal C. Voice terminal calls forward only once.

Mutually Exclusive Features

Call Forwarding—Don't Answer and Call Forwarding—Busy and Don't Answer cannot be assigned to the same class of service. Therefore, a voice terminal user can have access to either one of these two features, but not both. Both features are activated using the same dial access code (Encode 2) and using the same feature button (Encode 4). When either method of activation is used, the feature accessed depends on the class-of-service entry in Procedure 010, Word 1, Field 4: "2" defines Call Forwarding—Don't Answer, and "1" defines Call Forwarding—Busy and Don't Answer.

Hard and Soft Processor Swaps

Call Forwarding relationships are stored in a translation portion of switch memory. Therefore, if Call Forwarding is activated to forward calls to a local extension number and then a hard processor swap occurs, the forwarding relationship will endure after the hard swap is finished.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

When Call Forwarding—Don't Answer is active at the called voice terminal, the forwarding operation occurs before Attendant Call Waiting is allowed. There are two possible operations. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is idle, attendant calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, attendant calls continue ringing at the originally called voice terminal.

AUDIX (Audio Information Exchange)

The Call Forwarding—Don't Answer feature cannot be used to forward calls to the AUDIX extension number. When this is attempted, the switch returns intercept tone.

ACD (Automatic Call Distribution)

An ACD split supervisor cannot activate Call Forwarding—Don't Answer to forward the supervisor's calls. When an ACD split supervisor activates Call Forwarding—Don't Answer, the split's calls are forwarded to a local destination in an overflow condition (as if Call Forwarding—Busy and Don't Answer were instead assigned to the supervisor's class of service).

For ACD calls, there can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destination is an attendant or an ACD split (without the inflow level specified for the split), the remaining priority destinations would not be checked. Instead, a forwarded call would unconditionally enter the attendant's or the split's queue.

When an ACD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split.

Automatic Callback

Call Forwarding—Don't Answer has no effect on an Automatic Callback call origination. Callbacks always direct to the originating terminal, not to the forwarded-to terminal.

Busy Verification of Lines

Busy verification is allowed toward a line even if the line has Call Forwarding—Don't Answer active.

Call Coverage

Call Forwarding—Don't Answer takes precedence over Call Coverage in the following situations.

The following list shows the alternate treatments for cases where a covering user has Call Forwarding—Don't Answer active. These specified treatments are based on the assumption that the alternate voice terminal has at least one idle appearance.

- If Call Fowarding—Don't Answer is active for a principal who has coverage, calls to that principal's extension redirect according to call forwarding.
- If a coverage point has Call Forwarding—Don't Answer active, the point is not eligible to receive a coverage call.
  - If there is only one coverage point, the principal's voice terminal rings.
  - If the coverage point is the final (not the only) point, the previous coverage point rings.
  - If there is a subsequent coverage point in the path, the forwarded coverage point is skipped.

If a situation arises where the alternate voice terminal(s) does not have an idle appearance, the switch returns busy tone to the calling party.

When a call is forwarded to a principal with coverage active, the forwarded call does not redirect to coverage. If the principal doesn't answer, the forwarded call will ring (until abandoned) at the principal's voice terminal. However, if every appearance of the principal's voice terminal is busy, the switch continues ringing at the forwarding terminal in the don't answer condition.

CDR (Call Detail Recording)

The extension that a call forwards to is the extension number that CDR records in the called number field.

Call Vectoring

The Call Forwarding—Don't Answer feature cannot be used to forward calls to a VDN (Vector Directory Number).

With Call Vectoring assigned, an ACD split supervisor can use this feature to forward calls directed to the supervisor's individual extension. Without Call Vectoring assigned, Call Forwarding— ***Busy and*** Don't is instead used to forward calls which are directed to the splits queue. (This is true even if Call Forwarding—Don't Answer is assigned to the supervisor's class of service).

The destination for a "route to extension" step in a vector can be an extension with Call Forwarding—Don't Answer activated. When there is no answer at the forwarding extension, the VDN call will continue to ring the forwarding extension and will not forward.

Call Waiting

When Call Forwarding—Don't Answer is active at the called voice terminal, the forwarding operation occurs before Call Waiting is allowed. There are two possible operations. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is idle, calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal.

DCS (Distributed Communications System)

The Call Forwarding—Don't Answer feature is not transparent in the DCS environment. The forwarded-to extension in a call forwarding relationship cannot reside in a different DCS node.

EUCD (Enhanced Uniform Call Distribution)

An EUCD split supervisor cannot activate Call Forwarding—Don't Answer to forward the supervisor's calls. When an EUCD split supervisor activates Call Forwarding—Don't Answer, the split's calls are forwarded to a local destination in an overflow condition (as if Call Forwarding—Busy and Don't Answer were instead assigned to the supervisor's class of service).

For EUCD calls, there can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destination is an attendant or an EUCD split (without the inflow level specified for the split), the remaining priority destinations would not be checked. Instead, a forwarded call would unconditionally enter the attendant's or the split's queue.

When an EUCD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split.

Extension Number Portability

Call Forwarding—Don't Answer cannot be used to forward calls outside the switch. Thus, an extension number that has been ported to a new node cannot be a forwarded-to extension.

Hold

Hard hold ringbacks for single-appearance voice terminals are not forwarded. Instead, the held call rings back the terminal that placed the call on hold when the user goes on-hook.

Hunting

When a call forwards to a terminal in a hunt group, another terminal in the hunt group may receive the call rather than the designated terminal (hunting occurs).

When an incoming call reaches a terminal in a hunt group with Call Forwarding—Don't Answer active, the call forwards to the designated terminal if the call is not answered. When the forwarding terminal is busy, the call will hunt.

IPA (Interpartition Access)

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition group or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition group.

Last Extension Dialed

Call Forwarding can be used in conjunction with Last Extension Dialed. This is useful when activation of the Call Forwarding—Don't Answer feature toward the last extension dialed is desired. (Dial the Call Forwarding—Don't Answer DAC or press the feature button before pressing the LXD feature button).

Last Number Dialed

Call Forwarding can be used in conjunction with Last Number Dialed. This is useful when activation of the Call Forwarding—Don't Answer feature toward the last number dialed is desired. (Dial the Call Forwarding—Don't Answer DAC or press the feature button before pressing the LND feature button).

Leave Word Calling

Within the local switch, Leave Word Calling messages direct to the principal originally called even when the calls redirect via Call Forwarding—Don't Answer. The only exception to this is calls redirected to the attendant. Leave Word Calling is not allowed when a call is redirected to the attendant.

Line Lockout

A call directed toward a locked-out voice terminal with Call Forwading—Don't Answer active will not forward. Instead, the switch returns busy tone to the calling party.

Look-Ahead Interflow

At either a sending or receiving switch, the Call Forwarding—Don't Answer feature cannot be used to forward calls to a VDN.

Override

When a called terminal has activated Call Forwading—Don't Answer, an override call does not forward. Three-burst ringing is provided for the idle forwarding terminal.

Priority Calling

When Call Forwarding is in effect, the forwarding operation occurs before Priority Calling is allowed. There are two possible operations. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is idle, calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal.

Restriction—Attendant Control of Voice Terminals

Call Forwarding—Don't Answer cannot be activated to forward calls to a terminal that ***is already restricted*** by Controlled Terminal-to-Terminal, Controlled Termination, or Controlled Total restriction. When this is attempted, the switch returns intercept tone.

If calls are ***already being forwarded*** to a voice terminal before an attendant activates a termination restriction against the forwarded-to terminal, these forwarded calls are allowed to terminate to the restricted terminal.

If a voice terminal has forwarding activated, and then an attendant activates a termination restriction against the forwarding terminal, calls to the forwarding terminal do not terminate or forward. The switch returns intercept tone to the calling party.

Restriction—Voice Terminal Restrictions

Calls may not forward to a voice terminal with Restriction—Voice Terminal Restrictions (Termination, Manual Terminating Line, or Terminal-toTerminal Calling) activated.

Incoming calls on public or private network trunks may not forward to an Inward restricted terminal.

Call Forwarding—Don't Answer functions normally for an Origination restricted voice terminal when the attendant or a single-line voice terminal user activates the Call Forwarding—Don't Answer feature from a hold or recall dial tone state. The Origination restricted terminal is not allowed to activate Call Forwadng—Don't Answer from an idle state.

If an unrestricted voice terminal is selected as the forwarded-to voice terminal and then restricted by the Restriction—Voice Terminal Restrictions feature (class of service) in a way that would normally block the forwarded call, calls will still forward (forwarding takes precedence).

Ringing—Abbreviated and Delayed

Call Forwarding—Don't Answer takes precedence over Abbreviated and Delayed Ringing when the number of ringing cycles used to time both features is equal.

The details are as follows. If the timing interval for call forwarding ***is less than or equal to*** the timing interval for Abbreviated and Delayed Ringing, terminating calls forward to the destination extension without ringing the image(s) assigned delayed ringing.

However, if the timing interval for call forwarding ***is greater than*** the timing interval for Abbreviated and Delayed Ringing, terminating calls first ring the abbreviated ringing image(s). Then ringing transfers to the delayed ringing image(s), and these images ring for the rest of the call forwarding timing interval. After the call forwarding timing interval elapses, the call forwards to the destination extension.

Tenant Services

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition.

Timed Reminder

If the attendant presses the STA ID button after the extended call forwards, the attendant display indicates the extension number that the call forwarded to rather than the originally called extension number.

Restricting Feature Use

Voice Terminal Restrictions

The voice terminal restrictions that restrict call forwarding are:

- Termination restriction
- Manual Terminating Line restriction
- Terminal-to-Terminal Only Calling restriction.

Attendant Control of Voice Terminal Restrictions

The attendant can restrict terminals from being used as a destination for forwarding. The restrictions are:

- Termination restriction
- Total restriction
- Terminal-to-Terminal restriction.

The application of Attendant Control of Voice Terminal restrictions takes precedence over the Call Forwarding—Don't Answer feature only when applied before the Call Forwarding— Don't Answer feature is activated.

Hardware Requirements

None.

Feature Administration

Assignment of the Call Forwarding—Don't Answer feature is on a per-terminal class-of-service basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic *2* switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
CALL FORWARDING—DON'T ANSWER

PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the voice terminal class of service to an extension number.	Yes
010	1	Assigns Call Forwarding—Don't Answer to a voice terminal class of service. (Enter "2" in Field 4).	Yes
054	2	Administers the Call Forwarding—Don't Answer button to a multiappearance voice terminal. The applicable encode is: 4 Call Forwarding—Busy and Don't Answer	Yes
200	1	Specifies the don't answer timing interval (one to eight cycles) for call forwarding.	No
204	1	Designates the desired alphanumeric display for calls forwarded to an attendant. The applicable encode is as follows: R2 V1 to R2 V3: 290 Call Forwarding R2 V4 and later: 2290 Call Forwarding.	No
350	1	Assigns the first digit of the feature dial access codes (if required).	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 2 Call Forwarding—Busy and Don't Answer 3 Call Forwarding—Cancel.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — CALL FORWARDING—DON'T ANSWER	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns Call Forwarding—Don't Answer to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change terminal buttons	Assigns the Call Forwading—Don't Answer button to a multiappearance voice terminal.

Call Forwarding — Follow Me

Description

This feature forwards all calls directed toward a given extension number to another selected extension number, to the attendant queue, or to a telephone in the public network. Either the terminal user or an attendant can activate or cancel this feature.

When a call is placed to an extension with Call Forwarding—Follow Me active, ring ping [a quick burst of ringing (0.1 seconds)] is provided at the forwarding terminal. If the forwarding terminal is a data terminal, the display message "FORWARDED" and the ASCII bell ring character are sent to the terminal. This is to remind the user that the terminal is still in the forwarding mode.

Forwarding calls to another extension is useful when the user will be at the forwarded-to voice terminal, or the forwarded-to voice terminal user will be there. Forwarding calls to another extension also reduces attendant work load.

Feature History and Development

This feature was first available on System 85 in Release 1. Call Forwarding—Off Net was first available in the Release 2, Version 1 software package. This function allows users to forward all calls to a telephone in the nontoll public network.

User Operations

The following are the user operating procedures for this feature.

To Activate Call Forwarding—Follow Me to a Voice Terminal

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Follow Me access code,

   or

   Press the **[CALL FORWARD FOLLOW ME]** button. [Second dial tone]
3. Dial the destination extension number. [Confirmation tone]
4. Go on-hook.

From an attendant console for a voice terminal:

1. Press an idle loop button. [PA lamp goes out, and the ATND lamp lights].
2. Press **[START]** . [Dial tone]

3. Dial the Call Forwarding—Follow Me access code. [Second dial tone]
4. Dial the extension number of the forwarding terminal,

   or

   Press the appropriate DXS button. [Third dial tone]
5. Dial the extension number of the destination terminal,

   or

   Press the appropriate DXS button. [Confirmation tone]
6. Press **[RELEASE]** . [ATND lamp goes out, and PA lamp lights].

To Activate Call Forwarding to the Attendant Queue

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Follow Me access code,

   or

   Press the **[CALL FORWARD FOLLOW ME]** button. [Second dial tone]
3. Dial the attendant access code,

   or

   Dial a LDN (Listed Directory Number). [Confirmation tone]
4. Go on-hook.

To Activate Call Forwarding—Off Net

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Follow Me access code,

   or

   Press the **[CALL FORWARD FOLLOW ME]** button. [Second dial tone]
3. Dial the trunk-group access code,

   or

   Dial the ARS or WCR access code. [Third dial tone].
4. Dial the off-net telephone number. [Confirmation tone]
5. Go on-hook.

To Cancel Call Forwarding—Follow Me

From the forwarding terminal:

1. Go off-hook. [Dial tone]
2. Dial the Call Forwarding—Cancel access code,

   or

   Press the **[CALL FORWARD FOLLOW ME]** button. [Confirmation tone]
3. Go on-hook.

From an attendant console for a voice terminal:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights.]
2. Press **[START]** . [Dial tone]
3. Dial the Call Forwarding—Cancel access code. [Second dial tone]
4. Dial the extension number of the forwarding terminal. [Confirmation tone]
5. Press **[Release]** . [ATND lamp goes out, and PA lamp lights.]

Considerations

Double Forwarding Not Allowed

Terminal A may activate forwarding to terminal B, and terminal B may activate forwarding to terminal C. However, terminal A's calls will not forward to terminal C. Under normal circumstances, voice terminal calls forward only once.

Forwarding to an Attendant

Call Forwarding—Follow Me can be used to forward all calls to the attendant queue, but forwarding to a ***selected*** attendant is not allowed.

Call Forwarding—Follow Me override allows the forwarded-to extension to call the forwarding extension in a forwarding relationship. However, an attendant cannot call an extension that has activated Call Forwarding—Follow Me to the attendant queue.

The attendant queue can be the forwarded-to destination in a call forwarding relationship. However, the attendant cannot originate Call Forwarding in order to forward attendant-seeking calls to another destination.

Hard and Soft Processor Swaps

Call Forwarding relationships are stored in a translation portion of switch memory. Therefore, if calls are forwarded to a local extension, DCS extension, or public-network telephone and then a hard processor swap occurs, the forwarding relationship will endure the hard swap.

Intercept Tone

Intercept tone is heard when the terminal user attempts to establish a new destination for forwarding before canceling the previous destination.

Intercept tone is also heard when the forwarded-to voice terminal attempts to forward calls toward its own forwarding voice terminal.

Additionally, intercept tone is heard when attempting to activate Call Forwarding—Follow Me and Call Forwarding—Busy and Don't Answer at the same time on the same terminal.

Off-Net Forwarding

On System 85 and DEFINITY Generic 2.1 switches, Call Forwarding—Off-Net can only be activated to forward calls to nontoll 7-digit telephone numbers (including private network addresses using the AAR feature).

On DEFINITY Generic 2.2 switches, Call Forwarding—Off-Net may or may not be allowed toward a toll number (including international destinations), depending on extension class of service permissions and restrictions of the forwarding extension. Specifically, Call Forwarding Off Net Toll is activated in Procedure 000, Word 3, field 7 (Generic 2.2 only).

When a call is forwarded off-net to a toll destination, the forwarding extension is billed for the toll charges.

Tie Trunks

Call Forwarding—Follow Me cannot forward calls off net using a tie trunk dial access code. To forward calls off-net, either the AAR, ARS, or WCR feature must be used.

Switch Capacities

Call Forwarding can be simultaneously activated to as many as 3200 external telephone numbers and / or extension numbers on other switches in the DCS network.

Call Forwarding can be simultaneously activated to any number of local voice terminals.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

When Call Forwarding—Follow Me is active at the called voice terminal, the forwarding operation occurs before Attendant Call Waiting is allowed. There are two possible operations. Attendant calls forward to and ring at the forwarded-to voice terminal (if this voice terminal is idle). Otherwise, attendant calls forward to, and then wait on, the forwarded-to voice terminal (if this voice terminal is busy).

AUDIX (Audio Information Exchange)

The Call Forwarding—Follow Me feature can be used to forward all calls to the AUDIX extension number. When this is done, forwarded calls enter the AUDIX queue.

AAR (Automatic Alternate Routing)

For System 85 and DEFINITY Generic 2.1 switches the AAR feature can be used with the Call Forwarding—Follow Me feature to forward calls to stations on other nodes of a private networking arrangement. For forwarding off-net within a private networking arrangement, the AAR feature must be used, as Call Forwarding off-net to a tie trunk dial access code is not allowed.

ACD (Automatic Call Distribution)

Call Forwarding—Follow Me (Intraflow—All) when activated by (or for) an ACD split supervisor diverts the split's calls to a destination within the local switch. Call Forwarding—Follow Me (Interflow—All) when activated by (or for) an ACD split supervisor diverts the split's calls to a destination within the DCS network.

- Only the split supervisor or system supervisor can activate or deactivate Call Forwarding—Follow Me for an ACD split. Nonsupervisory split members cannot activate call forwarding for the entire split.
- Only ACD calls forward when Call Forwarding—Follow Me is active for the split. Calls to an individual extension number do not forward.
- Nonsupervisory split members may activate Call Forwarding—Follow Me for calls to the individual member's extension number.

Call Forwarding—Off Net

An ACD split supervisor or system supervisor is not allowed to activate Call Forwarding—Off Net for the split. When this is attempted, the switch returns intercept tone.

Automatic Callback

Call Forwarding—Follow Me has no effect on an Automatic Callback call origination. Callbacks always return to the originating terminal, not to the forwarded-to terminal.

If Call Forwarding—Follow Me is active at the called terminal when the Automatic Callback call is originated, the forwarded-to terminal is treated as the called line for the call.

If Call Forwarding—Follow Me is activated by the called terminal after an Automatic Callback request is placed, the Automatic Callback request remains active toward the originally called voice terminal. The callback request is not reapplied to the forwarding destination.

ARS (Automatic Route Selection)

For System 85 and DEFINITY Generic 2.1 switches the ARS feature can be used with the Call Forwarding—Follow Me feature to forward calls off-net to non-toll numbers. When administering Call Forwarding off net, all three ARS plans should be administered to contain patterns with at least one preference in Procedure 309, Word 1. Otherwise, when activating the Call Forwarding—Off Net feature, the ARS access code cannot be used as part of the destination's telephone number. Rather, the appropriate trunk-group dial access code would have to be dialed.

Also, when administering Call Forwarding—Off Net, the desired local office codes should be specified in the ARS Toll Table (Procedure 309, Word 2 and Procedure 309, Word 1, Field 9). Otherwise, when activating the Call Forwarding—Off Net feature, an office code that is not specifically assigned as ***local*** is presumed by the Call Forwarding—Off Net software to be a ***toll*** office code. And, since forwarding to the toll network is not provided, the switch would return intercept treatment.

Busy Verification of Lines

Busy verification is allowed toward a line with Call Forwarding—Follow Me active.

Call Coverage

Call Forwarding—Follow Me takes precedence over Call Coverage in the following situations.

The following list shows the alternate treatments for cases where a covering user has Call Fowarding—Follow Me active. These specified treatments are based on the assumption that the alternate voice terminal has at least one idle appearance.

- If Call Forwarding—Follow Me is active for a principal who has coverage, calls to that principal's extension redirect according to call forwarding.
- If a coverage point (other than a coverage group) has Call Forwarding—Follow Me active, that point is not eligible to receive a coverage call.
  - — If there is only one coverage point, the principal's voice terminal rings.
  - — If the coverage point is the final (not the only) point, the previous coverage point rings.
  - — If there is a subsequent coverage point in the path, the forwarded coverage point is skipped.

If a situation arises where the alternate voice terminal(s) does not have an idle appearance, the switch returns busy tone to the calling party.

When a call is forwarded to a principal with coverage active, the forwarded call does not redirect to coverage. If the principal doesn't answer, the forwarded call will ring (until abandoned) at the principal's voice terminal. However, if every appearance of the principal's voice terminal is busy, the switch returns busy tone to the calling party.

Also, if Call Forwarding—Follow Me is active for a group coverage point (ACD split), calls will cover to the split's queue and then intraflow or interflow according to call forwarding.

CDR (Call Detail Recording)

The extension that a call forwards to (designated terminal) is the extension number that is recorded in the called number field.

Call Vectoring

Call Forwarding—Follow Me can be used to forward calls to a Vector Directory Number. In this case, the forwarded-to vector controls call processing for the forwarded call. For example, the call could enter an ACD split's queue (including AUDIX), and be processed according to the vector's programming.

With Call Vectoring assigned, an ACD split supervisor can use this feature to forward calls for the supervisor's individual extension. Without Call Vectoring assigned, Call Forwarding—Follow Me is instead used to forward calls which are directed to the split's queue.

The destination for a "route to extension" step in a vector can be an extension with Call Forwarding—Follow Me activated. When a VDN call routes to a forwarding extension, the VDN call will forward to and ring a ***local*** forwarded-to extension. If the forwarded-to destination is an ***off-net*** telephone number, vector processing will consider the "route to" step to have an invalid destination (that is, skip the step) unless the "route to" step is a final effective step. In this case, vector processing will execute the "route to" step and allow the call to forward off-net.

VDN Override does not apply to calls that are forwarded to a VDN by the Call Forwarding—Follow Me feature. For calls that are forwarded to a VDN, the originally called terminal's name remains permanently associated with the call.

Call Waiting

When Call Forwarding—Follow Me is active at the called voice terminal, the forwarding operation occurs before Call Waiting is allowed. There are two possible operations. Calls forward to and ring at the forwarded-to voice terminal (if this voice terminal is idle). Otherwise, calls forward to and wait on the forwarded-to voice terminal (if this voice terminal is busy).

Data Call Setup

Call Forwarding—Follow Me functions normally for data terminals. When a data terminal has Call Forwarding—Follow Me active, the display message "FORWARDED" and the ASCII Bell Ring character are sent to the data terminal to alert the user to the fact that Call Forwarding is still in effect.

DDC (Direct Department Calling)

Call Forwarding—Follow Me, when activated for a DDC group, routes all DDC calls to a designated terminal, the attendant queue, the centralized attendant queue, or to another UCD or DDC group's queue immediately after dialing. If a call is already queued when this feature is activated, the call remains in queue for 7 seconds before forwarding.

- Only the controlling terminal or attendant can activate or deactivate Call Forwarding—Follow Me for a DDC or UCD group. Other group terminals cannot activate or cancel Call Forwarding—Follow Me even if authorized by their extension class of service.
- Only calls to the DDC or UCD group number forward when Call Forwarding—Follow Me is active. Calls to an individual terminal or controlling terminal number do not forward.
- The stop hunt option should be assigned to each DDC or UCD group member's class of service. Otherwise, when calls forward to a group member's extension, the call is treated as a call to the group (hunting occurs).

Display—Voice Terminal

The Display—Voice Terminal feature is fully compatible with the Call Forwarding—Follow Me feature. When a call is forwarded from a display capable terminal, the message "FORWARDED" appears on the terminal display.

DCS (Distributed Communications System)

The Call Forwarding—Follow Me feature is transparent in the DCS environment. The forwarded-to extension in a call forwarding relationship can reside in a different DCS node. If the attendant sets up call forwarding for a station on a remote DCS node, forwarding is limited to AAR, ARS features, or WCR networks 1 and 2, and to a 7-digit number.

EUCD (Enhanced Uniform Call Distribution)

Call Forwarding—Follow Me (Intraflow—All), when activated by (or for) an EUCD split supervisor diverts the split's calls to a destination within the local switch.

- Only the split supervisor or system supervisor can activate or deactivate Call Forwarding—Follow Me for an EUCD split. Nonsupervisory split members cannot activate call forwarding for the entire split.
- Only EUCD calls forward when Call Forwarding—Follow Me is active for the split. Calls to an individual extension number do not forward.
- Nonsupervisory split members may activate Call Forwarding—Follow Me for calls to the individual member's extension number.

ENP (Extension Number Portability)

Call Forwarding—Follow Me is a transparent DCS feature. However, calls to a terminal with forwarding active to an extension number that is subsequently ported do not route properly. To correct this problem, the forwarding activation must be canceled and then reapplied.

Hold

Hard hold ringbacks for single-appearance voice terminals are not forwarded. Instead, the held call rings back the terminal that placed the call on hold when the user goes on-hook.

Hunting

When a call forwards to a terminal in a hunt group, another terminal in the hunt group may receive the call rather than the designated terminal (hunting occurs).

When a terminal in a hunt group has Call Forwarding—Follow Me active, the terminal is temporarily removed from the hunt group. Calls hunting through the hunt group bypass this terminal.

If the designated terminal has Call Waiting active and is in a hunt group, a call forwarded to the busy designated terminal hunts for an idle line first and then waits on the designated terminal if no idle line is found.

IPA (Interpartition Access)

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition group or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition group.

LXD (Last Extension Dialed)

Call Forwarding can be used in conjunction with Last Extension Dialed. This is useful when activation of the Call Forwarding—Follow Me Answer feature toward the last extension dialed is desired. (Dial the Call Forwarding—Follow Me DAC or press the feature button before pressing the LXD feature button.)

LND (Last Number Dialed)

Call Forwarding can be used in conjunction with Last Number Dialed. This is useful when activation of the Call Forwarding—Follow Me Answer feature toward the last number dialed is desired. (Dial the Call Forwarding—Follow Me DAC or press the feature button before pressing the LND feature button.)

Leave Word Calling

Leave Word Calling messages are addressed to the principal originally dialed, even when a redirection feature is active. The only exception to this is calls redirected to the attendant. Leave Word Calling is not allowed when a call is redirected to the attendant.

Line Lockout

A call directed toward a locked-out voice terminal with Call Forwarding active will forward to the designated terminal.

Look-Ahead Interflow

At a sending switch, the Call Forwarding—Follow Me feature can be used to forward calls to a VDN. If the VDN's associated vector contains a Look-Ahead Interflow "route to" step, calls that forwarded to the VDN will receive the same treatment from the "route to" step as calls that were directly dialed to the VDN.

Modem Pooling

The Call Forwarding—Follow Me feature functions normally for data calls.

One data call problem occurs when Modem Pooling is required. A data call requiring Modem Pooling should not be forwarded to an attendant. An attendant established (transferred) call does not receive a Modem Pooling conversion resource. The attendant must extend the call to a multiappearance voice terminal with one button transfer capability (see Data Call Setup feature) to complete the call for a conversion resource to be provided.

Override

An Override call does not forward when Call Forwarding—Follow Me is active. Three-burst ringing is provided for an idle forwarding terminal, and the override call enters the conversation of a busy forwarding terminal.

Priority Calling

When Call Forwarding is active at the called voice terminal, the forwarding operation occurs before Priority Calling is allowed. There are two possible operations. Calls forward to and ring at the forwarded-to voice terminal (if this voice terminal is idle). Otherwise, calls forward to and wait on the forward-to voice terminal (if this voice terminal is busy).

Queuing

If a callback attempt is made from a tandem switch to a subtending switch, the call appears as an ordinary incoming tie trunk call to the subtending switch. Therefore, if the called terminal has Call Forwarding—Follow Me active, the callback forwards to the designated terminal.

When a callback attempt is made to a local terminal, it is not forwarded if Call Forwarding—Follow Me is activated. The callback call is placed to the line which placed the call in queue.

Restriction—Attendant Control of Voice Terminals

See the "Restricting Feature Use" section of this feature description.

Restriction—Voice Terminal Restrictions

Calls may not forward to a voice terminal with Restriction-Voice Terminal Restrictions (Termination, Manual Terminating Line, or Terminal-to-Terminal Only Calling) activated.

Incoming calls on public or private network trunks may not forward to an Inward restricted terminal.

Call Forwarding—Follow Me functions normally for an Origination restricted voice terminal when an attendant activates the feature for the voice terminal. However, the Origination restricted voice terminal is only allowed to activate Call Forwarding—Follow Me from a hold or recall dial tone state. If Call Forwarding—Follow Me is activated, only the attendant can deactivate it.

If an unrestricted voice terminal is assigned as the forwarded-to voice terminal and then Restriction—Voice Terminal Restrictions (Inward, Termination, Terminal-to-Terminal Only Calling, or Manual Terminating Line) is activated in its extension class of service, calls will still forward to the voice terminal.

Call forwarding is permitted between two data modules even though both are Terminal-to-Terminal Only Calling restricted.

Ringing—Abbreviated and Delayed Ringing

Call Forwarding—Follow Me has precedence over Abbreviated and Delayed Ringing. Call Forwarding—Follow Me controls the redirection of ringing.

Ringing Cutoff

When Call Forwarding—Follow Me and Ringing Cutoff are both active at a called voice terminal, ring-ping is not provided as the call forwards.

Ringing Transfer

Call Forwarding—Follow Me takes precedence over Ringing Transfer. Call Forwarding—Follow Me controls the redirection of ringing.

Tenant Services

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition.

Call Forwarding—Off Net is allowed in a partitioned System 85 or DEFINITY Generic 2. This forwarding can be activated on System 85 or DEFINITY Generic 2.1 switches using the ARS/AAR access code or using the access code of a trunk group that is dedicated to the user's extension partition. For Generic 2.2 switches, the appropriate network access code for the World Class Routing feature is used.

Timed Reminder

When pressing the STA ID button, the attendant display indicates the extension number that the call forwarded to rather than the originally called extension number.

Unattended Console Service—Preselected Call Routing

When in the unattended console (night service) mode, the extension designated to receive can activate Call Forwarding—Follow Me. As long as the forwarded to station is on-net, there is not problem. However, calls should not be forwarded to an off-net station when in the unattended console mode. Attendant-seeking-calls will forward to an on-net station but not to an off-net station. If forwarding to an off-net station is attempted, attendant-seeking-calls will not forward but will wait in the attendant queue until the call is abandoned or an attendant console is activated.

UCD (Uniform Call Distribution)

Same as the Direct Department Calling interaction.

WCR (World Class Routing)

On DEFINITY Generic 2.2, the WCR feature replaces the AAR and ARS features. Some differences occur between interactions with AAR/ARS and WCR.

When administering Call Forwarding—Off Net, all time-of-day plans do not need to be administered as they do with ARS. An off-net forwarding arrangement is verified on the switch based on the time-of-day plan in effect at the time forwarding is initiated. If the forwarding party does not have off-net forwarding access under some other time-of-day plan, calls received during those plan periods do not forward off-net. Rather, they ring at the originally called extension.

Also, Call Forwarding—Off Net destinations may or may not be toll destinations with WCR (only non-toll destinations were allowed with ARS). Toll permission for Call Forwarding—Off Net with WCR is determined on an extension basis, specifically field 7 of Procedure 000, Word 3. With WCR, if Call Forwarding—Off Net is authorized for a particular extension, the forwarded-to address can include all otherwise accessible destinations including international call addresses (up to 31-digit numbers).

Restricting Feature Use

Voice Terminal Restrictions

The voice terminal restrictions that restrict call forwarding are:

- Termination restriction
- Manual Terminating Line restriction
- Terminal-to-Terminal Only Calling restriction.

Attendant Control of Voice Terminal Restrictions

The application of Attendant Control of Terminal Access restrictions takes precedence over the Call Forwarding—Follow Me feature only when applied before the Call Forwarding—Follow Me feature is activated. The restrictions are:

- Termination restriction
- Total restriction
- Terminal-to-terminal restriction.

Hardware Requirements

None.

Feature Administration

Assignment of the Call Forwarding—Follow Me feature is on an extension class of service basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

Administration Procedures
Call Forwarding—Follow Me

Procedure	Word	Purpose	SMT
000	1	Assigns the extension class of service to an extension number.	Yes
000	2	Specifies the destination AUDIX adjunct for calls forwarded to the AUDIX extension number (Field 10).	Yes
000	3	For Generic 2.2, assigns extension permissions including Call Forwarding—Off Net toll permission.	N/A
010	1	Assigns Call Forwarding—Follow Me to an extension class of service.	Yes
010	2	Assigns Call Forwarding—Off Net to an extension class of service.	Yes
054	2	Administers the Call Forwarding—Follow Me button to a multiappearance voice terminal. The applicable encode is as follows: 3 Call Forwarding—Follow Me.	Yes
204	1	Designates the desired alphanumeric display for calls forwarded to the attendant. The applicable encode is as follows: R2 V1 to R2 V3: 290 Call Forwarding R2 V4 and later: 2290 Call Forwarding.	No
309	1	For System 85 and Generic 2.1, assigns the three ARS (time-of-day) plans to routing tables for use by Call Forwarding—Off Net. This is not required with Generic 2.2.	Yes
309	2	For System 85 and Generic 2.1, assigns the local office codes to the ARS Toll Table for use by Call Forwarding—Off Net. (Local office codes that are not assigned to the toll table are presumed to be ***toll*** office codes.)	Yes
350	1	Assigns the first digit of the dial access codes (if required).	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 1 Call Forwarding—Follow Me 3 Call Forwarding—Cancel.	No

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Call Forwarding—Follow Me	
Path Name	**Purpose**
terminal-change class-of-service attributes	Assigns Call Forwarding—Follow Me and Call Forwarding—Off Net to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number. Also, specifies the AUDIX number (1 to 4) for calls forwarded to the AUDIX extension number.
terminal-change terminal buttons	Administers the the Call Forwarding—Follow Me button to a multiappearance voice terminal.

Notes:

Call Park

Description

This feature is used to place a call on hold and then transfer the call to an answer-back channel. That call can then be answered on any other voice terminal within the switch. The call is reanswered by dialing the answer-back access code and the answer-back channel number.

This procedure is useful when call-related information is in another area, or the call could be handled more conveniently in another area. This feature also allows an option to have music while waiting for an answer-back.

Feature History and Development

This feature was first available on System 85 in Release 1. The enhancements to this feature include:

- Automatic attendant recall was provided for Release 2, Version 2.
- An option to have music while waiting for an answer-back was added for Release 2, Version 3.
- An administrable recall button was provided for Release 2, Version 4 and was also retrofitted to the Release 2, Versions 2 and 3.

User Operations

The following are the user operating procedures for this feature.

To Park a Call

Using a single-appearance voice terminal:

1. Press the **[RECALL]** button,

   or

   Momentarily press the switchhook. [Second party is placed on hold. Recall dial tone is heard.]
2. Dial the Call Park trunk-group access code. [Second dial tone]
3. Dial the Call Park zone number.
4. Dial an idle answer-back channel number. [Confirmation tone]
5. Go on-hook. [Held party is transferred to the previously dialed answer-back channel and hears ringback tone or music while waiting for the answer-back call.]

Using a multiappearance voice terminal:

1. Press **[TRANSFER]** . [Second party is placed on hold. Dial tone is heard.]
2. Dial the Call Park trunk-group access code. [Second dial tone]
3. Dial the Call Park zone number.
4. Dial an idle answer-back channel number. [Confirmation tone]
5. Press **[RECALL]** . [Ringback tone]
6. Press **[TRANSFER]** . [Held party is transferred to the previously dialed answer-back channel and hears ringback tone or music while waiting for the answer-back call.]
7. Go-on-hook.

To Pickup a Parked Call:

1. Be sure, as a user, that the Call Park feature is activated.
2. Go off-hook. [Dial tone]
3. Dial the answer-back access code.
4. Dial the previously used answer-back channel number. [Ringback tone or music is removed from the waiting line, both parties hear confirmation tone, a 2-party connection is established, and the answer-back channel is released.]

Considerations

Attendant Console Restriction

The Call Park feature is designed for use from a voice terminal. Attendant seeking calls or calls redirected to an attendant can be placed in Call Park from the attendant console. However, an attendant console cannot pickup a parked call. The answer-back access code cannot be dialed from an attendant console.

Busy Tone

Busy tone is heard if the call park zone or answer-back channel is busy.

Intercept Tone

Intercept tone is heard if an invalid access code, invalid zone number, invalid answer-back code, or invalid channel number is dialed. Intercept tone is also heard if the answer-back code was dialed, and the second party is no longer waiting for the answer-back call.

Answer-Back Channels

Nine answer-back channels are available. These are the same answer-back channels used with the Loudspeaker Paging feature. These answer-back channels are shared by both features.

Availability

Call Park is available only with the Loudspeaker Paging. A single-line voice terminal user must be assigned the Conference-Three Party feature to use Call Park.

Parked Trunk Calls

Beginning with R2 V2, incoming trunk calls and outgoing trunk calls (with Disconnect Supervision assigned in Procedure 101, Word 1) that are parked by a voice terminal will automatically recall an attendant after two minutes At this time, the attendant can. appropriately handle the call.

Parked Outgoing Trunk Calls

Beginning with R2 V2, outgoing trunk calls (without Disconnect Supervision assigned in Procedure 101, Word 1) that are parked by a voice terminal are automatically disconnected after two minutes.

Administrable Recall Buttons

Some multiappearance voice terminals do not have a fixed RECALL button. If RECALL buttons are needed for these voice terminals, RECALL buttons can be assigned to the terminals using Procedure 054, Word 1.

Interactions With Other Features

The following System 85 or DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

The switch denies Attendant Call Waiting toward a line that is parked. Busy tone is returned.

Busy Verification of Lines

A line that is in Call Park cannot be busy verified using the Busy Verification of Lines feature. Intercept tone is returned.

Call Coverage

When a covering user activates the Call Park feature, the Temporary Bridged Appearance at the principal's voice terminal is removed, and the principal is unable to bridge onto the parked call.

Call Detail Recording (CDR)

The CDR feature records the extension number of the last voice terminal in a Call Park connection.

Call Waiting

The switch denies Call Waiting toward a line that is in call park.

Loudspeaker Paging Access

The Loudspeaker Paging Access feature and the Call Park feature are closely related. Administering Loudspeaker Paging, in effect, also enables Call Park. A paging zone that is not assigned for Loudspeaker Paging is assigned for use by Call Park. The paging zone assigned to Call Park requires an auxiliary trunk circuit to prevent alarms; however, it is not necessary to connect a paging amplifier to Call Park auxiliary trunk circuits. Both Call Park and Loudspeaker Paging Access share the same nine answer-back channels.

Music-on-Hold Access

When Music-on-Hold is implemented, music can be provided for a call on hold in an answer-back channel.

Override

The switch denies Override toward an extension in call park.

Priority Calling

The switch denies Priority Calling toward a line that is parked.

Tenant Services

The call park zones for the Call Park feature are not partitioned. By default, the provided zones are equally accessible to voice terminal users in any extension partition.

Voice terminal access to the Call Park feature can be limited in the voice terminal class of service. To limit voice terminal access, assign a Miscellaneous Trunk Restrictions group containing the Call Park trunk group to a voice terminal class of service in Procedure 010, word 3.

Answer-back channels for the Call Park feature are not partitioned. A parked call can be retrieved by dialing the answer-back access code from any voice terminal in the switch.

Trunk Verification—Attendant and Voice Terminal

The Trunk Verification feature cannot verify a trunk that is in Call Park.

Unattended Console Service—Preselected Call Routing

When Preselected Call Routing is active, if a trunk party is placed in Call Park and the 2-minute timer for ringback tone times out, the trunk call is routed to the assigned preselected voice terminal.

Hardware Requirements

The following additional or special hardware is required for the Call Park feature.

For Traditional Modules:

- SN231, Auxiliary Trunk Circuit Pack

  Each Call Park zone requires an auxiliary trunk circuit (four circuits per SN231).

For Universal Modules:

- TN763C, Auxiliary Trunk Circuit Pack

  Each Call Park zone requires an auxiliary trunk circuit (four circuits per TN763C).

Feature Administration

Assignment of the Call Park feature is on a per-system basis.

On System 85 switches, this feature is administered using the Maintenance and Administration Panel (MAAP). The customer can partially administer this feature using the System Management Terminal (SMT) or the Terminal Change Management (TCM) feature.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

To provide music while waiting for answer back, the Music-on-Hold feature must also be assigned.

The following are the applicable administration procedures.

Administration Procedures — Call Park			
Procedure	**Word**	**Purpose**	**SMT**
000	1	Assigns the voice terminal class of service to an extension number.	Yes
010	1	Assigns Conference-Three Party/Transfer to a voice terminal class of service for use with the Call Park feature.	Yes
010	3	Assigns Miscellaneous Trunk Restrictions to a voice terminal class of service.	Yes
054	1	Assigns the RECALL button to a multiappearance voice terminal without a fixed RECALL button. The applicable encode is: 27 Recall.	Yes
100	1	Administers the trunk-group dial access code and the trunk type for the Call Park feature. The applicable trunk-type encode includes: 54 Loudspeaker paging interface.	No
102	—	Administers the Miscellaneous Trunk Restriction group associated with the Call Park trunk-pup dial access code.	Yes
150	—	Assigns the SN231 or TN763C equipment location and trunk feature of a Call Park trunk to its trunk-group number.	No
275	1	Assigns Call Park to the system class of service (Field 7) and assigns the music-on-hold option.	Yes
350	1	Assigns the first digit of the dial access codes (if required).	No
350	2	Assigns the feature dial access code. The applicable encode is: 17 Paging answer back.	No

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Call Park	
Path Name	**Purpose**
terminal-change class-of-service attributes	Assigns Conference-Three Party/Transfer to a voice terminal class of service for use with the Call Park feature. Also, use this screen to assign Miscellaneous Trunk Restriction groups to the terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Notes:

Call Pickup

Description

This feature allows anyone within a specified call pickup group to answer at their own voice terminal a call that is ringing at another extension within the group. This provides a simple means of answering unattended voice terminal calls.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Answer a Call at Another Extension Within the Same Pickup Group

At a single-appearance or multiappearance voice terminal:

1. Be sure a voice terminal in your pickup group is ringing.
2. Go off-hook,

   or

   Press an idle appearance button. [Dial tone]
3. Dial the Call Pickup access code. [The call pickup user and the calling party are connected. The called terminal stops ringing.]

At a multiappearance voice terminal with a CALL PICKUP button:

1. Be sure a voice terminal in your pickup group is ringing. [The CALL PICKUP lamp flashes.]
2. Press **[CALL PICKUP]**. [The call pickup user and the calling party are connected, called terminal stops ringing, and flashing CALL PICKUP status lamp goes out.]

Considerations

Busy Tone

Busy tone is heard when attempting to pick up an Automatic Callback or Queuing callback call.

Intercept Tone

Intercept tone is heard when attempting to pickup a call in another pickup group or attempting to pickup a phone that is not ringing.

Limitations

Any number of extension numbers can be assigned to a Call Pickup group. A maximum of 999 Call Pickup groups can be assigned.

Terminal Locations

Call Pickup cannot be used to answer callback calls or calls to terminals in other pickup groups.

Because a Call Pickup user needs to know when a terminal in the group is ringing, place single-appearance terminals within a pickup group close together.

Colocating the voice terminals is not as necessary for multiappearance terminals. On multiappearance terminals, the Call Pickup lamp can flash to alert the Call Pickup user to a group call. This occurs whenever a voice terminal in the group is ringing and an appearance of an extension in the Call Pickup group is either manually or automatically preselected (the red status lamp is lit for an appearance of the extension).

Multiple Ringing Terminals

If more than one voice terminal in a pickup group is ringing when Call Pickup is activated, the switch selects the call which the Call Pickup user answers The algorithm used is that a Call Pickup activation answers the most recent call that is ringing a voice terminal in the call pickup group.

Hard and Soft Processor Swaps

The extension numbers in a Call Pickup group are stored in a translation portion of switch memory. Therefore, the members in a Call Pickup group will remain unchanged after a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Automatic Callback

A terminal user cannot pickup an Automatic Callback call to the originating terminal. Busy tone is heard.

Call Coverage

The switch denies activation of the Call Pickup feature to a Call Coverage user when attempting to:

- Pick up a Temporary Bridged Appearance on a principal's voice terminal.
- Pick up a coverage call during Coverage Tone or during the Caller Response Interval.

If a member of the covering user's pickup group uses Call Pickup to pickup a redirected call, the temporary bridged appearance on the principal's voice terminal remains intact. After the redirected call is picked up, the principal can still go off-hook on the temporary bridged appearance to join the call with the Call Pickup user.

CDR (Call Detail Recording)

When a call is answered using the Call Pickup feature, the answering terminal extension is recorded, not the called terminal extension.

Data Call Setup

Although a data terminal (or more precisely, its extension number) can be assigned to a Call Pickup group, Call Pickup is blocked for data terminals. Data terminals cannot be used to pickup calls, nor can calls directed to a data terminal be picked up by other members of the pickup group.

Data Protection

Use of Call Pickup toward a call directed to a terminal with Data Protection-Permanent active is denied.

Extension Number Portability

An extension number must be removed from a pickup group, if assigned, before it can be ported to another node.

Hold

A single-appearance voice terminal user is denied use of the Call Pickup feature while holding a call on hard hold and soft hold at the same time.

A voice terminal user is allowed to place a call on hard hold and then answer another call using Call Pickup. If this is done using soft hold, the held call is moved to hard hold and can be retrieved using the Call Hold access code.

Intercom

The Call Pickup feature cannot be used to pickup an Intercom call.

Interpartition Access

There are no tests in Procedure 000, Word 2 to ensure that a Call Pickup group is only assigned to extensions residing in the same partition group. The system manager should ensure that every member of each Call Pickup group belongs to the same partition group.

When Call Pickup groups have been assigned to overlap partition-group boundaries, the call-processing software provides partitioning for the feature. If a Call Pickup group member in one partition group tries to pickup a call to another group member residing in a different partition group, intercept treatment is returned by the switch.

Leave Word Calling

Leave Word Calling messages are always addressed to the principal originally called, even when a call redirects via Call Pickup.

Look-Ahead Interflow

At a receiving switch, the Look-Ahead Interflow feature and the Call Pickup feature are compatible. A member of a Call Pickup group at the receiving switch can normally answer Look-Ahead Interflow calls that are ringing at another extension in the group.

Queuing

A terminal user cannot pickup a local Queuing callback. Busy tone is heard.

A callback from a tandem switch looks like an incoming tie trunk call and can be picked up. Also, a callback call between a main and subtending switch appears as a normal tie trunk call and can be picked up.

Restriction—Attendant Control of Voice Terminals

A voice terminal that is otherwise restricted from receiving calls (by Controlled Terminal-to-Terminal, Outward and Terminal-to-Terminal, Controlled Termination, or Outward and Termination Restriction) is allowed to pickup group members' calls using Call Pickup.

Restriction—Voice Terminal Restrictions

A voice terminal with Voice Terminal Restrictions (Inward, Termination, Terminal-to-Terminal Only Calling, or Manual Terminating Line) activated may pickup a call directed to another voice terminal in the restricted voice terminal's Call Pickup group.

Serial Calls

When Serial Calls is in effect, pressing the RECALL button at a local terminal recalls the attendant. Therefore, a terminal user cannot access the Call Pickup feature during a serial call from a single-appearance terminal.

Tenant Services

There are no tests in Procedure 000, Word 2 to ensure that a Call Pickup group is only assigned to extensions residing in the same extension partition. The system manager should ensure that every member of each Call Pickup group belongs to the same extension partition.

When Call Pickup groups have been assigned to overlap partition boundaries, the call-processing software provides partitioning for the feature. If a Call Pickup group member in one partition tries to pickup a call to another group member residing in a different extension partition, intercept treatment is returned by the switch.

Restricting Feature Use

The Controlled Total Restriction and Origination Restriction features can prevent an extension from using the Call Pickup feature.

Hardware Requirements

None.

Feature Administration

Assignment of the Call Pickup feature is on a per-terminal basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer Call Pickup using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — CALL PICKUP			
PROCEDURE	WORD	PURPOSE	SMT
000	2	Assigns Call Pickup group numbers to a set of extensions.	Yes
054	2	Administers the Call Pickup button to a multiappearance voice terminal. The applicable encode is as follows 7 Call Pickup.	Yes
075	1	Displays the voice terminals sharing a Call Pickup group assignment.	Yes
350	1	Assigns the first digit of the dial access code (if required).	No
350	2	Assigns the feature dial access code. The applicable encode is 5 Call Pickup.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — CALL PICKUP	
PATH NAME	PURPOSE
terminal-change group pickup	Displays or prints a report of the Call Pickup groups.
terminal-change extensions attributes	Assigns Call Pickup group numbers to a set of extension numbers.
terminal-change terminal buttons	Assigns the Call Pickup button to a multiappearance voice terminal.

CallVisor™ ASAI Gateway Interface

Description

The CallVisor™ ASAI (Adjunct/Switch Application Interface) Gateway Interface feature (hereafter called ASAI Gateway Interface) provides an interface between a DEFINITY Generic 2 and an ASAI Gateway. An ASAI Gateway is a hardware and software package that provides a gateway between the switch and call-center software, enabling the call-center software to monitor and control certain incoming, outgoing, and internal calls. The ASAI Gateway software resides on an AT&T 3B2 computer. The call-center software resides on a separate host computer. The call-center software is not part of the ASAI Gateway Interface feature. The call center's owner is responsible for developing or obtaining the call-center software.

Compared to a telemarketing operation, a call-center operation has a broader scope. A telemarketing operation typically handles only incoming calls. A call-center operation handles high volumes of incoming and outgoing calls as well as call transfers and conferences.

Figure 33-1 shows an example of an ASAI Gateway Interface configuration. The ASAI Gateway is a 2-way gateway; Information travels through the ASAI Gateway from the switch to the call-center software and from the call-center software to the switch. Each answering position, typically an ACD (Automatic Call Distribution) agent, has a voice terminal and a data terminal or a work station with voice and data capabilities.

Feature History and Development

The ITGI (Integrated Telemarketing Gateway Interface) feature was first available with Issue 3.0 of DEFINITY Generic 2.1.

Beginning with DEFINITY Generic 2.2, the name of the feature changes to CallVisor ASAI Gateway Interface.

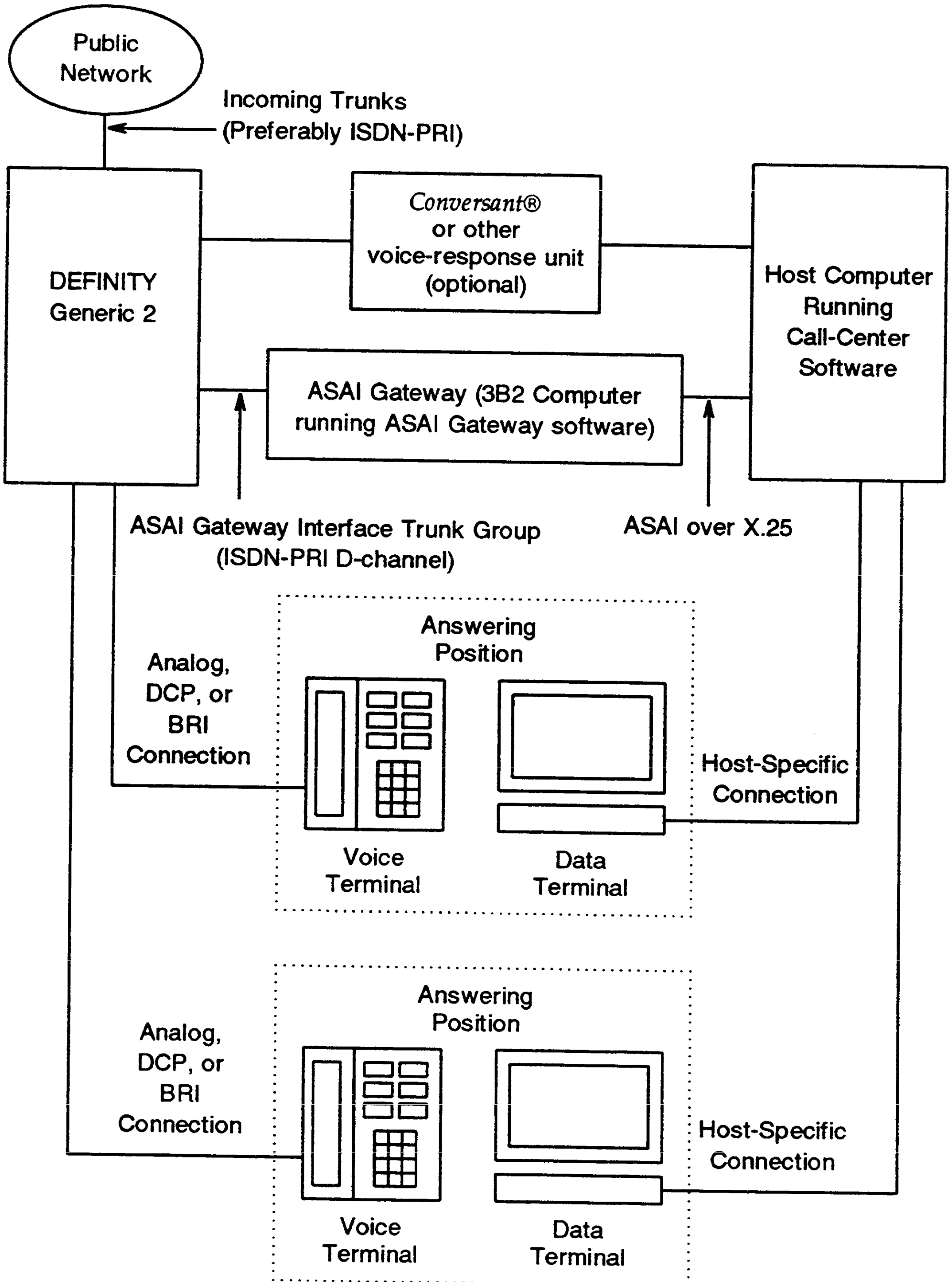


Figure 33-1. Example ASAI Gateway Interface Configuration

Required Features

The following DEFINITY Generic 2 features must be assigned to the system.

- ISDN—PRI

  At least one ASAI Gateway Interface (ISDN—PRI) trunk group is required for each ASAI Gateway and ISDN must be enabled in the system class of service (Procedure 275 Word 4).

- Automatic Alternate Routing

  For System 85 and DEFINITY Generic 2.1, the AAR feature is used to route call-control messages between the switch and the ASAI Gateway. ASAI Gateway Interface trunk groups are assigned to one or more AAR routing patterns. The AAR feature can also be used to route outgoing calls controlled by the call-center software.

- Call Vectoring

  The Call Vectoring feature is used to process incoming calls and to route call-control messages between the switch and the ASAI Gateway. Call Vectoring can also be used to send calls controlled by the call-center software to an internal or external destination.

- World Class Routing

  For DEFINITY Generic 2.2, the WCR feature is used to route call-control messages between the switch and the ASAI Gateway. ASAI Gateway Interface trunk groups are assigned to one or more WCR routing patterns. The WCR feature can also be used to route outgoing calls controlled by the call-center software.

Related Features

The following DEFINITY Generic 2 features are commonly used with the ASAI Gateway Interface feature.

- Automatic Route Selection

  For System 85 and DEFINITY Generic 2.1, the ARS feature can be used to route outgoing calls controlled by the call-center software.

- Automatic Call Distribution

  An ACD agent is usually the answering destination for calls controlled by the call-center software.

- Look-Ahead Interflow

  The Look-Ahead Interflow feature can be used with the Call Vectoring and ACD features to intelligently interflow calls between locations connected by ISDN—PRI facilities.

- World Class Routing

  For DEFINITY Generic 2.2, the WCR feature can be used to route outgoing calls controlled by the call-center software.

Related Document

The following documents contain detailed information about the ASAI Gateway and describe how to integrate the ASAI Gateway into a call-center operation.

- *ASAI Gateway System Description and Planning* (585-246-201)

  Describes the ASAI Gateway and provides information for a successful design, configuration, and implementation.

- *ASAI Gateway Installation and Maintenance* (585-246-101)

  Provides procedures and information for installing administering, and troubleshooting the ASAI Gateway.

- *ASAI Gateway Call-Center Software Development* (585-246-202)

  Provides reference material to help system and software developers design and build the call-center software.

Because of the complexity of the ASAI Gateway Interface feature, an implementation should not be attempted without a thorough understanding of these documents.

Call Management Services

The ASAI Gateway Interface feature provides the following call management services:

- Incoming call management,
- Outgoing call management, and
- Transfer/conference management.

These call management services enhance agent call-handling capabilities and increase agent productivity in a call-center environment.

For incoming calls, the switch sends call information through the ASAI Gateway to the call-center software. The call-center software uses the information to determine how to handle the call and to retrieve relevant caller-related database information. Call-handling information is sent through the ASAI Gateway to the switch, which uses the information to deliver the incoming call to an available answering position. Database information, for example a customer account or catalog order form, is delivered to the same answering position.

Outgoing calls are handled in a similar way. By way of the ASAI Gateway, the call-center software sends the switch instructions for placing a call and the switch sends the call-center software information about the call. Outgoing calls can be initiated by an agent (using a data terminal) or by the call-center software.

Incoming and outgoing calls and the associated database information can be transferred from one answering position to another, or another answering position can be conference (added) onto a call.

Communication Links

The switch communicates with the ASAI Gateway by exchanging call-control messages over the D (signaling) channel of a special ISDN (Integrated Services Digital Network)—PRI (Primary Rate Interface) link called an ASAI Gateway Interface trunk group. No B (bearer) channels are assigned to the ASAI Gateway Interface trunk group because the voice portion of a call is not sent from the switch to the ASAI Gateway.

Because the link between the switch and the ASAI Gateway is a special ISDN—PRI facility, the ASAI Gateway can be located in the same room as the switch, in a different room in the same building, or in a different building.

The call-center software can monitor and control as many as 1024 active 2-party calls per ASAI Gateway Interface trunk group simultaneously. As many as four ASAI Gateway Interface trunk groups can be administered on a DEFINITY Generic 2. Each ASAI Gateway Interface trunk group can connect to a separate ASAI Gateway or more than one ASAI Gateway Interface trunk group can connect to the same ASAI Gateway. For example, if the system has two ASAI Gateways, two ASAI Gateway Interface trunk groups could connect to each ASAI Gateway. One trunk group could serve as the primary link to each ASAI Gateway and the other trunk group could serve as a backup if the primary link fails.

The link between the ASAI Gateway and the host computer is ASAI over X.25 protocol. ASAI is a protocol through which adjunct processors and switches cooperate to provide services that permit adjunct-based software applications to initiate, receive, and control calls or make use of switch features. X.25 is a standard data communications protocol defined by the CCITT (International Telegraph and Telephone Consultative Committee).

Incoming Call Management

Incoming call management enables the call-center software to monitor and control certain incoming calls. Figure 33-2 shows an example of an incoming call controlled by the call-center software. The arrows labeled V1 through V3 show the flow of the voice portion of the call, Ml and M2 show the call-control messaging associated with the call, and D1 through D3 show the flow of information to and from the call-center software running on a host computer.

A caller dials a telephone number associated with a call-center operation (V1). The call is routed through the public network to the DEFINITY Generic 2 (V2). Using Call Vectoring and AAR (WCR beginning with DEFINITY Generic 2.2), the switch sends call-notification and any available call information to the ASAI Gateway by way of the ASAI Gateway Interface trunk group (Ml). Based on call information received from the ASAI Gateway (Dl), the call-center software sends call-handling information back to the ASAI Gateway (D2). The ASAI Gateway passes the call-handling information on to the switch (M2), which routes the call to a local answering position (V3).

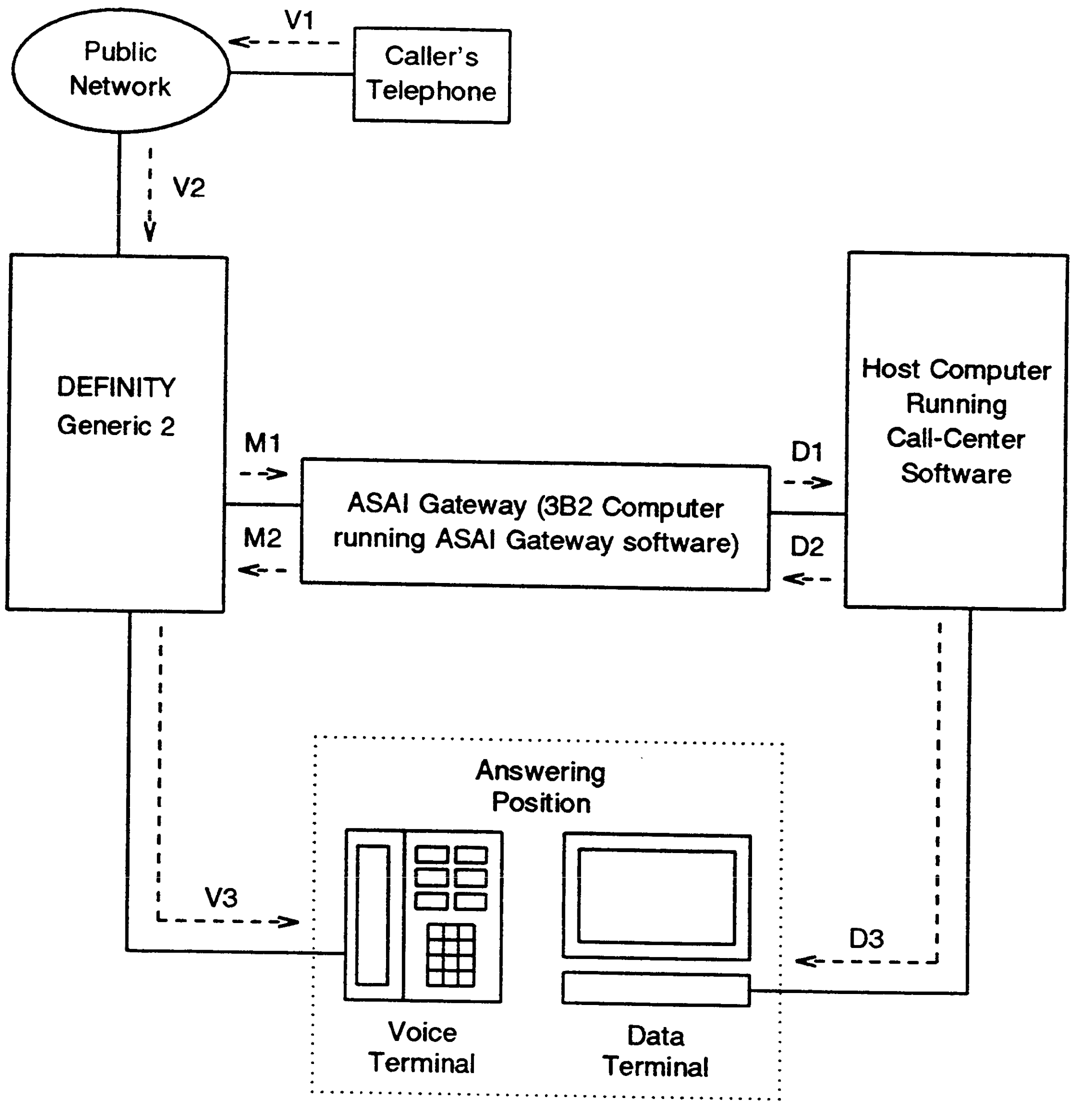


Figure 33-2. Incoming ASAI Gateway Interface Call

The call-center software uses call information to retrieve caller-related information from a database and deliver the information to an answering position. The timing of these events can be coordinated so that the database information arrives at an agent's data terminal (D3) at the same time as (or before) the voice call arrives at the agent's voice terminal. When an agent completes an incoming call session, the call-center software can disconnect the call immediately. For example, as soon as a caller's database record is released (from the data terminal), the voice call can be disconnected.

Call Destinations

Calls controlled by the call-center software can be sent to the following destinations:

- A VDN (Vector Directory Number). The vector associated with VDN can be programmed to process calls in a number of differenct ways, for example, calls can be queued to a local ACD split or routed to an internal or external destination.
- A specific ACD agent. If the system has CMS (Call Management System), this type of call can be labeled as an ACD call.
- A public-network number or a private-network number.
- A local non-agent extension number.

A call's first destination may or may not be its final destination. Switch features such as Call Forwarding, Call Coverage, Call Vectoring, and Look-Ahead Interflow may forward or redirect the call to one or more subsequent destinations.

Calls delivered to local (internal) destinations or routed to remote (external) destinations over end-to-end ISDN facilities can be monitored completely by the call-center software. Routing calls to remote destinations over facilities that are not end-to-end ISDN reduces the monitoring capabilities of the call-center software.

Call-Related Event Reports

Based on call-related events, the call-center software can determine the success or failure of each call and identify the answering position that handled the call. The call-center software also uses event reports to monitor a call's progress. The following call-related events can be reported to the call-center software.

- Call Offered

  This event is reported when the ASAI Gateway tells the switch to deliver a call to a VDN. The event is reported before vector processing begins.

- Alerting

  This event is reported when the destination voice terminal is selected. If the call encounters more than one destination (for example a call that goes to coverage after being delivered to its original destination) more than one alerting event is reported.

- Connected

  For local destinations, this event is reported when the answering position is connected to the call. For remote destinations, this event is reported when answer supervision is received.

- Transferred / Conferenced

  This event is reported when a local answering position transfers a call (using a voice terminal) to another local answering position or conferences (adds) another local answering position on to a call.

- Drop / Disconnect

  This event is reported when a party is dropped or disconnected from a call. One drop / disconnect event is reported for every party involved in a call.

- Call Ended

  This event is reported when a call terminates. That is, when the last party involved in a call disconnects from the call.

Automatic Number Identification and Dialed Number Identification Service

Depending on the type of incoming trunks and the available network services, the DEFINITY Generic 2 can receive call information from the public network and send it through the ASAI Gateway to the call-center software. To receive ANI (Automatic Number Identification), the incoming trunks (from the AT&T network) must be ISDN—PRL Furthermore, *Megacom*® 800 service and INFO-2 service are required. DNIS (Dialed Number Identification Service) can also be sent to the call-center software (by way of the ASAI Gateway).

The call-center software can route calls to specific agents or splits based on the ANI or DNIS. For example, calls from foreign language speaking Customers can be sent to a multilingual agent or split.

For calls abandoned before an agent answers, the ANI can be stored for later callback.

Voice Response Units

As shown in Figure 33-1, an optional *Conversant* or other voice-response unit can request, collect, and then send caller-supplied information to the call-center software. The *Conversant* can collect caller information that is not available, for example, because the incoming trunks (to the switch) are not ISDN—PRI, or because ANI is not available for a particular incoming call. As described previously, the call-center software can retrieve caller-related information from a database or make call-handling decisions based on the caller-supplied information. Switch administration and voice-response-unit software are required to route calls to and from the voice-response unit. The ASAI Gateway and voice-response unit are not directly connected.

Outgoing Call Management

Outgoing call management enables the call-center software to monitor and control certain outgoing calls. An outgoing call can be initiated by an agent or by the call-center software. An outgoing call initiated by the call-center software can be delivered to a specific agent or to the next available agent in an ACD split. For all types of outgoing calls, the call-center software can deliver database (called-party) information to the agent who handles the call and the switch can identify the agent to the call-center software.

Assigning agents to outgoing calling sessions when incoming call volumes slacken can increase agent productivity.

Agent-Initiated Outgoing Calls

Figure 33-3 shows an example of an agent-initiated outgoing call controlled by the call-center software. The arrows labeled V1, V2, and V3 show the flow of the voice portion of the call; M1, M2, and M3 show the call-control messaging associated with the call; and D1 through D5 show the flow of information to and from the call-center software running on a host computer.

Using a data terminal, an agent notifies the call-center software that he or she is available for an outgoing calling session (D1). The call-center software reserves the agent (so that the agent does not receive any incoming calls during the outgoing calling session) (V1) by sending call-handling information through the ASAI Gateway to the switch (D2, Ml). Based on database information received from the call-center software (D3), the agent initiates an outgoing call (D4). (This capability is known as preview dialing because the agent views a call list before initiating the call.) The call-center software sends instructions for placing the outgoing call to the ASAI Gateway (D5). The ASAI Gateway first tells the switch to connect the agent to the call (M2, V2). Then, after the agent is connected, the ASAI Gateway tells the switch to place the outgoing call (M3, V3).

For agent-initiated outgoing calls, the agent is involved in the call from start to finish. The agent initiates the call, listens as the call progresses through the local switch and the public or private network, and determines if the destination answers, does not answer, or is busy.

Call-Center-Software-Initiated Outgoing Calls

Two types of call-center-software-initiated outgoing calling are possible: agent classified and anticipatory dialing. The primary difference between these types of calls and agent-initiated outgoing calls is that the agent is not involved in the call from start to finish. The primary difference between the two types of call-center-software-initiated outgoing calling is the point at which the agent becomes involved in the call.

For agent-classified outgoing calls, the call-center software sends instructions for placing the outgoing call to the ASAI Gateway. The ASAI Gateway first tells the switch to connect an available agent to the call. Then, after the agent is connected, the ASAI Gateway tells the switch to place the outgoing call. The agent determines if the destination answers, does not answer, or is busy.

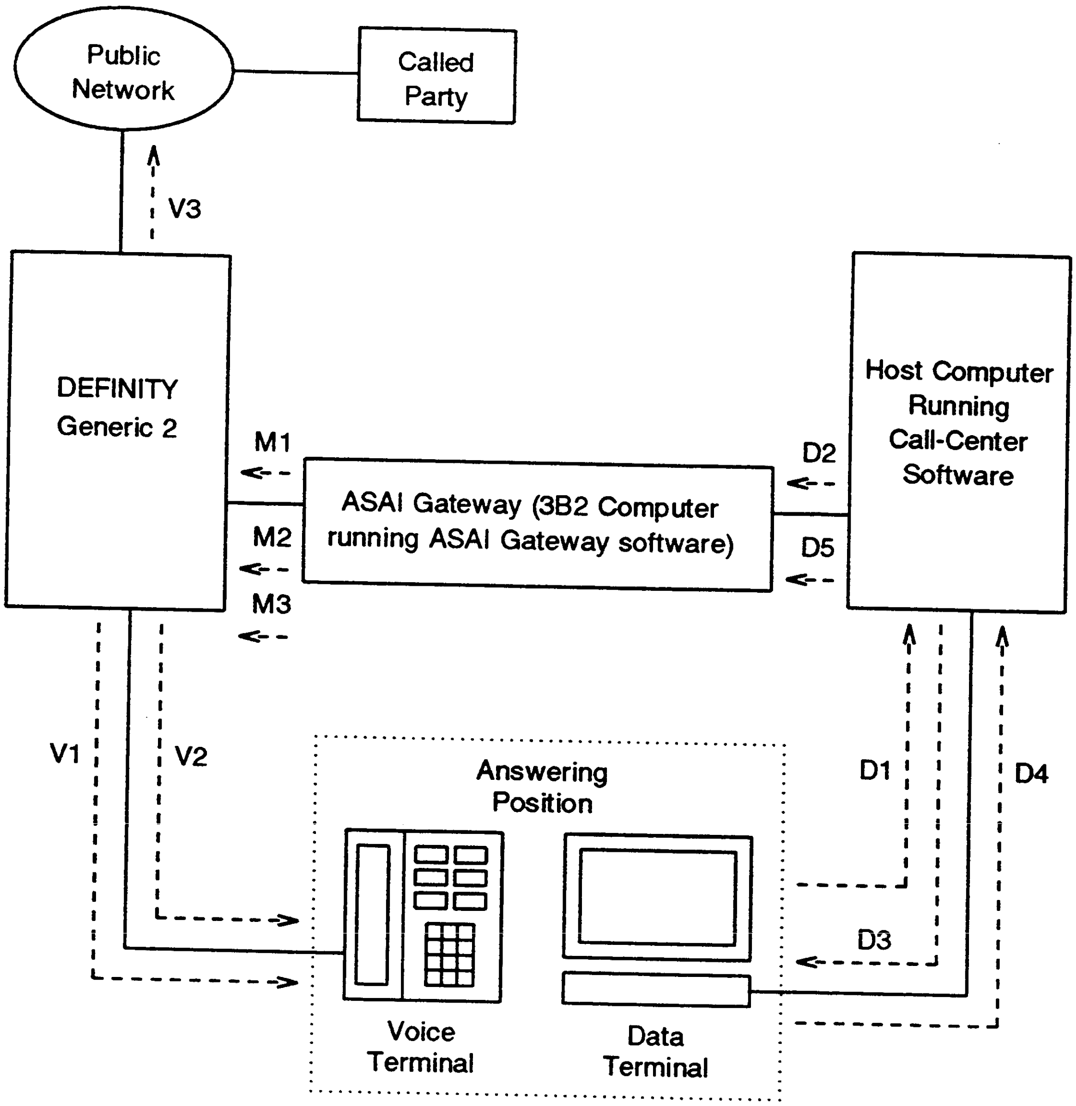


Figure 33-3. Outgoing ASAI Gateway Interface Call

With anticipatory dialing, the call-center software sends instructions for placing the outgoing call to the ASAI Gateway. The ASAI Gateway first tells the switch to place the outgoing call. Then, if the call is answered, the ASAI Gateway tells the switch to deliver the call to an available agent. As soon as an agent is selected, the call-center software can send database (called-party) information to the agent. For this type of outgoing call, the call-center software anticipates that an agent will be available when the call is answered.

Anticipatory dialing can increase agent productivity because agents do not have to wait while calls are being established or have to be involved in calls that are not answered. Unlike agent-initiated and agent-classified outgoing calls, with anticipatory dialing the ASAI Gateway (not the agent) determines if the destination does or does not answer.

Transfer/Conference Management

Transfer/conference management enables the call-center software to monitor an incoming or outgoing call and, if requested, transfer the call and the associated database information from one answering position to another. An answering position can also conference with another answering position with or without caller (or called-party) involvement. Or, the answering positions can exchange information before adding the caller (or called party) into the conference. That is, as part of the conference (or transfer) operation, the call-center software can put one or more parties on hold.

Transfer/conference management only applies to calls controlled by the call-center software that are in a stable 2-party state. Typically, a call transfer or call conference is initiated from an agent's data terminal, but a transfer/conference can be initiated from an agent's voice terminal.

If a call controlled by the call-center software is transferred to an external destination over ISDN—PRI facilities, call and database information can be sent with the call. If the receiving switch has the ASAI Gateway Interface feature, this information can be passed to the call-center software by way of the ASAI Gateway at the receiving switch. Sending call and database information from one location to another can reduce or eliminate the time required to retrieve the information again at the receiving switch. The call-center software at the receiving switch could, for example, be notified that the call has been transferred and could be given the caller's account number, which was retrieved from a database at the sending switch.

Sending call and database information from one location to another gives the agent at the receiving switch the information necessary to greet the caller properly. Instead of responding "Hello, how may I help you?", the agent at the receiving switch could respond "Hello Mr. Jones, I have the information you requested from our Denver office."

Considerations

Vector Processing Time

The ASAI Gateway Interface feature can use Call Vectoring to process an incoming call and to deliver the same call to an answering destination. A system that is confirmed this way may use twice as much vector processing time (per incoming call) as a system that only uses Call Vectoring to process incoming calls.

Call Traffic

To determine how much ASAI Gateway call traffic a particular system can support, factors such as current call traffic and processor occupancy and system limits must be considered. Refer to *ASAI Gateway System Description and Planning (585-246-201)* for more information.

NPA-NXX Designator

An NPA-NXX designator must be assigned to any local answering position that might receive calls controlled by the ASAI Gateway. This includes answering positions to which a call is transferred, forwarded, or redirected.

Call-Center Software

The call-center software is not part of the ASAI Gateway Interface feature. The call center's owner is responsible for developing or obtaining the call-center software, which can be extensive.

User Operations

The ASAI Gateway Interface feature does not limit any voice terminal user operations. Refer to the ACD feature and to other appropriate features for descriptions of voice terminal user operations.

Interactions With Other Features

Automatic Alternate Routing/Automatic Route Selection

For System 85 and DEFINITY Generic 2.1, AAR is used to route call-control messages between the switch and the ASAI Gateway. AAR, ARS, or both features can be used to route outgoing calls controlled by the call-center software.

Automatic Call Distribution

Multiple call handling works normally when used with the ASAI Gateway Interface feature. However, if the hold operation is initiated from the agent's voice terminal, information about the held call is not sent to the call-center software and, therefore, the call-center software cannot coordinate the delivery of caller information for subsequent voice calls. If the hold operation is initiated from the agent's data terminal or the call-center software, the multiple call handling function is not used.

A City-of-Origin announcement cannot be provided for incoming calls controlled by the call-center software because the incoming trunks (to the switch) do not terminate directly to an ACD split. However, the call-center software could provide similar information.

Some ASAI Gateway capabilities, for example reserving an agent for an outgoing calling session, require specific button configurations on agent voice terminals. Refer to *ASAI Gateway System Description and Planning (585-246-201)* for more information.

Automatic Circuit Assurance

Automatic Circuit Assurance should not be activated for an ASAI Gateway Interface trunk group.

Call Detail Recording

CDR should not be activated for an ASAI Gateway Interface trunk group. However, CDR can be activated for incoming and outgoing trunks (to and from the switch) that are used for ASAI Gateway calls.

An incoming call controlled by the call-center software that is delivered to a local answering position is recorded in the same way as an incoming call that is not controlled by the call-center software. An outgoing call initiated by a station (usually an ACD agent) and controlled by the call-center software is recorded in the same way as an outgoing trunk call that is not controlled by the call-center software.

Call Forwarding—Follow Me

A call controlled by the call-center software can be forwarded to a destination outside the switch. If the call uses facilities that are not end-to-end ISDN—PRI, the monitoring and control capabilities of the call-center software are reduced. The switch can only report the dialed number (to the call-center software) and monitor answer supervision and disconnect supervision. The ability to monitor transfers (requested by an answering position at the receiving switch) is lost.

Call Vectoring

The ASAI Gateway Interface feature can use Call Vectoring to process an incoming call and to deliver the same call to an answering destination. Because a call may encounter more than one VDN, the commands that make up the vectors associated with the VDNs should be coordinated so that they provide the appropriate calling-party treatment.

The vector that processes incoming calls can be programmed with alternate answering destinations in case an ASAI Gateway or an ASAI Gateway Interface trunk group fails. Alternate answering destinations may include local extensions or ACD splits, remote extensions or ACD splits, other VDNs, and other ASAI Gateway Interface trunk groups to the same or a different ASAI Gateway.

ISDN—PRI (Primary Rate Interface)

At least one ASAI Gateway Interface (ISDN—PRI) trunk group is required for each ASAI Gateway.

Look-Ahead Interflow

A call controlled by the call-center software can be routed to a destination outside the switch using the Call Vectoring, Look-Ahead Interflow, and ISDN—PRI features. For this type of call, the sending switch's ability to monitor transfers (requested by an answering position at the receiving switch) is lost.

If more than one vector is used to process Look-Ahead Interflow calls at the receiving switch, the first vector should accept or reject calls, rather than the vector that routes the call to an answering position.

Restriction—Voice Terminal Restrictions

A voice terminal assigned inward restriction in its line class of service cannot receive incoming calls controlled by the call-center software. A voice terminal assigned outward restriction in its line class of service can be used to initiate outgoing calls controlled by the call-center software.

World Class Routing

Beginning with DEFINITY Generic 2.2, WCR is used to route call-control messages between the switch and the ASAI Gateway. The WCR feature can also be used to route outgoing calls controlled by the call-center software.

Hardware Requirements

At least one ASAI Gateway Interface (ISDN—PRI) trunk group is required for each ASAI Gateway. Depending on call traffic, a dedicated traditional or universal module may be required.

Traditional Modules

One circuit pack is required for each ASAI Gateway Interface trunk group.

- ANN35 ISDN Primary Rate Port Circuit Pack

As many as 4 ISDN—PRI circuit packs (ANN35) can be assigned to the system.

Universal Modules

Two circuit packs are required for each ASAI Gateway Interface trunk group.

- TN767 DS1 Interface Circuit pack

  The TN767 provides the ISDN D-channel and B-channels (B-channels are not used for the ASAI Gateway Interface feature).

- TN555 DS1 Packet Adjunct

  The TN555 must be used with the TN767 to support ISDN type signaling. Without the TN555, the TN767 is not capable of handling the ISDN message oriented signaling.

As many as 4 ISDN—PRI interfaces (TN767 and TN555) can be assigned to the system.

Regardless of Module Type:

- An AT&T 3B2 Computer running ASAI Gateway software is required.

  An AT&T 3B2 Model 600, Model 700 or Model 1000 is required to run the ASAI Gateway software. The 3B2 computer requires an X.25 interface to communicate with the host computer.

Feature Administration

Assignment of the ASAI Gateway Interface feature is on a per-system, and per-trunk group basis.

On DEFINITY Generic 2, this feature is administered using DEFINITY Manager II.

This feature can also be administered using the Manager IV.

At least one ASAI Gateway Interface (ISDN—PRI) trunk group must be administered for each ASAI Gateway. Refer to the ISDN—PRI feature description for information about administering the ISDN—PRI feature.

The Call Vectoring and AAR (WCR beginning with DEFINITY Generic 2.2) features are used to process incoming calls. Network routing features (AAR, ARS, and WCR) can be used to route outgoing calls. Refer to the appropriate feature descriptions for information about administering these features.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES ASAI GATEWAY INTERFACE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	4	Assign an NPA-NXX designator to a group of extension numbers. This procedure is used with Procedure 354, Word 3 to provide connected party information (the answering position that handled the call) to the call-center software (by way of the ASAI Gateway).	Yes
100	1	Assigns the trunk type to an ASAI Gateway Interface trunk group. The applicable encode is: 47 TIE ETN 2-way dial repeating.	No
100	3	Assigns the signaling type to an ASAI Gateway Interface trunk group. The applicable encode is: 20 Digital multiplex interface ISDN message-oriented signaling. Field 8 (Optional ISDN Information Inhibited) must be set to zero.	No
100	7	Associates a trunk-group number with the equipment location of an ISDN—PFU board and allocates software trunk records.	No
103	1	Assigns network features and capabilities to a trunk group.	Yes
275	1	Assigns features to the system class of service. Tandem Tie Trunk and Trunk-Trunk Calling should be assigned.	Yes
276	1	Assigns ASAI Gateway Interface (called Integrated Telemarketing Gateway Interface in DEFINITY Generic 2.1) to the feature group class of service.	No
354	3	Associates an NPA-NXX designator with an NPA, NXX, and thousand's digit. This procedure is used with Procedure 000, Word 4 to provide connected party information (the answering position that handled the call) to the call-center software (by way of the ASAI Gateway).	No

Call Vectoring

Description

The Call Vectoring feature is an enhanced and highly flexible way of processing incoming calls to the System 85 or DEFINITY Generic 2 switch. Vectors are the basis of the Call Vectoring feature. These vectors are programmed using methods that resemble a "high-level" programming language. Using a "vector" (a discrete set of predefine call-processing steps), the customer can design appropriate and desirable ways of treating specific incoming calls.

Calls access vectors using VDNs (vector directory numbers). A VDN is a "soft" extension number that is assigned an internal line number but is not assigned to an equipment location. (Each VDN can be published to enable public access to a vector's call-processing sequence.) Vectors are assigned to VDNs. In turn, these VDNs can either be preassigned to incoming (or 2-way) trunk groups or passed in digit form to the System 85 or DEFINITY Generic 2 by the serving switch.

Since more than one VDN can terminate to the same vector, the answering party can respond appropriately to the call with knowledge of the dialed number. Moreover, since a set of vectors can terminate to the same answering destination, the call-processing sequence can vary according to the vector reached. These ideas are shown in Figure 34-1.

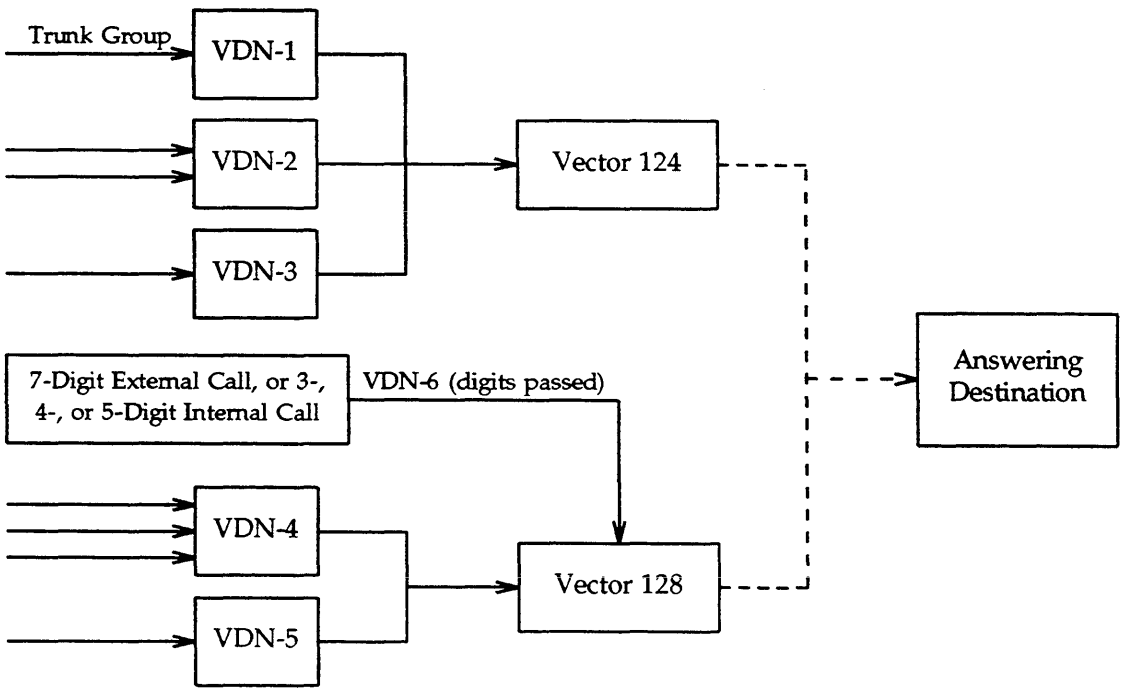


Figure 34-1. Trunk Groups, VDNs, Vectors, and Answering Destinations

Feature History and Development

This feature was first available for System 85 in Release 2, Version 4.

Beginning with R2 V4, Issue 1.3 and DEFINITY Generic 2, the Look-Ahead Interflow feature can be used in conjunction with ACDs that ***also*** use the Call Vectoring and the ISDN (Integrated Services Digital Network)/PRI (Primary Rate Interface) features (Refer to the Look-Ahead Interflow chapter of this manual for a detailed description of this operation.)

Beginning with Issue 1.3 of R2 V4 System 85, and DEFNITY Generic 2, the flexibility is increased for associating names with direct VDN calls. An option called VDN Override can be assigned to a VDN. When VDN Override is assigned, the name associated with the originally called VDN can change as vector processing diverts the call.

Beginning with Issue 3.0 of DEFINITY Generic 2.1, an RLT (Release Link Trunk) can terminate to an ACD split, VDN (Vector Directory Number), or attendant console. Before this enhancement, an RLT could only terminate to an attendant console.

The following changes apply to the Call Vectoring feature beginning with DEFINITY Generic 2.2:

- 511 Vectors Per System

  The number of vectors that can be administered per system increases from 128 to 511.

- Go To Vector Command

  This new command enables vector processing to branch to a different vector, similar to the way the "go to step" command enables vector processing to branch to a different step in the same vector.

- 255 Recorded Announcement Trunks

  The number of recorded announcement trunks increases from 84 to 255.

- 475 Abbreviate Dialing Group-List Items

  The number of Abbreviated Dialing group-list items (used by the "route-to" command) increases from 95 to 475 (from 1 group list to 5 group lists).

- Administration Changes

  Beginning with DEFINITY Generic 2.2, Procedure 030, Word 3 is used to program the steps of a vector and Procedure 030, Word 2 is used to transfer a vector from the scratch pad to permanent memory. Prior to DEFINITY Generic 2.2, Procedure 030, Word 3 was used to program the steps of a vector and to transfer a vector from the scratch pad to permanent memory.

  The allowed values for the "wait" command (called "delay" prior to DEFINITY Generic 2.2) have been changed from even values between 2 and 998 seconds to even values between 0 and 998 seconds.

Related Features

The following System 85 and DEFINITY Generic 2 features are commonly used with the Call Vectoring feature.

- ASAI Gateway Interface
- Automatic Call Distribution
- Expert Agent Selection
- Look-Ahead Inteflow

Vectoring in the General System 85/DEFINITY Generic 2 Environment

In the general System 85/DEFINITY Generic 2 environment, the Call Vectoring feature can be applied in several beneficial ways. Some of these applications represent new or enhanced functionality. Others represent alternate, sometimes simpler, ways to replace existing functionality. A representative list of these applications follows. See "Sample Applications of Vectoring" for examples of these and other vectoring applications.

- Information announcements for calling party

  The human intervention needed to distribute common messages can be minimzed with information announcements. A group of people with common interest can be instructed to call a specific number (VDN) that terminates to a specific announcement vector. The vector's announcement can be periodically updated to provide current information to the callers. Vectors providing information announcements are easily programmed. Refer to the pair of vectors under Vector J, "Providing an Information Announcement for Callers," for examples of this type of vector.

- Vector processing before routing to the attendant queue

  Two realistic applications for vector processing attendant calls include "forced first announcement" and "customized night service." A forced-first-announcement vector is shown in Vector E, and a night-service vector is based on Vector G.

- Night Service for Message Center

  A vector can be programmed to provide automatic AUDIX "night coverage" for calls that would otherwise redirect to an "unstaffed" Message Center split The VDN that terminates to this vector is assigned as the final point in the principals' coverage paths. In this way, redirected calls automatically cover to AUDIX Call Answering at night, while covering to Message Center during work hours. Refer to Vector I, "Using AUDIX to Provide Night Service for Message Center," for an example of this type of vector.

- Recent-Disconnect Announcements and Tenant Services

  Multiple "recent-disconnect" announcements can be desirable for a partitioned switch (for example, "You have reached a disconnected number of the ______ Corporation."). When Call Vectoring is used on a partitioned switch, as many as 84

(255 beginning with DEFINITY Generic 2.2) Differenct recent-disconnect announcements can be provided.

Whenever a voice terminal is taken out of service, the voice terminal can be removed using Procedure 000, Word 1 or 052, Word 1. Once the voice terminal is removed, the extension number is temporarily assigned as a VDN (in Procedure 000, Word 1) that points to a specific partition's "recent-disconnect" vector (Procedure 031, Word 1). Each recent-disconnect vector would contain a single "forced disconnect with announcement" step that specifies the actual tenant called, and provides that tenant's LDN (for example, "You have reached a disconnected number of the Jericho Company. For assistance, please call 737-2100.")

- Call Coverage to the Attendant Queue

  Call Vectoring can redirect coverage calls to the attendant queue. Attendant coverage can be beneficial for some System 85s and DEFINITY Generic 2s. Using this coverage, the attendant group can serve as the final coverage point for one or more principals.

  A VDN can be assigned as the final point in a coverage path. One of these VDNs can be assigned to a vector with a single "route to" step. The "route to" step within this coverage vector contains an Abbreviated Dialing list item that outpulses the attendant dial access code [usually "0," or an LDN (Listed Directory Number)].

  Since "route to" steps can direct calls to LDNs, partitioned switches can also cover to the shared attendant queue. Each extension partition desiring attendant coverage can have a vector that directs calls to the LDN for the attendant partition assigned to that extension partition. In this way, attendant coverage is a partitioned function of the Tenant Services environment.

ACD (Automatic Call Distribution) in a Call Vectoring Environment

One of the primary applications of the Call Vectoring feature is to provide customized treatment for incoming ACD calls. For systems that have the ACD and Call Vectoring features, the following standard ACD functions (that is, ACD without the Call Vectoring feature) are replaced by similar but more flexible Call Vectoring capabilities:

- Associating trunk groups with splits
- Routing calls to ACD splits using associated extension numbers
- Providing a first delay recorded announcement for a split
- Providing a second system-wide delay recorded announcement
- Providing intraflow (redirection of ACD calls to a local destination) from a split's queue by way of the Call Forwarding feature
- Providing interflow (redirection of ACD calls to a remote destination) from a split's queue by way of the Overload Balancing function or the Call Forwarding feature
- Specifying a split number (from 1 to 60) as the final point in a coverage path.

The more flexible and customized calling-party interface provided by Call Vectoring can be desirable when:

- The agents in a split answer calls for more than one purpose.

  This need can be addressed by assigning a set of VDNs and/or trunk groups (each controlled by a discrete vector) that terminate to a single ACD queue. Since these VDNs and trunk groups are controlled by different vectors, a unique announcement can be provided for each type of call. In this way, an appropriate announcement can be provided for each of the different types of calling parties.

- A split experiences peak periods of incoming calling activity.

  Vectors can be programmed to provide a variety of treatments for a calling party. For example, during periods of heavy call traffic, these calls can be redirected based on the number of available agents or the oldest call wait time. Also, Call Vectoring can limit the number of calls in an ACD queue. When this limit is reached, the switch can disconnect, return busy tone, or redirect calls to another split or other answering position.

- An automatic form of night service is desired for ACD splits.

  A vector can be programmed to provide automatic night service for ACD splits (or the attendant queue). Callers automatically receive a "night service" announcement when these calls are placed while the split is off duty. Otherwise, during normal work hours, calls to the split are processed in a normal and appropriate way. Refer to Vector G, "Providing Conditional Night Service for the Attendant Queue or an ACD Queue," for an example of this type of vector.

- A split's agents answer emergency calls, and queue limiting is desired.

  During widespread emergencies, incoming calls to emergency numbers produce bursts of heavy calling activity. Meanwhile, more localized emergencies produce a much lower steady volume of calling activity. To address this call-answering scenario, an ACD split can be provided to answer these calls. The vector that queues calls to this emergency split can be programmed to encourage callers to hang up when a predefined number of queued calls is reached. The limit would need to be considerably higher than the prevailing steady volume and yet considerably lower than the volumes that could be reached in a widespread emergency. In this way, these agents can routinely handle localized emergencies in an effective manner, and yet not be flooded with many calls reporting a known problem. Refer to Vector B, "Providing an ACD Split to Handle Emergency Calls," for an example of this type of vector.

  Once the specific emergency is known, another vector can be programmed to replace the queue limiting vector. This vector contains a more specific announcement (for example, "We are aware of the power outage in Plainfield.") that assures the caller that his/her specific problem is being addressed. Given this assurance, the caller is more likely to hang up. Refer to Vector C, "Providing a Specific Emergency Announcement," for an example of this type of vector.

  NOTE: Call Vectoring does not provide queue limiting for the attendant queue.

- An enhanced form of priority queuing is desired for ACD queues.

  The ACD feature without Call Vectoring provides two levels of priority queuing, but the Call Vectoring feature offers up to four levels of entry to an ACD queue (including AUDIX and Message Center queues). The four levels of entry include 0 ("low" priority), 1 ("medium" priority), 2 ("high" priority), and 3 ("top" priority). Using these four levels (selected in Procedure 030, Word 3, Field 6), the switch administrator can give preferential answering treatment to certain incoming calls based on various criteria. These criteria might include the cost of various trunking facilities, the amount of revenue generated by various calls, and courtesy to executive personnel.

  To implement an ACD queue with four levels of entry, there would usually be four vectors that queue calls to the same split. The "queue to main split" step in each vector is assigned a different level of priority from 0 (low priority) to 3 (top priority). In turn, each vector is assigned to a different VDN that is either passed to call-processing in digit form or assigned to an incoming trunk group.

Methods of Routing Incoming Calls into Vector Processing

As previously mentioned, there are two ways to route incoming VDN calls to the System 85 or DEFINITY Generic 2 switch. These two methods include:

- Digit-oriented routing

  Using this method, a VDN's digits are passed through the serving switch (usually, the serving Central Office) and to the local switch in a manner similar to the way that DID calls are routed. As the VDNs digits are analyzed by the System 85's or DEFINITY Generic 2's call-processing software, the dialed number is recognized as a VDN that terminates to a specific vector. In turn, the call-processing software gives control of the incoming call to vector processing.

- Trunk group-oriented routing

  Using this method, a VDN call is recognized by the serving switch (usually, the serving central Office) as a call that is routed to the local System 85 or DEFINITY Generic 2 over a specific trunk group. In turn, the local switch accepts the call from over the incoming (or 2-way) trunk group, and recognizes this as a call assigned to terminate to a specific VDN. Likewise, this VDN terminates to a specific vector. So far, this method of routing resembles "non-DID routing" to the attendant queue, or "automatic-in routing" to an ACD split's queue. However, Call Vectoring can add an important element to this linkage. The vector that assumes control of the VDN call can be programmed with a "route to" command as the first step in the vector, In this way, an "automatic-in VDN call" can terminate to an assortment of destinations that previously could not be accessed using trunk group-oriented routing.

The trunk types that can be assigned to terminate to a VDN in Procedure 031, Word 2 include:

— 16 = CO 1-way in attendant-completing

— 19 = CO 2-way attendant-completing in/DOD out

— 20 = CO 2-way with party test attendant-completing in/DOD out

— 21 = FX 1-way in attendant-completing

— 24 = FX 2-way attendant-completing in/DOD out

— 25 = FX 2-way with party test attendant-completing in/DOD out

— 26 = WATS 1-way in attendant-completing

— 35 = TIE 1-way in automatic

— 38 = TIE 2-way automatic in/dial repeating out

— 39 = TIE 2-way automatic in and out

— 50 = Remote Access 2-way.*

— 66 = CAS release link trunk 1-way incoming at main.

Figure 34-2 shows a simplified drawing of the alternate routing methods.

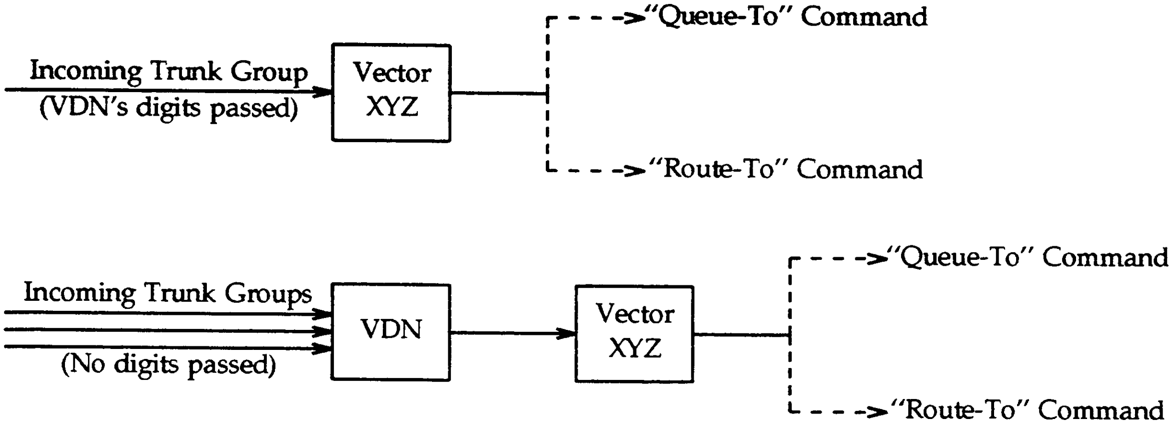


Figure 34-2. Methods of Routing Incoming VDN Calls

* Used only for Remote Access speaker verification.

DNIS (Dialed Number Identification Service)

In the initial availability of DNIS, ACD agents equipped with a display voice terminal (for example, CALLMASTER, 7405D with a display module, 7406D With Display, 7407D, 7506, or 7507) receive visual displays that specify the dialed number for calls terminating to the agents' voice terminals.

In traditional ACD arrangements, groups of agents are organized into "splits" (functional groups of answering positions). Using this approach, an agent is trained to answer calls for one specific purpose in an efficient and professional manner. However, ACD managers are recognizing the need to relax this concept of limiting each split to a single call-answering task.

The alternative is to provide splits where each group of agents is proficient with several types of calls. The desired gain is to provide adequate service for the several call types with fewer agents and with less administrative intervention by the ACD manager. Using this approach, the changing staffing needs of the several call types are averaged in time, and enough agents are staffed to provide adequate service for the prevailing average load. Where five agents might be needed in each of 3 smaller splits (15-agent total) to handle 3 types of calls, only 11 or 12 agents might be needed in the single (more general) split.

This idea of averaging the call-handling load is sound for certain applications, but the goal of improved agent efficiency is more readily achieved with the DNIS capability. The DNIS function of the Call Vectoring feature allows each answering agent to know the purpose of each incoming call as the call terminates to the agent's voice terminal. As a result, the natural efficiencies of the single split/single call type arrangement are not compromised. With the calling number display provided by DNIS, agents are aware of each call's purpose, and can answer each incoming call with the appropriate greeting. Agents need not invest time merely to determine the purpose of calls.

Table 34-A shows sample displays that Call Vectoring DNIS might provide to an ACD agent.

TABLE 34-A. DNIS Display Information

Type of Call	Display
Inside call	a=R JONES to CLAIMS
Outside call	a=OUTSIDE CALL to SALES
ISDN call	a=212-291-7733 to SERVICE

Configuration of Call Vectoring DNIS

Call Vectoring provides a simple and direct means of providing the DNIS functionality. In Procedure 012, Word 1, a distinct and appropriate name is assigned to each VDN and/or trunk group that directs incoming calls to the split's queue.

"Route To VDN" Commands and Voice Terminal Displays

Prior to R2 V4, Issue 1.3, the voice terminal displays for incoming calls to VDNs were permanently associated with the call. Whenever a call was placed directly to a VDN, the switch initially tagged the call with the called-party name assigned (in Procedure 012, Word 1) to the VDN or to the VDN's incoming trunk group. However, once the called-party name was associated with a call, this name was not allowed to change even if vector processing subsequently diverted the call to a different VDN with a "route to VDN" command.

Maintaining the name of the originally called VDN served the useful purpose of preserving the original DNIS information for diverted calls. In this way, a voice terminal display would still show the answering party the original purpose of diverted calls.

VDN Override

Beginning with Issue 1.3 of R2 V4 System 85 and DEFINITY Generic 2, the flexibility is increased for associating names with direct VDN calls. An option called VDN Override can be assigned to a VDN (in Procedure 031, Word 1, Field 9). When VDN Override is assigned, the name (assigned in Procedure 012, Words 1 and 2) associated with the originally called VDN can change as vector processing diverts the call.

For example, VDN Override could be used in conjunction with a vector that routes the *older* calls in queue to a VDN controlled by a different vector. Perhaps, calls that have exceded 90 seconds of vector processing in one vector are routed to a special VDN with a different vector that queues these calls to an ACD split at top priority. In this example, VDN Override can be assigned to the first VDN. Then, as the original vector executes the "route to VDN" step, the switch can associate the new VDN's name with the call. Therefore, the agent in the alternate split who answers the call knows that the calling party has been waiting for a long time and can handle the call accordingly.

> **NOTE:** When assigned, VDN Override only applies to calls placed directly to a VDN. VDN Override does not apply to calls that are redirected to a VDN by the Call Coverage or the Call Forwarding—Follow Me feature. For calls that are redirected to a VDN, the originally called voice terminal's name remains permanently associated with the call.

Definitions of the Ten Vector Commands

A set of ten vector commands is provided for the Call Vectoring feature. A list of these commands and their definitions follows.

- Queue to main split

  Queue the call to the specified main split at the specified priority. This command is ***unconditional.*** (After a call is queued, regular ACD software periodically checks the main split's status to determine whether the split is staffed. If the split is staffed, ACD software periodically scans for an available agent. If an agent is available, the call at the head of the queue is connected to the available agent.) Therefore, if an agent is available at the time a call is queued, regular ACD software quickly distributes the call to the available agent. If no agents are available at the time the call is queued, the caller can immediately begin to advance in queue. If the split is not staffed at the time the call is queued, the Call Vectoring software ***still queues*** the call, and vector processing moves to the next step.

> **NOTE:** To avoid queuing calls to an unstaffed ACD split, program a conditional "go to step" command (that checks for less than one staffed agent) before the "queue to main split" command so that the "queue to main split" command is bypassed in this condition.

Multiple "queue to main split" commands are allowed in the same vector. However, VDN calls are not allowed to be queued to more than ***one split*** at a time.

When vector processing encounters a second "queue to main split" command, the call is removed from the first split's queue and then requeued to the second split. (The second split could actually be the same split as the first.)

As a "queue to main split" command distributes a call to an ACD agent with automatic answering the switch delivers a 1-burst zip tone to the answering agent. This single burst of zip tone designates a call dialed directly to the agent's split.

Also, when a "queue to main split" command distributes a call to an ACD agent and a VDN-, city-, or queue-of-origin announcement is assigned, the switch delivers the announcement to the answering agent.

- Check backup split

  While a call is queued to a main split, check the specified backup split at the specified priority for one of the following:

  — Number of available agents in the backup split

  — Number of staffed agents in the backup split

  — Number of queued calls in the backup split's queue

  — Amount of time that the oldest call in the backup split's queue has waited to be answered.

  A "check backup split" command can specify only one of these conditions, however, a vector may contain more than one "check backup split" command.

  > **NOTE:** Whenever a vector has one or more "check backup split" commands, make sure that calls have been queued by a previous "queue to main split" command.

  Once a "check backup split" command is executed, the backup split is tested at 2-second intervals until:

  — The condition specified in the command is met, or

  — The call is answered at the primary answering destination, or

  — The calling party abandons the call.

If the condition specified in the "check backup split" command is met, the call is removed from queue and placed in the backup split's queue.

The condition specified in a "check backup split" command may not be tested for up to 2 seconds. If a "forced disconnect," "route to," "go to vector," or "forced busy" command immediately follows a "check backup split" command, calls could be disconnected, routed, or receive busy tone before the condition is tested. To avoid this problem, place a "delay (wait)" command between the "check backup split" command and a "forced disconnect," "route to," "go to vector," or "forced busy" command. A delay (wait) interval of 2 seconds or more will insure that the condition specified in the "check backup split" command is tested before the next step in the vector is executed.

A "check backup split" command that tests a split queue based on the number of calls or the oldest call wait time can specify a priority level. The test considers only those calls with the specified priority level or higher; calls with a lower priority level are not considered. To test all calls in a split queue, the "check backup split" command should specify low priority.

As a "check backup split" command distributes a call to an ACD agent with automatic answering, the switch delivers a 2-burst zip tone to the answering agent. These two bursts of zip tone designate a call originally dialed to a different split within the local switch. These calls are considered as "intraflowed" calls since a "check backup split" command delivered the calls to the alternate split.

Also, when a "check backup split" command distributes a call to an ACD agent and a continuous (that is, VDN-, city-, or queue-of-origin) announcement is assigned, the switch delivers the appropriate announcement to the answering agent.

- Route to

Route the call to:

  - A local extension number
  - The attendant queue
  - The CAS attendant queue
  - A Host Computer Access trunk group
  - Another VDN
  - A remote location [using the AAR, ARS, WCR (World Class Routing), DCS, or Main/Satellite feature].

An Abbreviated Dialing group-list item is specified as part of the administration for a "route to" command. Whenever the "route to" command is executed, the digits stored in the list item are obtained and then used to route the call. If a list item called by a "route to" step contains either an invalid destination or no destination, vector processing either treats the step as a "stop" step (if the final effective step) or skips the step and continues processing with the next sequential vector step.

If a "route to" command is the final effective step* in a vector and the command fails due to a resource failure (for example, no Originating Register or trunks available), the step is retried at 2-second intervals. This can have a significant impact on processor occupancy and, if possible, should be avoided. For more information about the criteria for success or failure of the "route to" and other vector commands, refer to Table 34-B.

Beginning with DEFINITY Generic 2.2, the number of Abbreviated Dialing group-list items that can be used for Call Vectoring has been increased from 95 (1 group list) to 475 (5 group lists).

An extension can be designated as the "controlling extension" for a Call Vectoring group list. When this is done, the user of the controlling extension can change the destinations of "route to" commands without using the Manager II, MAAP, or SMT.

Each vector can contain as many as 15 call-processing steps. This 15-step maximum can be exceeded by "chaining" two or more vectors together. This can be done by programming a "route to VDN" step as the final step in each vector (except the last vector in the chain). The route to VDN" step serves as the "link" between successive vectors. The VDN specified in the "route to VDN" step of a vector is administered to terminate to the next vector in the chain. In this way, two chained vectors could provide as many as 29 call-processing steps. Three chained vectors could provide as many as 43 call-processing steps.

> **NOTE:** When "route to VDN" steps are used to chain vectors, these commands are not allowed to route calls back to the same vector. When this is attempted, vector processing ignores the request, and continues processing with the next vector step. If the "route to VDN" step is the last step in the vector (or immediately followed by a "stop" step), vector processing treats this "route to VDN" step as a "stop" step.
>
> To repeat execution of the same vector, use the command "go to step 1." This command has a similar effect to the disallowed "route to" command and uses less call-processing time.

When "route to VDN" steps are used to chain vectors, vector processing stops for the current vector, and then assumes control of the continuation vector. As a result of this discontinued processing, calls are removed from any queue they are in, and any scanning invoked by a "check backup split" command also stops. Therefore, whenever a call should be queued or whenever scanning should proceed in the continuation vector, the "queue to main split" ***and*** "check backup split" step(s) ***must be repeated*** as steps in the continuation vector. Delay (wait) treatment established in the current vector carries over to the continuation vector.

* A "final effective step" of a vector is either the last vector step or a vector step that is followed by a "stop" step.

Beginning with DEFINITY Generic 2.2, the "go to vector" command provides a better way to chain vectors, because calls that were queued by the current vector (or a previous vector in the chain) remain queued when a "go to vector" command is executed.

NOTE: For more information about the "route to" command, refer to the Look-Ahead Interflow chapter of this manual.

- Announcement

Connect the call to the specified "delay" recorded announcement.

If answer supervision has not already been returned for an incoming call, this signal is sent to the serving switch just before vector processing executes an "announcement" step.

Based on time-slot and TMS-blockage considerations, as many as 255 callers per module can listen to the same recorded announcement at the same time.

The 13A digital announcement system, single-channel digital announcer (KS-65270), or 4-channel digital announcer (KS-65272) can be used to store "delay" recorded announcements. Refer to the DEFINITY Generic 2 and System 85 System Description (555-105-201) for more information about recorded announcement sets.

- Delay (beginning with DEFINITY Generic 2.2, the name of this command changes to "wait")

Delay vector processing for a specified number of seconds while the caller hears silence, ringback, or music.

If answer supervision has not already been returned for an incoming call, this signal is sent to the serving switch just before vector processing executes a "delay (wait) ***with music***" step.

The allowed values for delay intervals are even numbers between 2 and 998 seconds. Beginning with DEFINITY Generic 2.2, the allowed values for wait intervals are even numbers between 0 and 998 seconds. If a wait interval of 0 seconds is specified, caller feedback (silence, ringback, or music) begins and vector processing immediately continues with the next vector step.

If the "delay (wait)" step is a ***final effective step*** of the vector, the caller feedback continues beyond the specified delay (wait) interval. For these "delay (wait)" steps, ringback or music continues until the call is either answered or abandoned.

For calls routed to vectors from over CO trunks, the calling party hears the initial ringback from the Central Office (not from vector processing). However, it is a good idea to design and program vectors as if the caller feedback were always provided by the vectors. In this way, every vector can be more generally applied. When a vector is programmed to provide ringback that is actually provided by the

CO, no harm is done. The time slot's ringback is ignored by the CO. But now, this same vector can be applied, for example, to internal calls and DID trunks.

- Go to step

Go to another step in the same vector and continue vector processing at the specified step. This command provides conditional or unconditional branching.

When conditional, the allowed conditions include:

— The time of day and day of week

— The number of available agents in the specified or default split

— The number of staffed agents in the specified or default split

— The number of queued calls in the specified or default split's queue

— The amount of time that the oldest call in the specified or default split's queue has waited to be answered.

> **NOTE:** The default split (that is, no split is specified) is the split the call is currently queued to.

For all conditions except time of day, if a call is not queued and no split is specified, the test fails and processing continues with the next vector step.

A "go to step" command that tests a split queue based on the number of calls in queue or the oldest call wait time can specify a priority level. The test considers only those calls with the specified priority level or higher; calls with a lower priority level are not considered. To test all calls in a split queue, the "go to step" command should specify low priority.

- Go to vector

Go to another vector and continue vector processing at step 1 of the specified vector. This command is available beginning with DEFINITY Generic 2.2 and provides conditional or unconditional branching.

When conditional, the allowed conditions include:

— The time of day and day of week

— The number of available agents in the specified or default split

— The number of staffed agents in the specified or default split

— The number of queued calls in the specified or default split's queue

— The amount of time that the oldest call in the specified or default split's queue has waited to be answered.

> **NOTE:** The default split (that is, no split is specified) is the split the call is currently queued to.

For all conditions except time of day, if a call is not queued, the test for these conditions fails.

The "go to vector" command can be used to chain two or more vectors together. Like the "route to VDN" command, check-backup split scanning stops when a "go to vector" command is executed. Therefore, whenever scanning should proceed in the continuation vector, the "check backup split" step ***must be repeated*** in the continuation vector. However, unlike the "route to VDN" command, calls that were queued by the current vector (or a previous vector in the chain) remain queued when a "go to vector" command is executed. Also, delay (wait) treatment established in the current vector carries over to the continuation vector.

A "go to vector" command that tests a split queue based on the number of calls or the oldest call wait time can specify a priority level. The test considers only those calls with the specified priority level or higher; calls with a lower priority level are not considered. To test all calls in a split queue, the "go to vector" command should specify low priority.

- Forced disconnect

  Disconnect the calling party from the switch. (An optional disconnect announcement is available.)

  Without optional announcement

  If answer supervision was not previously returned for an incoming call, the System 85 or DEFINITY Generic 2 sends an answer supervision signal to the non-ISDN serving switch, waits 4 seconds, and then disconnects the call. This operation is necessary because switching systems cannot disconnects calls that have not been "answered." The 4-second interval is necessary because some switches do not recognize a disconnect signal immediately following an answer supervision signal Incoming toll calls to which this disconnect operation is applied are billed the minimum applicable charge. The customer is billed for each call that uses 800 Service trunks. The individual calling parties are billed for calls that use other trunk types.

  With optional announcement

  If answer supervision has not already been returned for an incoming call, this signal is sent to the serving switch just before vector processing executes the announcement option.

- Forced busy

  Return busy tone to the calling party (except with automatic CO trunks). Callers on automatic CO trunks do not hear busy tone from the switch. Instead, these callers continue to hear ringback from the CO.

  Answer supervision is ***not*** returned as this step is executed.

- Stop

  Stop vector processing (of additional steps) for this call.

  If the call is queued to a split, the call remains in the queue. If a "check backup split" command was previously executed, scanning continues for the call. If a "delay (wait)" step is active when the "stop" step is encountered, the specified treatment (that is, music, silence, or ringback) continues until the call is either answered or abandond.

Table 34-B shows the success and failure criteria and the vector processing disposition for each of the ten vector commands. Notice that the success and failure criteria and vector processing disposition for attendant calls are, for some vector commands, different from trunk and voice terminal calls. Attendant calls are processed differently to prevent them from waiting in queue. Also, notice that if a call's destination is a station with Call Coverage and the switch determines that the call should go to coverage, each point in the station's coverage path is checked to determine if it can accept the call. If a VDN is one of the coverage points, the associated vector is scanned and, if the vector contains one or more of the following commands, the call covers to the VDN:

- Queue to main split (with staffed agents)
- Route to
- Forced disconnect with announcement (beginning with R2 V4, Issue 1.2).

The principal's voice terminal continues to ring and the call does not go to coverage, if the vector does not contain one of these commands.

TABLE 34-B. Criteria for Success/Failure of Vector Commands

Call Type	Command	Succeed/Fail Criteria	Vector Processing Disposition
Trunk or voice terminal calling VDN directly	**Queue to main split**	Always succeeds.	Continue vector processing with next sequential step.
	Check backup split	Always succeeds.	Continue vector processing with next sequential step.
	Route to *		
	ATTENDANT	Always succeeds.	Exit vector processing. Pass control to processing for attendant queue.
	CAS	Always succeeds.	Exit vector processing. Pass control to CAS processing.
	VDN	Fails if call would route to the ***same*** vector.	If final effective step in vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
		Fails if routed-to VDN does not terminate to a vector.	If final effective step in vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
		Otherwise, succeeds.	Exit previous vector processing (if applicable, remove call from queue and stop backup split scanning). Pass control to new vector.

* Tenant Services partitioning checks are made.

TABLE 34-B. Criteria for Success/Failure of Vector Commands (Contd)

Call Type	Command	Succeed/Fail Criteria	Vector Processing Disposition
Trunk or voice terminal calling VDN directly (Contd)	**Route to** * (Contd)		
	EXTENSION	Fails if extension is active.	If final effective step in vector, check for an idle appearance, idle forwarding destination, or idle hunt-group destination. Step is retried at 2-second intervals. Otherwise, continue vector processing with next sequential step.
		Otherwise, succeeds.	Exit vector processing. Pass control to call processing.
	OFF-SWITCH †	Fails due to a resource failure (for example, no Originating Register or no trunks).	If final effective step in vector, retry at 2-second intervals. Otherwise, continue vector processing with next sequential step.
		Fails due to unknown destination or insufficient FRL to access outgoing trunks	If final effective step in vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
		Otherwise, succeeds.	Exit vector processing (if applicable, remove call from queue and stop backup split scanning). Pass control to call processing.
	OTHER	Fails due to unknown destination.	If final effective step in vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
	Announcement	Always succeeds.	Pass control to announcement software.

* Tenant Services partitioning checks are made.

† "Route to" steps for off-switch destinations involve some complex considerations. Refer to the Look-Ahead Interflow feature for a detailed description of these "route to" steps.

TABLE 34-B. Criteria for Success/Failure of Vector Commands (Contd)

Call Type	Command	Succeed/Fail Criteria	Vector Processing Disposition
Trunk or voice terminal calling VDN directly (Contd)	**Delay (Wait)**	Always succeeds.	Apply delay treatment.
	Go to step	Fails if step condition is not met.	Continue vector processing with next sequential step.
		Succeeds if step condition is met	Continue vector processing with the destination step.
	Go to vector *	Fails if step condition is not met.	Continue vector processing with next sequential step.
		Succeeds if step condition is met.	Continue vector processing at step 1 of the destination vector.
	Forced disconnect	Always succeeds.	Conditionally under control of announcement software. Then, exit vector processing (if applicable, remove call from queue and stop backup split scanning). Pass control to call processing.
	Forced busy	With an automatic CO trunk, fails if answer supervision has not already been returned.	Exit vector processing (if applicable, remove call from queue and stop backup split scanning). Pass control to call processing.
		Otherwise, succeeds.	Apply busy tone. Exit vector processing (if applicable, remove call from queue and stop backup split scanning). Pass control to call processing.
	Stop	Always succeeds.	Exit vector processing. Continue ACD scanning if caller is queued to split (call remains queued and backup

* The "go to vector" command is available beginning with DEFINITY Generic 2.2.

TABLE 34-B. Criteria for Success/Failure of Vector Commands (Contd)

Call Type	Command	Succeed/Fail Criteria	Vector Processing Disposition
Trunk or voice terminal to a station with a VDN in its coverage path	**Queue to main split**	Fails if an available agent is not found.	Continue vector processing with next sequential step.
		Otherwise, succeeds.	Call covers to VDN.
	Check backup split	Not applicable.	Continue vector processing with next sequential step.
	Route to	Fails if the switch is part of a DCS and the DCIU link to the destination switch is down.	Continue to ring the principal's voice terminal.
		Otherwise, succeeds.	Call covers to VDN.
	Announcement	Always succeeds.	Call covers to VDN.
	Delay (Wait)	Not applicable.	Continue vector processing with next sequential step.
	Go to step	Fails if step condition is not met.	Continue vector processing with next sequential step.
		succeeds if step condition is met.	Continue vector processing with the destination step.
	Go to vector *	Fails if step condition is not met.	Continue vector processing with next sequential step.
		Succeeds if step condition is met.	Call covers and begins vector processing at step 1 of the destination vector.

* The "go to vector" command is available beginning with DEFINITY Generic 2.2.

TABLE 34-B. Criteria for Success/Failure of Vector Commands (Contd)

Call Type	Command	Succeed/Fail Criteria	Vector Processing Disposition
Trunk or voice termial to a station with a VDN in its coverage path (Contd)	**Forced disconnect**	Succeeds for a "forced disconnect with announcement" step.	Call covers to VDN.
		Otherwise, fails.	Continue to ring the principal's voice terminal.
	Forced busy	Always fails.	Continue to ring the principal's voice terminal.
	Stop	Always fails.	Continue to ring the principal's voice terminal.
Attendant originated or attendant extended without releasing	**Queue to main split**	Fails if an available agent is not found.	Continue vector processing with next sequential step.
		Otherwise, succeeds.	Call covers to VDN.
	Check backup split	Not applicable.	Continue vector processing with next sequential step.
	Route to	Always fails.	Attendant hears busy tone.
	Announcement	Not applicable.	Continue vector processing with next sequential step.
	Delay (Wait)	Not applicable.	Continue vector processing with next sequential step.
	Go to step	Fails if step condition is not met.	Continue vector processing with next sequential step.
		Succeeds if step condition is met.	Continue vector processing with the destination step.

TABLE 34-B. Criteria for Success/Failure of Vector Commands (Contd)

Call Type	Command	Succeed/Fail Criteria	Vector Processing Disposition
Attendant originated or attendant extended without releasing (Contd)	**Go to vector ***	Always fails.	Attendant hears busy tone.
	Forced disconnect	Always fails.	Attendant hears busy tone.
	Forced busy	Always fails.	Attendant hears busy tone.
	Stop	Always fails.	Attendant hears busy tone.
Attendant extended releasing within 4 seconds	**All vector commands.**	Same as voice terminal call to a station with VDN in its coverage path.	Same as voice terminal call to a station with VDN in its coverage path.
Attendant originated or attendant extended without releasing to a station with a VDN in its Coverage Path	**Queue to main split**	Fails if an available agent is not found. Otherwise, succeeds.	Continue vector processing with next sequential step. Call covers to VDN.
	Check backup split	Not applicable.	Continue vector processing with next sequential step.
	Route to	Always fails.	Continue to ring the principal's voice terminal.
	Announcement	Not applicable.	Continue vector processing with next sequential step.
	Delay (Wait)	Not applicable.	Continue vector processing with next sequential step.
	Go to step	Fails if step condition is not met. Succeeds if step condition is met.	Continue vector processing with next sequential step. Continue vector processing with the destination step.
	Go to vector *	Always fails.	Continue to ring the principal's voice terminal.

* The "go to vector" command is available beginning with DEFINITY Generic 2.2.

TABLE 34-B. Criteria for Success/Failure of Vector Commands (Contd)

Call Type	Command	Suceed/Fail Criteria	Vector Processing Disposition
Attendant originated or attendant extended without releasing to a station with a VDN in its Coverage Path (Contd)	**Forced disconnect**	Always fails.	Continue to ring the pticipal's voice terminal.
	Forced busy	Always fails.	Continue to ring the principal's voice terminal.
	Stop	Always fails.	Continue to ring the principal's voice terminal.
Attendant extended releasing within 4 seconds to a station with a VDN in its coverage path.	All vector commands.	Same as voice terminal call to a station with a VDN in its coverage path.	Same as voice terminal call to a station with a VDN in its coverage path.

Fundamentals of Designing and Programming Vectors

Designing and programming vectors is a fairly straightforward process. The Call Vectoring "programming language" provides three basic types of "control flow" to pass vector-processing control from one vector step to another.

- Serial flow

  Serial flow passes vector-processing control from the current vector step to the following step.

- Conditional branching

  The "go to step" and "go to vector" commands provide conditional or unconditional branching, which passes control from the current step to a step other than the following step. With conditional branching, the vector command specifies a condition that must be met before the command is executed If the specified condition is not met, vector processing skips the current step and processes the next vector step (branching does not occur). If the specified condition is met, vector processing branches to the step or vector specified in the command.

  For example, the following vector command,

  go to step 8, if the time of day is between 5:00 p.m. and 7:00 a.m.,

  passes control from the current vector step to Step 8 between 5:00 p.m. and 7:00 a.m.

- Unconditional branching.

  Unconditional branching always passes control from the current vector step to the specified step or vector.

  For example, the following vector command,

  go to vector 25,

  passes control from the current vector to Step 1 of vector 25.

  Vector processing stops when the call is answered, when the System 85 or DEFINITY Generic 2 recognizes that the calling party has abandoned the call, or when the last step in the vector is processed.

  The vectors in the next section demonstrate serial flow, conditional branching, and unconditional branching.

Sample Applications of Call Vectoring

Vector A: Limiting an ACD Queue

Vector A uses a conditional "go to step" command to limit the number of calls that can be queued to an ACD split. When the limit is reached, callers hear an announcement and calls are disconnected.

1. go to step 4, if more than 7 calls are queued to split 15 at low priority
2. queue to main split 15 at low priority (Split 15 contains perhaps eight agents.)
3. stop
4. forced disconnect with announcement 17 ("Every line is busy. Please call back later.")

Step 1 tests split 15 and branches to Step 4 if more than 7 calls are in queue at low priority or higher.

When 7 or fewer calls are in split 15's queue (the test specified in Step 1 fails), control passes from Step 1 to Step 2 and calls are queued to split 15. When more than 7 calls are in split 15's queue, control passes from Step 1 to Step 4, callers hears announcement 17, and calls are disconnected.

Steps 2 and 4 must be separated by a "stop" command, otherwise, all calls would be disconnected after announcement 17, even calls that are queued to split 15.

Vector B: Providing an ACD Split to Handle Emergency Calls

Vector B uses an announcement to limit the number of calls queued to an ACD split. When more than 30 calls are in queue, callers hear an announcement that asks them to call back later, but callers are allowed to continue waiting in queue.

1. queue to main split 9, at low priority
2. go to step 4, if more than 30 calls are in (split 9's) queue at low priority
3. stop
4. announcement 20 ("We are aware of the current situation and we are trying to correct the problem. If your call is not urgent, please call back later. If it is, please wait. Your call will be answered as soon as possible.")

Step 1 queues calls to split 9 at low priority.

Step 2 tests split 9 and branches to Step 4 if more than 30 calls are in queue. Because calls are queued to split 9 in Step 1, the "go to step" command does not have to specify a split number. By default, the split that calls are currently queued to is tested.

When more than 30 calls are in split 9's queue, callers hear announcement 20 but are allowed to remain in queue.

Steps 2 and 4 must be separated by a "stop" command, otherwise, all calls would reach the announcement step, even when 30 or fewer calls are in queue.

Vector C: Providing a Specific Emergency Announcement

Vector C provides an emergency announcement for callers waiting in queue and allows callers who need help or more information to continue waiting in queue.

1. queue to main split 9 at low priority (This is the same split as in the previous vector.)
2. announcement 99 ("We are aware of the power outage in Plainfield. If you still need help, please wait.")
3. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with music
4. stop

Vector C is similar to Vector B except all callers hear the recorded announcement.

Step 4, the "stop" command, is not required. The music provided by Step 3 continues until the callers hang up whether Step 3 is followed by a "stop" command or not.

Vector D: Providing a Forced Announcement to Handle Emergency Calls

Vector D provides a recorded announcement and gives callers time to hang up before queuing calls.

1. announcement 21 ("We are aware of the current situation and we are trying to correct the problem. If your call is not urgent, please call back later.")
2. delay (wait) 2 seconds with music
3. queue to main split 9 at low priority
4. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with music

Because Step 4, the "delay (wait)" step, is the ***final effective step*** in the vector, the caller feedback (music) continues until calls are answered or until callers hang up.

Vector E: Providing a Forced Announcement for the Attendant Queue

Vector E provides a recorded announcement and gives callers time to hang up before routing calls to the local attendant queue.

1. announcement 21 ("We are aware of the current situation and we are trying to correct the problem. If your call is not urgent, call back later.")
2. delay (wait) 2 seconds with music
3. route to 3400 (LDN to access the attendant queue)

Vector F: Basing Delay Intervals on the Number of Calls in an ACD Queue

Vector F bases the amount of time callers wait before hearing a delay announcement on the number of calls in queue.

1. queue to main split 6 at low priority (Split 6 contains perhaps 16 members)
2. go to step 5, if more than 16 calls in queue at low priority
3. go to step 6, if more than 8 calls in queue at low priority
4. go to step 7
5. delay (wait) 6 seconds with ringback
6. delay (wait) 6 seconds with ringback
7. delay (wait) 8 seconds with ringback
8. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")

Step 1 queues calls to split 6 at low priority.

Step 2 tests split 6 and branches to Step 5 if more than 16 calls are in queue at low priority or higher.

If the test specified in Step 2 fails, control passes to Step 3. Step 3 tests split 6 and branches to Step 6 if more than 8 calls are in queue.

If the test specified in Step 3 fails, control passes to Step 4. Step 4 unconditionally branches to Step 7.

Vector processing directs calls to Step 5 only if more than 16 calls are in split 6's queue (the test specified in Step 2). Step 5 provides 6 seconds of caller feedback (ringback tone). When more than 16 calls are in queue, callers wait 20 seconds (6 seconds for Step 5, 6 seconds for Step 6, and 8 seconds for Step 8) before hearing announcement 16.

Calls are directed to Step 6 if more than 8 or more than 16 calls are in split 6's queue. Step 6 provides 6 seconds of ringback tone. The caller wait time for more than 16 calls in queue is described in the previous paragraph. When more than 8 but 16 or fewer calls are in queue, callers wait 14 seconds (6 seconds for Step 6 and 8 seconds for Step 8) before hearing announcement 16.

Calls are directed to Step 7 if 8 or fewer, more than 8, or more than 16 calls are in split 6's queue. Step 7 provides 8 seconds of ringback tone. The caller wait times for more than 16 and more than 8 but 16 or fewer calls in queue are described in the two previous paragraphs. When 8 or fewer calls are in queue, callers wait 8 seconds before hearing announcement 16.

Vector G: Providing Conditional Night Service for the Attendant Queue or an ACD Queue

Vector G provides alternate caller treatment based on time of day.

1. go to step 8, if T.O.D. between 5:00 p.m. and 7:00 a.m.
2.
3.
4. (Steps 2 through 7 represent vector processing during work hours.)
5.
6.
7. stop
8. route to 3300 (VDN to access a common vector for the night announcement)

Step 1 tests for T.O.D (Time of Day). During business hours, 7:00 a.m. to 5:00 p.m., Steps 2 through 7 are executed. These steps either queue calls to a local ACD split or route calls to a local attendant. After business hours, 5:00 p.m. to 7:00 a.m., vector processing branches to Step 8. Step 8 routes calls to a VDN that provides the Night-Service Announcement (see vector H).

Vector H: Providing a Night-Service Announcement

Vector H is the night-service announcement to which Vector G routes calls.

1. forced disconnect with announcement 19 ("We are closed for the evening. Please call back between the hours of 7:00 a.m. and 5:00 p.m.")

Vector I: Using AUDIX to Provide Night Service for Message Center

Vector I provides alternate caller treatment for a Message Center split based on time of day. However, this type of alternate treatment could be provided for any type of ACD split.

1. go to step 8, if T.O.D. between 4:30 p.m. and 7:30 a.m.
2. queue to main split 47 at high priority (Split 47 is a Message Center split.)
3.
4.
5. (Steps 2 through 7 represent vector processing for Message Center.)
6.
7. stop
8. queue to main split 54 at high priority (Split 54 is an AUDIX split.)
9.

10.

11. (Steps 8 through 13 represent vector processing for AUDIX.)

12.

13.

Step 1 tests for time of day. During business hours, 7:30 a.m. to 4:30 p.m., Steps 2 through 7 are executed, which queue calls to a Message center split and could provide calling-party treatment such as a repeating delay announcement (see Vector K). After business hours, 4:30 p.m. to 7:30 a.m., Steps 8 through 13 are executed, which queue calls to an AUDIX split and could provide alternate calling-party treatment in case AUDIX cannot answer calls within a reasonable amount of time.

Step 7, the "stop" command, separates vector processing for Message Center from vector processing for AUDIX.

Vector J: Providing an Information Announcement for Callers

1. forced disconnect with announcement 25 ("Today has been declared a snow day. Please report for work tomorrow at 8:00 a.m.")

   or

1. announcement 26 ("The factory is closed for the Christmas holidays. We will reopen on January 6. Please call back after this date")
2. forced disconnect

Vector K: Providing a Repeating Delay Announcement

1. queue to main split 53 at low priority
2. delay (wait) 30 seconds with ringback
3. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
4. delay (wait) 20 seconds with music
5. go to step 3

Steps 1 through 3 queue calls to split 53 at low priority and provide a delay announcement if calls wait in queue for more than 30 seconds.

Step 4 provides caller feedback (music) and Step 5 creates a loop that repeats the delay announcement and caller feedback.

Vector L: Providing Intraflow for the Older Calls in Queue

Vector L provides intraflow, diversion of ACD calls to a local destination, when calls have waited approximately 2 minutes in queue.

1. queue to main split 45 at low priority

2. delay (wait) 30 seconds with ringback
3. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
4. delay (wait) 60 seconds with music
5. check backup split 46, queue at low priority if more than 1 available agent
6. check backup split 47, queue at low priority if more than 1 available agent

Steps 1 through 3 queue calls to split 45 at low priority and provide a delay announcement if calls wait in queue for more than 30 seconds.

Step 4 provides caller feedback (music) for 60 seconds. Steps 2 through 4 give agents in the main split time to answer the call before the backup splits are checked.

Steps 5 and 6 check backup splits 46 and 47 for available agents. If more than one agent is available in split 46 or 47, the call is removed from split 53's queue and queued to one of these splits. Multiple "check backup split" commands should be sequenced in descending order of preference. When the conditions of more than one "check backup split" command are met, the call queues to the first split in the sequence that meets the conditions (in this case, Split 46). A call will only be queued to split 47 if split 46 does not have more than one available agent and split 47 does.

Vector M: Providing Conditional Intraflow for an ACD Split

Vector M intraflows calls when the oldest call in queue has waited more than 45 seconds. Calls are queued to a backup split if the backup split has fewer than 5 calls.

1. queue to main split 21 at medium priority
2. go to step 4, if oldest call in main split's queue has waited more than 45 seconds at low priority
3. stop
4. check backup split 22, queue at low priority if less than 5 calls in backup split's queue

> **NOTE:** This version of "conditional intraflow" differs from the ACD version. This vector diverts the ***newest call*** in queue, whereas the ACD version diverts the ***first call*** in queue.

Step 1 queues calls to split 21 at medium priority.

Step 2 tests split 21 and branches to Step 4 if the oldest call in queue has waited more than 45 seconds.

Steps 2 and 4 must be separated by a "stop" command, otherwise, backup split 22 would always be checked whether the test executed in Step 2 passed or failed.

When the oldest call in split 21's queue has waited more than 45 seconds, vector processing checks backup split 22 and queues calls to that split if fewer than 5 calls are in queue.

Vector N: Providing Unconditional Intraflow for an ACD Split

Vector N unconditionally intraflows calls when an ACD split is unstaffed. This version of unconditional intraflow is slightly different from the standard ACD version of unconditional intraflow. This version activates automatically after every agent enters the unstaffed mode, whereas the standard ACD version is activated by the split supervisor.

1. go to step 5, if there are no staffed agents in split 21
2. queue to main split 21 at medium priority
3. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with ringback
4. stop
5. route to 3500 (VDN of another local ACD split)

Step 1 tests split 21 and branches to Step 5 if no agents are staffed. Remember, whenever a "go to step" command that tests for number of staffed agents precedes the first "queue to main split" command, a split number must be specified or the test will fail.

If the test specified in Step 1 fails, Step 2 queues calls to split 21 at medium priority.

Step 3 provides caller feedback (ringback tone) until calls are answered.

Steps 3 and 5 must be separated by a "stop" command, otherwise, calls that waited in queue for more than 2 seconds would be routed to VDN 3500.

Vector processing directs calls to Step 5 only when split 21 is unstaffed. Step 5 routes calls to VDN 3500, which is associated with a vector that queues calls to another local ACD split.

Vector O: Providing Conditional Interflow for an ACD Split

Vector O provides interflow, diversion of ACD calls to a destination outside the switch, when the oldest call in queue has waited more than 45 seconds.

1. queue to main split 21 at medium priority
2. go to step 4, if oldest call in main split's (split 21's) queue has waited more than 45 seconds at low priority
3. stop
4. route to 8 + RNX-XXXX (Private-network number of the interflow destination)

Step 1 queues calls to split 21 at medium priority.

Step 2 tests split 21 and branches to Step 4 if the oldest call in queue has waited more than 45 seconds.

Steps 2 and 4 must be separated by a "stop" command, otherwise, calls would always interflow whether the test executed in Step 2 passed or failed.

Vector processing directs calls to Step 4 only when the oldest call in split 21's queue has waited more than 45 seconds. Step 4 routes calls to a private-network destination.

Vector P: Providing Unconditional Interflow for an ACD Split

Vector P interflows calls when an ACD split is unstaffed.

1. go to step 5, if there are no staffed agents in split 21
2. queue to main split 21 at medium priority
3. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with ringback
4. stop
5. route to 9 + NPA-NXX-XXXX (Public-network number of the interflow destination)

Step 1 tests split 21 and branches to Step 5 if no agents are staffed. Remember, whenever a "go to step" command that tests for number of staffed agents precedes the first "queue to main split" command, a split number must be specified or the test will fail.

If split 21 is staffed (the test specified in Step 1 fails), Step 2 queues calls to split 21 at medium priority.

Step 3 provides caller feedback (ringback tone) until calls are answered.

Steps 3 and 5 must be separated by a "stop" command, otherwise, calls that waited in queue for more than 2 seconds would be routed to the public-network number specified in step 5.

Vector processing directs calls to Step 5 only when split 21 is unstaffed. Step 5 routes calls to a public-network destination.

Providing Look-Ahead Interflow for an ACD Split

Beginning with R2 V4, Issue 1.3 and DEFINITY Generic 2, the Look-Ahead Interflow feature can be used in conjunction with ACDs that ***also*** use the Call Vectoring and the ISDN (Integrated Services Digital Network)/PRI (Primary Rate Interface) features.

With this arrangement, vector programming (at the switch that initially received an ACD call) first decides whether the interflow operation is necessary. After the need to interflow is determined, the interflow destinations are specified by one or more "route to" steps within the same vector. However, before an ACD call is redirected to the VDN of another switch, the "sending switch" queries the "receiving switch" over the D channel of the appropriate ISDN—PRI tie-trunk group. This process allows the receiving switch to decide whether it can adequately handle the call. The receiving switch makes this decision according to vector programming within its own vector (assigned to the VDN specified in the "route to" step at the sending switch).

If the receiving switch can handle the call, it accepts the call with a D-channel message, and the sending switch sends the call. If the receiving switch cannot handle the call, it refuses the call with a different message. At this time, the sending switch either queries another switch (according to subsequent "route to" steps) or executes the alternative action programmed within its own local vector.

> **NOTE:** Refer to the Look-Ahead Interflow section of this manual for a detailed description of this operation.

Vector Q: Scanning Multiple Backup Splits

Vector Q uses "check backup split" commands to check the availability of agents in three backup splits when the main split has fewer than 16 staffed agents. Calls interflow if more than one agent is available in one of the backup splits.

1. queue to main split 37 at low priority (Split 37 contains perhaps 30 agents)
2. go to step 4, if the number of staffed agents is less than 16
3. stop
4. check backup split 11, queue at low priority if more than 1 available agent
5. check backup split 12, queue at low priority if more than 1 available agent
6. check backup split 13, queue at low priority if more than 1 available agent

Step 1 queues calls to split 37 at low priority.

Step 2 tests split 37 and branches to Step 4 if there are fewer than 16 agents staffed.

Step 3, the "stop" command, separates Step 2 from Steps 4,5, and 6. Without Step 3, the backup splits would be checked no matter how many agents were staffed in split 37.

Vector processing directs calls to Steps 4, 5, and 6 only if fewer than 16 agent are staffed in split 37. Steps 4, 5, and 6 check backup splits 11, 12, and 13 for available agents. If more than one agent is available in split 11, 12, or 13, the call is removed from split 37's queue and is queued to one of these splits. Multiple "check backup split" commands should be sequenced in descending order of preference. Usually, when the conditions of more than one "check backup split" command are met, the call queues to the first split in the sequence that meets the conditions. A call will only be queued to split 13 if splits 11 and 12 do not have more than one available agent and split 13 does.

Vector R: Combining the Conditions of "Check Backup Split" Commands

Vector R uses two "check backup split" commands that specify different conditions to check the same backup split. Calls intraflow if either or both of the conditions are met.

1. queue to main split 56 at medium priority (Split 56 contains perhaps 10 agents)
2. check backup split 57, queue at high priority if backup split has more than 2 staffed agents
3. check backup split 57, queue at high priority if the oldest call in backup split's queue has waited less than 34 seconds
4. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with ringback

Step 1 queues calls to split 56 at medium priority.

Step 2 checks backup split 57 for more than 2 staffed agents. Step 3 checks split 57's queue for an oldest call wait time of less than 34 seconds. The "check backup split" command in Step 3 tests split 57's queue for calls queued at high or top priority. Calls queued at lower priorities are not considered. The result of these "check backup split" commands is that a call will be removed from split 56's queue and queued to split 57 if either or both of the specified conditions are met.

Step 4 provides caller feedback (ringback tone) until calls are answered.

Vector S: Gracefully Closing an ACD Split

Vector S uses several "go to step" and "announcement" commands to provide alternate caller treatment based on time of day and day of week and to warn callers waiting in queue that the ACD split will close soon.

1. go to step 14, if T.O.D. between 4:00 p.m. and 8:00 a.m.
2. go to step 14, if D.O.W. is Sunday
3. queue to main split 52 at low priority
4. go to step 9, if T.O.D. between 3:56 p.m. and 4:01 p.m.
5. delay (wait) 20 seconds with ringback
6. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
7. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with music
8. stop
9. announcement 22 ("It is nearly 4:00, closing time for this office. We are trying to serve your call. If we can't, please call back between 8:00 and 4:00, Monday through Saturday.")
10. delay (wait) 6 seconds with music
11. go to step 13, if T.O.D. between 4:00 p.m. and 4:15 p.m.
12. go to step 10

13. forced disconnect with announcement 23 ("We are sorry. This office has closed. To be assured of service, please call back between 8:00 and 3:45 Monday through Saturday.")
14. forced disconnect with announcement 24 ("Please call back during business hours: 8:00 to 4:00, Monday through Saturday.")

Step 1 tests for T.O.D (Time of Day) and branches to Step 14 between the hours of 4:00 p.m. and 8:00 a.m., when the office is closed.

Step 2 tests for D.O.W. (Day of Week) and branches to Step 14 if it is Sunday, when the office is closed.

Vector processing directs calls to Step 3 only if the T.O.D and D.O.W tests specified in Steps 1 and 2 fail (Monday through Saturday from 8:00 a.m. to 4:00 p.m.). Step 3 queues calls to split 52 at low priority.

Step 4 tests for T.O.D and branches to Step 9 between 3:56 p.m. and 4:01 p.m. Monday through Saturday. (The upper time boundary is set to 4:01 p.m. to "trap" the occasional calls that cross the previous 4:00 p.m. boundary.)

Step 5, 6, and 7 provide ringback tone for 20 seconds, a delay recorded announcement, and music until calls are answered.

Step 8, the "stop" command separates Steps 1 through 7 (vector processing for normal business hours) from Steps 9 through 14 (the "Almost closing time" and "Please call back" announcements).

Only calls that arrive between 3:56 p.m. and 4:01 p.m. Monday through Saturday reach Step 9. Step 9 provides a recorded announcement that warns callers that the office is going to close soon and that their call may not be answered.

For calls that reach Step 9, Step 10 provides caller feedback (music) until the calls are answered.

Step 11 tests for T.O.D and branches to Step 13 between 4:00 p.m. and 4:15 p.m. Monday through Saturday. This step directs calls that are still in queue at 4:00 p.m. to a "Please call back during office hours" announcement.

Step 12 unconditionally branches to Step 10 creating a loop that provides music and checks to see if announcement 23 (Step 13) should be played. This loop also separates Steps 9 through 12 from Steps 13 and 14.

Step 13 provides a "Please call back during office hours" announcement for calls that were waiting in queue and could not be answered before the office closed.

Only out-of-business-hours calls (between 4:00 p.m. and 8:00 a.m Monday through Saturday and between 12:00 a.m. Sunday and 8:00 a.m. Monday) reach Step 14. These calls are not queued to split 52. Callers hear announcement 24 and are disconnected.

Vector T: Example of a Chained Vector for ACD

Vector T contains vector commands similar to those presented in previous sample vectors. However, because it contains more than 15 commands, a "route to VDN" command (Step 15) is used to link two vectors Steps 2 and 3 direct out-of-business-hours calls to Step 15 which routes them to the continuation vector.

1. queue to main split 59 at low priority (Split 59 contains perhaps 25 agents.)
2. go to step 15, if D.O.W. is Saturday or Sunday (Branch for weekends)
3. go to step 15, if T.O.D. between 4:30 p.m. and 7:30 a.m. (Branch for evenings)
4. go to step 12, if more than 35 calls in main split's queue at low priority (Branch when severely overloaded)
5. go to step 10, if oldest call in main split's queue has waited more than 36 seconds (Intraflow - mild overload)
6. delay (wait) 30 seconds with ringback (Normal processing for Queue 59)
7. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
8. delay (wait) 20 seconds with music
9. go to step 7
10. check backup split 1, queue at low priority if less than 10 calls in backup Split's queue
11. go to step 6
12. announcement 17 ("Every line is busy. Please call back later.")
13. forced busy
14. stop
15. route to VDN 3307 ("3307" is a VDN to access the continuation vector.)

VDN 3307 — Continuation Vector

1. announcement 18 ("Please call back during business hours: 7:30 to 4:30, Monday through Friday.")
2. forced disconnect

Vector U: Call Wait Announcements with Voice Mail for Callback

Vector U uses recorded announcements to tell callers approximately how long they will have to wait before their call is answered. When the wait time is about three minutes, calls are routed to an AUDIX split so that callers can leave a callback message.

Vector U uses "go to vector" commands to link or "chain" several vectors together. The "go to vector" command is available beginning with DEFINITY Generic 2.2.

Vector 114 (queues calls to Service Hot Line)

1. queue to main split 31 at low priority
2. delay (wait) 2 seconds with ringback
3. go to vector 115 if oldest call in main split's queue (split 31) has waited more than 30 seconds at low priority
4. delay (wait) 20 seconds with ringback
5. go to vector 120

Vector 115 (informs caller of approximate wait time while call remains in queue)

1. go to step 6 if oldest call in split 31's queue has waited less than 60 seconds at low priority
2. go to step 8 if oldest call in split 31's queue has waited less than 120 seconds at low priority
3. go to step 10 if oldest call in split 31's queue has waited less than 180 seconds at low priority
4. announcement 2 ("All agents are busy. It may take more than 3 minutes to answer your call. Please leave a message and someone will call you back.")
5. route to VDN 2500
6. announcement 3 ("All agents are busy. Your call will be answered in about 1 minute.")
7. go to step 11
8. announcement 4 ("All agents are busy. Your call will be answered in about 2 minutes.")
9. go to step 11
10. announcement 5 ("All agents are busy. Your call will be answered in about 3 minutes.")
11. go to vector 120

VDN 2500 (Voice Mail Box for Service Hot Line)

Vector 12 (Allows caller to leave a message. Call is removed from split 31's queue.)

1. queue to main split 10 (AUDIX split) at low priority
2. delay (wait) 180 seconds with ringback
3. go to vector 105

Vector 120 (Continue waiting in split 31's queue with repeating delay announcement)

1. announcement 8 ("All agents are busy, please wait.")

2. delay (wait) 20 seconds with music
3. go to step 1

Vector 105 (Returns call to split 31's queue)

1. queue to main split 31 at medium priority
2. delay (wait) 20 seconds with music
3. announcement 8 ("All agents are busy, please wait.")
4. go to step 2

Vector 114 queues calls to split 31 at low priority (Step 1), provides ringback tone (Step 2), and then tests split 31 and branches to Vector 115 if the oldest call in queue has waited more than 30 seconds at low priority or higher (Step 3). If the oldest call in split 31's queue has waited 30 seconds or less, Step 5 directs calls to Vector 120, which provides a repeating delay announcement and caller feedback (music) until calls are answered.

Vector processing directs calls to Vector 115 (from Vector 114) if the oldest call in split 31's queue has waited more than 30 seconds. Steps 1,2, and 3 of Vector 115 test split 31 and branch to the appropriate announcement based on the amount of time the oldest call in queue has waited. Calls only reach Step 4 of Vector 115 when the oldest call in split 31's queue has waited between 2 and 3 minutes. After callers hear announcement 2 (Step 4), calls are routed to an AUDIX split (Step 5) so that callers can leave a message. Each "call wait" announcement (Steps 6, 8, and 10) is followed by a "go to step" or "go to vector" command that directs calls to Vector 120, which provides music and a repeating delay announcement.

Vector 12, which is associated with VDN 2500, queues calls to an AUDIX split at low priority. Calls are directed to this vector from Vector 115 if the oldest call in split 31's queue has waited between 2 and 3 minutes. Step 2 of Vector 12 provides 3 minutes of caller feedback (ringback tone). If AUDIX does not answer a call within 3 minutes, Step 3 directs the call to Vector 105, which queues the call to split 31 again, but this time at medium priority.

Vector processing directs calls to Vector 120 after hearing a "call wait" announcement in Vector 115. Vector 120 provides a delay announcement followed by music. This cycle repeats until calls are answered.

Calls are directed to Vector 105 from Vector 12 if AUDIX does not answer a call within 3 minutes. Vector 105 queues calls to split 31 again, but this time at medium priority. Steps 2 through 4 of Vector 105 provide a repeating delay announcement and caller feedback (music) until calls are answered.

User Operations

To Extend an Attendant Call to a Vector Directory Number

An attendant should:

1. Press the **[ANSWER]** button. [Establishes talking connection between the attendant and the calling party.]
2. Press the **[START]** button. [The switch returns dial tone and places the calling party on soft hold.]
3. Dial the vector directory number. [The switch returns ringback tone to the attendant.]
4. Press the **[RELEASE]** button ***within 4 seconds*** . [Vector processing begins for the calling party.]

   NOTE: If the attendant does not release within 4 seconds, the switch will attempt to complete the call to an idle answering destination. If an available answering destination is not found, the switch will return busy tone to the attendant.

A CAS attendant should:

1. Press the appropriate loop button.

   or

   Press **[ANSWER]** . [PA lamp goes out] (Calling party is connected.)
2. After receiving the calling party's instructions, press **[START]** . [Dial tone] (Calling party is placed on hold.)
3. Dial the vector directory number. [Ringback tone]
4. Press the **[RLT RELEASE]** button ***within 4 seconds.*** [Vector processing begins for the calling party.]

   NOTE: If the centralized attendant does not release within 4 seconds, the switch will complete the call to an idle answering destination. If an available answering destination is not found, the switch will return reorder tone to the attendant.

To Transfer (Extend) an Incoming RLT Call to a Branch Location:

*An ACD agent (or other voice terminal user) should:**

1. After receiving the calling party's instructions, press **[TRANSFER]** or **[CONFERENCE]** . [Dial tone] (Calling party is placed on hold.)
2. Dial the requested number.
3. If ringback (or Call Waiting ringback) is heard, press **[DISCONNECT]** , **[RELEASE]** , or Go on-hook. (The RLT is released.)

   NOTE: If the called party does not answer before the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main.

   or

   If busy tone is heard, press **[TRANSFER]** or **[CONFERENCE]** (voice terminal user is reconnected to the calling party) and inform the calling party that the extension is busy.

   If the calling party does not want to wait for the called party to answer, press **[DISCONNECT]** , **[RELEASE] ,** or Go on-hook. (The RLT is released.)

   or

   If the calling party wants to wait for the called party to answer, press **[TRANSFER]** or **[CONFERENCE]** . [Dial tone] (Calling party is placed on hold at the main.)

   Dial the Remote Hold dial access code. [Confirmation tone] (Calling party is placed on hold at the branch.)

   Press **[DISCONNECT]** , **[RELEASE]** , or Go on-hook. [Dial tone] (The RLT is released.)

   NOTE: If the calling party is a station at a branch location, Remote Hold will be denied.

   NOTE: When the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main. The terminal user can attempt to complete the call again.

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* For RLTs that terminate to ACD splits or VDNs, any voice terminal with a Conference or Transfer button can be used to transfer incoming RLT calls from the CAS main to a branch location.

* For incoming RLT calls, the Conference button works the same as the Transfer button. That is, pressing the Conference button transfers an incoming RLT call back to the originally called branch location. Pressing the Conference button does not set up a 3-party conference.

To Program a List Item in the Vector-Group List

The controller of the vector-group list should:

1. Press the **[ABRVDIAL PROGRAM]** button,

   or

   Dial the Program access code (Encode 93). [Confirmation tone]

2. Press the **[GROUP]** list-selection button,

   or

   Dial the Group-List access code. [Dial tone]

3. Dial an item number (from 1 to 95) in the group list. [Second dial tone]

4. Dial the number to be used as a destination for a "route to" step. (Do not program ***special function*** characters)

5. Press the **[GROUP]** list-selection button again to enter the new number,

   or

   Dial **[#]** if no list-selection button is provided. [Confirmation tone]

To Verify a Recorded Announcement

An ACD split supervisor should:

1. Go off-hook on an idle appearance. [Dial tone]

2. Dial the Announcement Verify access code,

   or

   Press the **[VERIFY ANNCT]** button (an Abbreviated Dialing button with the Announcement Verify access code as the stored number). [Second dial tone]

3. Dial the 2-digit announcement number.

4. Listen to the announcement.

5. Go on-hook,

   or

   Press the **[RELEASE]** button.

Considerations

Switch Capacity

System 85 or DEFINITY Generic 2 software allows for as many as 128 (511 beginning with DEFINITY Generic 2.2) vectors to be defined within the switch. As many of the 128 (511 beginning with DEFINITY Generic 2.2) vectors can terminate to the same answering destination.

Each vector can be assigned to as many VDNs as desired, and each VDN can be assigned to as many of the 982 administrable trunk groups as desired.

Recorded Announcement Limit

For System 85 and DEFINITY Generic 2.1, as many as 84 SN231 recorded announcement trunks (numbers 16 through 99) can be assigned for use with Call Vectoring. The Intercept Treatment feature also uses recorded announcement trunks. The first 15 recorded announcements are reserved for the Intercept Treatment feature.

Beginning with DEFINITY Generic 2.2, as many as 255 recorded announcement trunks can be assigned for use with the Call Vectoring and Intercept Treatment features. As many as 15 announcement trunks can be assigned to the Intercept Treatment feature and these announcement trunks can be assinged to any trunk ciruits. That is, unlike previous versions of DEFINITY Generic 2 and System 85 software, announcement trunk circuits are not reserved for Intercept Treatment.

Based on time-slot and TMS-blockage limitations, as many as 255 callers per module can listen to the same recorded announcement at the same time. In practice, the limit is considerably lower.

The Call Vectoring software adds calling parties who are waiting for an announcement to an unlimited "waiting queue." Between announcement cycles, the calls at the head of the waiting queue are moved to a "listening queue." This software can move up to 144 callers to the listening queue before the beginning of the next announcement, but will continue to add parties after the announcement begins. In the worst case, the 255th caller would not hear the first 1.4 seconds of this announcement.

If more than 144 calling parties are routinely being added to the same listening queue, consider reengineering the ACD configuration.

Continuous Announcements versus Delay Announcements

As previously mentioned, as many as 84 recorded announcements (255 beginning with DEFINITY Generic 2.2) can be provided for Call Vectoring. Each of these announcements has its own trunk group assigned as Trunk Type 90 (Vectoring Recorded Announcement) in Procedure 100, Word 1.

Moreover, these announcements are shared between "delay" announcements and "continuous" announcements. Delay announcements provide information for calling parties and are programmed as part of vectors. Continuous announcements provide information (that is, VDN-, city-, or queue-of-origin) for an answering ACD agent and ***cannot*** be programmed as a step in a vector.

As an alternative to continuous announcements, VDNs and incoming trunk groups can be assigned names (Procedure 012, Word 1). For ACD agents equipped with display terminals, the displayed name substitutes for the continuous announcement.

If continuous announcements are provided, the switch administrator should implement these announcements in the same way that city- or queue-of-origin announcements for ACD are implemented. (See the heading "ACD From the Agent's Perspective" in the ACD section of this manual.)

When Call Vectoring is enabled, the switch administrator must distinguish between continuous and delay announcements in Field 11 of Procedure 150. (To define a continuous announcement, enter a "1" in Field 11. Enter "0" for a delay announcement.)

After an announcement is defined as a continuous announcement, this announcement ***cannot*** be programmed as a vector step in Procedure 030, Word 3. When this is attempted, the Error 95 will occur.

As with the previous versions of DEFINITY Generic 2 and System 85, for DEFINITY Generic 292, Call Vectoring recorded announcements are shared between "delay" announcements and "continuous" announcements.

"Scratch-Pad" Vector Administration

Vectors can be administered while the switch is actively processing calls. However, once a vector assumes control of processing for a specific call, that vector cannot change until processing for the call is finished.

When a vector is modified, System 85 or DEFINITY Generic 2 enters a transitional phase of vector processing. During the transition, the switch accepts the new vector as a "temporary vector," and immediately applies this vector to every new call that accesses the vector. Meanwhile, the old "permanent vector" still controls processing for every call that accessed the vector before the modification. After the switch finishes processing these "old calls," the old vector is replaced with the new vector as the new "permanent vector."

Answer Supervision and Incoming ISDN—PRI Trunks

When Call Vectoring controls call processing for an incoming ISDN call, the R2 V4 System 85 or DEFINITY Generic 2 returns answer supervision (a CONNECT message) just before the call is answered, a "delay (wait) with music" step, an "announcement: step, or a "forced disconnect with announcement*" step.

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* The System 85 or DEFINITY Generic 2 does not return answer supervision for "forced disconnect without announcement" steps during ISDN—PRI calls. Using ISDN—PRI facilities, a call need not be answered to be disconnected.

As the R2 V4 System 85 or the DEFINITY Generic 2 receives an incoming ISDN—PRI call destined for vector processing, the R2 V4 System 85 or DEFINITY Generic 2 returns a CALL PROCEEDING message to the serving switch. When the billing switch receives this message, it infers that the ISDN call has successfully negotiated for a B channel in an ISDN—PRI trunk group, but that answer supervision is not yet being returned. Then, just before an agent answers or an appropriate vector step executes, the R2 V4 System 85 or DEFINITY Generic 2 returns an "Accept" (for example, a CONNECT, ALERTING, or PROGRESS) message to the serving switch. When the billing (sending) switch receives a CONNECT message, billing begins for the ISDN—PRI call.

Special Function Characters and "Route To" Commands

When an Abbreviated Dialing list item for Call Vectoring is programmed, the special function characters (for example, pause, wait, and mark) ***must not*** be used. The networking features (AAR, ARS, and WCR) automatically handle the timing to complete these calls. So, only the digits for a "route to" destination should be programmed. If special characters ***are*** programmed into a Call Vectoring group-list item, the "route to" command that uses the list item will fail.

Adjunct Machine Numbers

At most, each vector can point to ***one*** Message Center AP ***and one*** AUDIX adjunct. Therefore, vectors with multiple "queue to main split" or "check backup split" commands should be carefully programmed.

As an example, it is legal to queue to a Message Center split on one AP. It would also be legal to check several backup Message Center splits that correspond to the ***same*** AP. However, Procedure 030, Word 3 would return an administration error if one or more of the vector's "check backup split" commands pointed to a split on a different AP.

As another example, it is legal to queue to the AUDIX split corresponding to one AUDIX adjunct. It is also legal to base a conditional branch on the status of that queue. However, Procedure 030, Word 3 would return an administration error if the vector tried to overflow calls to a different AUDIX adjunct's split.

However, as previously shown in Vector I, it is legal to provide automatic night service for a Message Center split by directing calls to an AUDIX split after hours. Vector I reaches the limit by pointing to one AP and one AUDIX adjunct.

Testing Vectors

Customers are encouraged to use creativity in designing vectors that fully address an individual switch's needs. However, vector processing is a powerful software tool that, without exercising due caution, could produce highly undesirable and unexpected results in response to an ill-conceived or misprogrammed vector. It is strongly recommended that every vector (especially vectors with untried sequences of steps) be fully tested before they are placed in control of actual call processing or written to permanent translation on the HCMR or DTS tape.

To facilitate vector testing, one VDN, one vector, and a few answering destinations of each type can be reserved for testing purposes. The logic within this test vector, for example Vector 128, should be identical to the logic in the planned vector. However, the conditional parameters (for example, greater than 30 calls in queue) can be relaxed to simplify the testing process. In this way, the logic of Vector 128 can be tested to ensure effective and sane results. After a successful test phase, the conditional parameters in Vector 128 are set to the desired values, and then Vector 128 is copied to another vector number. (At this point, Vector 128 can be removed.) In turn, the new duplicate vector is assigned to the VDN planned for use in actual call processing.

Final "Delay (Wait)" Steps

The last "delay (wait)" step in a vector can be timed by vector processing for any even value between 2 seconds and 998 seconds (between 0 and 998 beginning with DEFINITY Generic 2.2). For ***any*** value chosen, the caller hears the programmed audible feedback (ringback or music) beyond the delay (wait) timing until the call is either answered or abandoned.

Therefore, setting the delay (wait) interval to 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) can be a good idea. In this way, the vector processing finishes as quickly as possible. This strategy has the benefit of conserving processor occupancy for the switch.

However, setting this delay (wait) intend to 2 seconds can also change the meaning of the CMS statistic "Total Time in Vector." As the final 2-second timing elapses for each call, the switch sends a message to the CMS indicating that vector processing has finished. So, vector processing can ***finish*** before an answering destination actually ***answers*** a call. In the CMS environment, it might be more desirable to ensure that the "Total Time in Vector" accurately reflects the amount of time for answering each VDN call. If so, set the timing for final "delay (wait)" steps to a much higher value.

Permanent Seizure Counters and Timers

To help the switch recover from permanent trunk seizures, the Call Vectoring software contains three counters. For VDN calls ***from over incoming trunks,*** one or more of these counters is invoked to limit the duration of permanent seizure conditions. These counters include:

- "Route To" Retry Counter

  While a final effective "route to" step is being retried, this counter's task is invoked at 5-minute intervals. After the counter has incremented 6 times (approximately 30 minutes), the switch assumes a permanent seizure condition and tears down the connection.

- "Check Backup Split" Command Counter

  During vector processing, this counter keeps track of the number of times that a "check backup split" command has been executed. After the counter has incremented 1000 times (2000 seconds for 1 "check backup split" command, 1000 seconds for 2 "check backup split" commands), the switch assumes a permanent seizure condition and tears down the connection.

- "Go To Step" Command Counter

  During vector processing, this counter keeps track of the number of times that a "go to step" command has been executed. After the counter has incremented 1000 times, the switch assumes a permanent seizure condition and tears down the connection. (This is the counter that prevents "endless loops.")

The Call Vectoring software also contains two 20-second timers to minimize permanent trunk seizures. These timers include:

- "Forced Busy" Command Timer

  Each time a vector executes a "forced busy" command on an incoming trunk call, this timer is invoked. After 20 seconds elapse, the switch checks the trunk connection. If the connection is still active, the switch assumes a permanent seizure condition and tears down the connection.

- "Forced Disconnect" Command Timer

  Each time a vector executes a "forced disconnect" command on an incoming trunk call, this timer is invoked. After 20 seconds elapse, the switch checks the trunk connection. If the connection is still active, the switch assumes a permanent seizure condition and tears down the connection.

- "Stop" Command Timer

  Each time a vector executes a "stop" command or a final effective vector step (except a final effective "route to" step) on an incoming trunk call where the call is ***not*** queued to a split, this timer is invoked After 20 seconds elapse, the switch checks the connection. If the connection is still active, the switch assumes a permanent seizure condition and tears down the connection.

Logical Patterns to Avoid

A "go to step" command should never direct vector processing to itself. As an example, consider this vector sequence that could have occurred by ***adding*** Steps 2 and 3 to an old vector.

1. queue to main split 46 at priority 0
2. check backup split 47, queue if less than 7 calls in split 47's queue
3. check backup split 48, queue if less than 7 calls in split 48's queue
4. go to step 4, if time of day is between 5:00 p.m. and 8:00 a.m.
5. stop
6. route to (a night service destination)

In addition to the logical problem caused by Step 4, this command causes a ***severe*** problem for vector processing*. Step 4 does nothing but consume switch processor time. This command would consume 6 milliseconds of processor time per calling party per second.

This same ill effect can also be consciously programmed by "daisy chaining" "go to step" commands†.

The following sequence is an example.

1. go to step 2
2. go to step 3
3. go to step 1

Since "check backup split" commands are heavy consumers of switch processor time, excessive use of these commands should be avoided. Each "check backup split" command consumes 1/2 millisecond of processor time per calling party per 2-second interval. So, ***one*** of these commands making checks for ***400*** calling parties would increase processor occupancy by approximately 10 percent. Or, two commands making checks for 200 calling parties would also increase processor occupancy by about 10 percent.

Limited Integrity Checks

Before accepting a vector to permanent memory, administration software performs three limited integrity checks on each programmed vector. However, these integrity checks do not substitute for thorough vector testing. The checks include:

- Ensuring that the destination step of every "go to step" command exists.
- Ensuring that a "go to step" command or a "go to vector" command does not direct vector processing to itself.
- Ensuring that every time-of-day branch has both a starting time and an ending time
- Ensuring that if the ***start time*** of a time-of-day branch is programmed as "every day," then the ***end time*** is also programmed as "every day" (and vice versa).

Upgrading to a System With Release 2 CMS and Call Vectoring

When upgrading from a system without Call Vectoring to a system with Release 2 CMS and Call Vectoring, CMS vectoring software must be activated (PEC 1208-012).

* An integrity check ensures that a "go to step" command does not direct vector processing to itself.
† There is no integrity check to prevent daisy chaining.

Hard and Soft Processor Swaps

Stable VDN calls will endure a hard processor swap.

Permanent vectors are stored in a translation portion of switch memory. Therefore, these vectors will endure a hard processor swap.

Transition vectors are stored in a translation portion of switch memory. Therefore, these vectors will endure a hard processor swap.

The unwritten scratch-pad vector is stored in a status portion of switch memory. Therefore, these vector statements are lost during a hard processor swap.

The contents of the Abbreviated Dialing group list for Call Vectoring are stored in a translation portion of switch memory. Therefore, these "route to" destinations will endure a hard processor swap.

Vector processing is disrupted during a hard processor swap. For example, VDN calls are removed from queue, check-backup-split scanning ceases, and the switch buffer for CMS messages is cleared.

The Call Vectoring feature operates normally during a soft processor swap.

Interactions With Other Features

Abbreviated Dialing

In addition to the System List which is authorized separately, the Abbreviated Dialing feature permits voice terminal users to access a maximum of two Abbreviated Dialing lists: two group lists, two personal lists, or one group list and one personal list. A vector-group list counts as one of these two lists. Therefore, if a voice terminal user is designated as the controller of a vector-group list, the controller can only have access to one other personal or group list.

If a vector-group list controller shares Abbreviated Dialing with a data terminal, both the voice and data terminal must use the same personal or group list for Abbreviated Dialing. Normally this will not pose a problem as that list can contain up to 95 entries. To help keep voice terminal and data terminal list entries separate, it is recommended that this list be allowed the full 95 members and that one terminal use only the low-order entries (01 to 49) and the other terminal use only the high-order entries (50 to 95).

Call Vectoring group-list items ***must not*** contain special function characters (for example, pause, wait, and mark). The networking features (AAR, ARS, and WCR) automatically control the timing for routing calls to the destinations of "route to" commands.

ASAI Gateway Interface

The ASAI Gateway Interface feature can use Call Vectoring to process an incoming call and to deliver the same call to an answering destination. Because a call may encounter more than one VDN, the commands that make up the vectors associated with the VDNs should be coordinated so that they provide the appropriate calling-party treatment.

The vector that processes incoming calls can be programmed with alternate answering destinations in case an ASAI Gateway or an ASAI Gateway Interface trunk group fails. Alternate answering destinations may include local extensions or ACD splits, remote extensions or ACD splits, other VDNs, and other ASAI Gateway Interface trunk groups to the same or a different ASAI Gateway.

Attendant Control of Trunk Group Access

If vector processing encounters a "route to" step where the call would route over a trunk group that is currently controlled by the attendant, the "route to" step is not executed. Instead, if the "route to" step is the final effective step in the vector, the switch treats the "route to" step as a "stop" step. If vector steps follow the "route to" step, vector processing continues with the next sequential step.

Attendant Direct Extension Selection With Busy Lamp Field

In a switch with 3- or 4-digit extensions, an attendant can use the appropriate DXS (Direct Extension Selection) buttons to place or extend calls to a VDN. However, since a VDNs associated vector is never really "busy," the BLF (Busy Lamp Field) lamps for these DXS buttons are never lit.

Attendant Display

When Call Vectoring diverts a coverage call to the attendant queue (using a "route to" step in the covering vector), the attendant who answers the redirected call receives the usual display information on the alphanumeric display (that is, the same display information that would have been provided for a direct call to the attendant). The called principal is not identified.

For direct calls to a VDN, if the associated vector has a "route to" step that routes calls to a local attendant, the attendant console displays a 4-character message (assigned in Procedure 031, Word 1).

Attendant Release Loop Operation

The Attendant Release Loop Operation feature does not apply to calls that an attendant extends to a VDN. Once an attendant-extended call enters vector processing, this call is not timed and no reminder will be given to the attendant. For more information, refer to Table 34-B.

AUDIX (Audio Information Exchange)

A vector can have a "queue to main split" step or a "route to" (AUDIX VDN) step that routes calls to an AUDIX split. However, if the DCIU link is down or if every voice port to that AUDIX is unstaffed, these calls are not directed to the AUDIX queue. Instead, the switch returns reorder tone to the calling party.

If the AUDIX system has answered a call and Call Transfer Out of AUDIX is activated to transfer the AUDIX call to a VDN, the Call Vectoring feature screens these transfers to limit undesirable vector treatment of the transferred call. A Call Vectoring subroutine quickly scans the VDN's vector on a per-call basis to be sure that one of a set of vector steps will operate on the call before the transfer is allowed. These steps include:

- Queue to main split (with staffed agents) step
- Route to step
- Forced disconnect ***with*** recorded announcement step (beginning with R2 V4, Issue 1.2).

AAR (Automatic Alternate Routing)

For System 85 and DEFINITY Generic 2.1 switches, the FRL administered to a VDN's class of service in Procedure 010, Word 3 is used to determine whether a call is allowed to route over available public or private network facilities.

When the AAR feature is used to route calls over private network facilities, the contents of a vector-group list item for a "route to" step must conform to one of the following forms:

- RNX (3-Digit Location Code) + XXXX (4-Digit Extension Number)
- RN (2-Digit Location Code) + XXXX (4-Digit Extension Number)
- RNX (3-Digit Location Code) + XXX (3-Digit Extension Number)
- RN (2-Digit Location Code) + XXX (3-Digit Extension Number).

ACD (Automatic Call Distribution)

When Call Vectoring is enabled, the abandon call search function of ACD operates normally. During any step of the call processing for incoming calls provided by a vector, the check for abandoned calls is performed just before ringing an idle agent. The switch only distributes the call to an agent when the trunk is found active at the CO (Central Office).

When Call Vectoring is enabled, the queue-status display function of ACD operates normally. These agents receive the full queue-status information as each ACD call is received. These agents can also manually update the display information using the NORMAL MODE button.

When Call Vectoring is enabled, the multiple call handling function of ACD operates normally. When an agent receives a distributed call that was processed by a vector, the agent can place the call on hold and remain available to receive another call that is distributed from the split's queue.

When Call Vectoring is enabled, ACD queue warning lamps can still be provided. (The warning lamp threshold for each split is still assigned in Procedure 026, Word 1, Field 5.) However, the meaning of lamp activity on the 30A8 is slightly modified. The appropriate lamp still lights when the number of calls in the split's queue is greater than or equal to the assigned threshold. With Call Vectoring, however, the assigned queue warning threshold ***can be a different value*** than the parameter(s) within the vector itself that actually divert calls to alternate destinations.

Automatic Callback

The switch denies activation of Automatic Callback toward a VDN. When this is attempted, the switch returns intercept tone to the activating party.

Whenever a "forced busy" step in a vector returns busy tone to a calling party, the switch denies activation of Automatic Callback in response to the busy tone. When this activation is attempted, the switch returns intercept tone to the calling party.

ARS (Automatic Route Selection)

For System 85 and DEFINITY Generic 2.1 switches, the FRL administered to a VDN's class of service in Procedure 010, Word 3 is used to determine whether a call is allowed to route over available public or private network facilities.

When the ARS feature is used to route calls over public network facilities, the contents of a vector-group list item for a "route to" step must conform to the public network rules for DDD (Direct Distance Dialing). The DDD formats for the public network can have one of the following forms:

- NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- NPA (3-Digit Area Code) + NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- International Telephone Number.

ARS Toll Restriction does not prevent "route to" steps from routing calls to destinations outside the switch. If ARS Toll restriction is assigned to a VDN's class of service, the assignment is ignored.

Bearer Capability

The Bearer Capability feature has no direct impact on the Call Vectoring feature. A vector does not have a BCCOS. Therefore, Bearer Capability is not checked when routing calls to a Vector Directory Number. However, if a "route to" step within a vector involves the insertion of a Modem Pooling conversion resource, then the Modem Pooling feature requires that Bearer Capability be checked.

Busy Verification of Lines

An attendant is not allowed to activate busy verification toward a VDN. When this is attempted, the switch returns intercept tone to the attendant.

Call Coverage

When the Call Vectoring feature is administered, an ACD split number cannot be assigned as the final point in a coverage path. However, a vector directory number can be assigned as the final point in a coverage path. When this is done, the flexibility of the Call Vectoring feature can be applied to the redirected call. The corresponding vector could be programmed to queue the redirected call to an ACD split (including an AUDIX or a Message Center split). Furthermore, the vector's processing could also vary by time of day (to provide night service) or by the status of the split's queue (to provide intraflow or interflow).

When a VDN is assigned as the final point in a coverage path, a principal's temporary bridged appearance is removed at the time that vector processing assumes control of a redirected call.

A vector can have a "route to" step that routes calls to an extension with coverage assigned. When this is done, the destination extension's coverage under all criteria is ignored.

If a redirected call covers to a VDN, the call Vectoring feature screens the calls to limit undesirable vector treatment. A Call Vectoring subroutine quickly scans the VDN's vector to make sure that it contains one of the following vector steps:

- Queue to main split (with staffed agents) step
- Route to step
- Forced disconnect ***with*** recorded announcement step (beginning with R2 V4, Issue 1.2).

For more information, refer to Table 34-B.

VDN Override does not apply to calls that are redirected to a VDN by the Call Coverage feature. For calls that are redirected to a VDN, the originally called terminal's name remains permanently associated with the call.

Call Forwarding—Busy and Don't Answer

The Call Forwarding—Busy and Don't Answer feature cannot be used to forward calls to a VDN.

With Call Vectoring assigned, an ACD split supervisor can use this feature to forward calls for the supervises individual extension. Without Call Vectoring assigned, Call Forwarding—Busy and Don't Answer is instead used to forward calls which are directed to the split's queue.

The destination for a "route to extension" step in a vector can be an extension with Call Forwarding—Busy and Don't Answer activated. When a VDN call routes to a forwarded extension, the VDN call will forward to and ring the forwarded-to extension if the forwarding extension is ***busy.*** When there is ***no answer*** at the forwarding extension, the VDN call will continue to ring the forwarding extension and will not forward.

Vector processing considers an extension with Call Forwarding—Busy and Don't Answer activated an invalid destination (that is, the step is skipped) unless the "route to extension" step is the final effective step in the vector.

Call Forwarding— Don't Answer

The Call Forwarding—Don't Answer feature cannot be used to forward calls to a VDN.

With Call Vectoring assigned, an ACD split supervisor can use this feature to forward calls directed to the supervisor's individual extension. Without Call Vectoring assigned, Call Forwarding— ***Busy and Don't Answer*** is instead used to forward calls which are directed to the splits queue. (This is true even if Call Forwarding—Don't Answer is assigned to the supervisor's class of service.)

The destination for a "route to extension" step in a vector can be an extension with Call Forwarding—Don't Answer activated. However, when there is no answer at the forwarding extension, the VDN call will continue to ring the forwarding extension and will not forward.

Vector processing considers an extension with Call Forwarding—Don't Answer activated an invalid destination (that is, the step is skipped) unless the "route to extension" step is the final effective step in the vector.

Call Forwarding—Follow Me

The Call Forwarding—Follow Me feature can be used to forward calls to a VDN. When this is done, the forwarded-to vector controls call processing for the forwarded call. For example, the call could enter an ACD split's queue (including AUDIX or Message Center) and be processed according to the vector's programming.

With Call Vectoring assigned, an ACD split supervisor can use this feature to forward calls for the supervisor's individual extension. Without Call Vectoring assigned, Call Forwarding—Follow Me is instead used to forward calls which are directed to the split's queue.

The destination for a "route to extension" step in a vector can be an extension with Call Forwarding—Follow Me activated. Vector processing considers an extension with Call Forwarding—Follow Me activated an invalid destination (that is, the step is skipped) unless the "route to extension" step is the final effective step in the vector.

VDN Override does not apply to calls that are forwarded to a VDN by the Call Forwarding—Follow Me feature. For calls that are forwarded to a VDN, the originally called terminal's name remains permanently associated with the call.

Call Waiting

If a single-appearance terminal is the destination of a "route to" command, an incoming call will not wait on the terminal. Instead, if the "route to" command is the last vector step, vector processing will attempt to redirect the call to an idle forwarding destination or coverage point. This final vector step is retried at 2-second intervals. If the "route to" command is not the last vector step, vector processing will continue with the next sequential step.

Centralized Attendant Service

Vector processing is not available for incoming calls to backup voice terminals at a branch location. Entering a VDN in Procedure 211, Word 2 as the extension number of a backup voice terminal is not allowed. When this is attempted, an administration error will occur.

Beginning with Issue 3.0 of DEFINITY Generic 2.1, an RLT (Release Link Trunk) can terminate to an ACD split, VDN (Vector Directory Number), or attendant console. The following CAS interactions apply to Issue 3.0 or later of DEFINITY Generic 2.1.

A centralized attendant or other answering position at the CAS main can extend (transfer) an RLT call to a VDN at a branch location. If the answering destination specified by the vector is an ACD split, when ringback tone is heard, the answering position should release the call within 4 seconds. This allows the call to enter vector processing.

A centralized attendant or other answering position at the CAS main cannot originate calls to a branch location by way of an RLT (Release Link Trunk). When this is attempted, the switch returns intercept tone. An answering position at the main can originate a call to a VDN at a branch location by way of a tie trunk.

An answering position (other than a CAS attendant) at the CAS main can receive either Call Identification Tones or ip tones, but not both. The same is true for origin announcements. If origin announcements (VDN-of-Origin, Queue-of-Origin, or City-of-Origin) are provided at the main, Call Identification Tones cannot be provided. However, an answering position that is equipped with a display voice terminal can be given information about the source of an incoming call. The combined information an answering position receives from zip tones, recorded announcements, and a display voice terminal is often a suitable substitute for Call Identification Tones. For more information, refer to *ACD From the Agent's Perspective in the* ACD feature description.

For RLTs that terminate to ACD splits or VDNs, the answering positions should be equipped with display voice terminals so that the user can distinguish between RLT and non-RLT calls. The reason is that the user operation for the Conference—Three Party and Transfer features is different for RLT calls. The Conference—Three Party and Transfer features work normally for non-RLT calls. For more information, refer to the User Operations section of this feature description.

For RLTs that terminate to ACD splits or VDNs, incoming calls should not be routed to a destination outside of the CAS arrangement (by way of Call Forwarding, Look-Ahead Interflow, or Call Vectoring "route to" steps). This type of routing disables the dropout and reuse capabilities that make RLTs desirable.

CDR (Call Detail Recording)

Call detail records show the extension number of the answering destination (rather than the VDN) as the called number when an internal or external call is processed by a vector.

VFCDR (Variable Format Call Detail Recording)

The Variable Format CDR feature can be administered to record the dialed VDN (instead of the trunk-group dial access code) as the calling number for calls placed to VDNs. In this way, when several VDNs complete calls to the same answering destination, the Variable Format CDR feature can provide hard-copy data for the calls placed to the separate VDNs.

Conference—Three Party

A multiappearance voice terminal user on a 2-party call is allowed to add the recipient of a VDN call to a 3-party conference. However, the second press of the CONFERENCE button in the conferencing operation is ignored until the recipient of the VDN call has actually answered the call. (That is, the second button press is ignored during vector processing.)

A single-appearance voice terminal user on a 2-party call is also allowed to add the recipient of a VDN call to a 3-party conference. However, the controlling party must execute the second button press (of the switchhook or the RECALL button) after the recipient of the VDN call has actually answered the call. If the controlling party executes the second button during vector processing, the soft held party is reconnected to the controlling party, and the conference attempt does not succeed.

When an agent adds another agent to an ACD call, an outgoing work-related call, or a personal call, the resulting conference is not considered a work-related activity for the second agent unless the second agent was reached by dialing a VDN. If the second agent was not reached by dialing a VDN, the second agent is not removed from the agent queue.

Beginning with Issue 3.0 of DEFINITY Generic 2.1, the operation of the Conference-Three Party feature has been changed for an ACD agent (or other voice terminal user) who extends an incoming RLT call to a branch location in a CAS arrangement. Refer to the Centralized Attendant Service interaction and to the User Operations section of this feature description for more information.

Beginning with DEFINITY Generic *2.2,* the switch notifies CMS if a measured ACD agent initiates a 3-party conference while handling an ACD call.

Display—Voice Terminal

When queue-status displays are assigned for Call Vectoring in the ACD environment, these displays use 8 characters (6 digits and 2 spaces) of the available 40 characters. Further, these 8 characters overlap with the source and destination fields on the 40-character display. Therefore, unless the source and destination names are fairly brief, these names are more likely to be truncated when queue-status displays are enabled.

DCS (Distributed Communications System)

A "route to" command can be used in vector processing to route calls to a distant DCS node. Beyond this functionality, no DCS transparency is provided. System 85 or

DEFINITY Generic 2 does not provide message correspondence over DCIU links that is unique to Call Vectoring. "Queue to main split" and "check backup split" commands can only be applied to ACD splits within the local DCS node.

Given these limitations, "route to" commands could be used to route calls to an associated extension number or VDN at a distant DCS node. The distant associated extension number or VDN can, in turn, terminate to a centralized AUDIX or Message Center queue.

DID (Direct Inward Dialing)

If the DID restriction is assigned to a VDN's class of service, this restriction is ***not applied*** to vector processing for the VDN. Even if this restriction is assigned to the class of service, DID calls are allowed to terminate to the vector.

Hunting

A vector directory number cannot be assigned to a Hunting sequence. When this is attempted, an administration error will occur.

A member of a hunt group can be assigned as the destination of a "route to" vector step. If the "route to" step is the final effective step in the vector and the routed-to extension is busy, vector processing checks for an idle extension in the hunt group. If the "route to" step is **not** the final effective step in the vector and the routed-to extension is busy, vector processing goes on to the next vector step wothout checking for an idle extension in the hunt group.

Interpartition Access

A voice terminal user (in a partition other than Extension Partition 0) is allowed to call (using an extension number) a VDN assigned to art extension partition in the same partition group or assigned to Extension Partition 0. If the user tries to call a VDN assigned to another group's extension partition, the switch will return intercept treatment.

The "route to" vector command can route calls to an extension in the VDN's partition group or to Extension Partition 0. If a "route to" command is programmed to route calls to another extension partition, the switch will treat a final effective "route to" step as a "stop" step. Otherwise, the "route to" step is ignored, and vector processing continues with the next sequential step.

Look-Ahead Interflow

The Call Vectoring feature is required at both the sending and receiving switches in a Look-Ahead Interflow configuration. For Look-Ahead Interflow, vector processing at the sending switch first decides whether the interflow operation is necessary. Subsequently, vector processing at the receiving switch makes the decision whether to accept or reject the Look-Ahead Interflow call.

The full flexibility of the Call Vectoring feature can be utilized at both the sending and receiving switches. Branches of any type can appear in the vector programs at both switches. Vector commands of any type can also appear within these vectors. Refer to the Look-Ahead Interflow feature description for a description of these capabilities.

DNIS Names and VDN Override

For direct calls to a VDN, the VDN Overrride option of the Call Vectoring feature can always apply to a VDN call within the switch that originally received the VDN call. Before the sending switch executes a Look-Ahead Interflow "route to" step destined for the receiving switch, VDN Override can change the name associated with the originally called VDN.

As the Look-Ahead Interflow "route to" step is executed, the sending switch sends the most recent VDN name in the DNIS Name field of the Look-Ahead Interflow IE (Information Element). When this name arrives at the receiving switch, this switch will use the received name regardless of its own VDN Override assignments.

If, because no names were assigned, the sending switch sends a blank DNIS Name field in the Look-Ahead Interflow IE, then the receiving switch will apply the name associated with its receiving VDN to the call, and this name can change according to the rules of VDN Override.

Modem Pooling

The "route to" step of the Call Vectoring feature is compatible with the Modem Pooling feature Whenever a conversion resource is needed to complete the call to a "route to" step's destination, these conversion resources will be inserted.

Multiple Listed Directory Numbers

A vector directory number cannot be assigned as a system LDN in Procedure 204, Word 1. When this is attempted, an administration error will occur.

In order to route public network calls to the attendant queue using a VDN, the associated vector must contain a "route to" command that directs these calls to the attendant queue. However, vector treatment of calls within the attendant queue is not available. Vector steps (for example, recorded announcements, conditional go to steps, forced busy, and forced disconnect) can be applied to these calls ***before they enter the queue.*** However, once a call enters the attendant queue, vector processing stops.

Music-on-Hold Access

The music interface that can be provided with a "delay (wait)" step is functionally independent of the system-wide Music-on-Hold feature. To provide music for vector processing only, the music source should be provided, and the Music-on-Hold software should be fully administered. Then, to turn off the ***regular*** Music-on-Hold feature, Field 11 of Procedure 275, Word 1 is set to "0."

Override

Overrride calls cannot be placed to VDNs. When this is attempted, the switch returns intercept tone.

Precedence Calling

The precedence level of AUTOVON calls that are directed to or forwarded to a VDN are checked. If the precedence level is higher than "Routine," the call does not terminate to the vector. Instead, the call is redirected to the attendant queue.

Priority Calling

Priority calls cannot be placed to VDNs. When this is attempted, the switch returns intercept tone.

Restriction—Attendant Control of Voice Terminal

An attendant is not allowed to activate an Attendant Control of Voice Terminals restriction against a VDN. When this is attempted, the switch returns intercept tone.

Restriction—Code Restriction

The Code Restriction feature does not prevent "route to" steps from routing calls to destinations outside destinations outside the switch. "Route to" steps use network routing software, which does not make Code Restriction checks.

Restriction—Toll Restriction

Toll Restriction does not prevent "route to" steps from routing calls to destinations outside the switch. If Toll restriction is assigned to a VDN's class of service, the assignment is ignored.

Restriction—Voice Terminal Restrictions

Voice terminal restrictions ***do not apply*** to VDNs. For example, if Termination Restriction is assigned to Class of Service 1, and Class of Service 1 is assigned to VDN 7300, this restriction is ignored by the Call Vectoring feature. Calls are allowed to terminate to the VDN.

Voice terminal restrictions do not prevent "route to" steps from routing calls to an answering destination. For example, if Origination or Outward Restriction is assigned to a VDN's class of service, the assignment is ignored.

Route Advance

Using extension number steering to steer to a trunk-group dial access code, the first trunk group in a Route Advance sequence can be programmed as the destination of a "route to" vector step. When this is done, the call can route over an idle trunk in an alternate trunk group in the sequence.

Tenant Services

Trunk group-oriented routing to vectors is not partitioned. There are no checks in Procedure 031, Word 2 to ensure that the partition of an automatic-in trunk group matches the partition of the assigned VDN. It is the responsibility of the system manager to ensure that these partition numbers match.

Digit-oriented routing to vectors is partitioned. When a VDN's digits are passed to System 85 or DEFINITY Generic 2 from the serving switch or dialed from inside the switch, the Tenant Services feature makes the necessary partitioning checks. Vector directory numbers are assigned to partitions using Procedure 000, Word 4.

The "queue to main split" and "check backup split" vector commands are not partitioned. For vectors containing either of these commands, there are no checks in call-processing software to ensure that the extension partition of the answering agent matches the extension partition of the VDN or the split supervisor. It is the responsibility of the System Manager to ensure that "queue to main split" and "check backup split" commands do not cause calls to cross partition boundaries.

The "route to" vector command can route calls to an extension in the VDN's extension partition or to Extension Partition 0. If a "route to" command is programmed to route calls to another extension partition, the switch will treat a final effective "route to" step as a "stop" step. Otherwise, the "route to" step is ignored, and vector processing continues with the next sequential step.

If a vector contains a "route to" step that routes calls to the shared attendant queue, the call will terminate to an attendant partition that is assigned to the VDN's extension partition. If no attendant partition is assigned to the VDN's extension partition, the switch returns Intercept Treatment to the calling party.

If a vector contains a "route to" step that routes calls outside the switch, the call uses the FRL assigned to the VDN and the Call Category assigned to the VDN's extension partition.

In Procedure 030, Word 1, one Abbreviated Dialing group list (5 Abbreviated Dialing group lists beginning with DEFINITY Generic 2.2) is assigned to control the destinations of "route to" commands. The controlling terminal for this group list (assigned in Procedure 059, Word 1) can belong to any extension partition. However, it is strongly recommended that this controlling terminal belong to Extension Partition 0.

The calling party announcements for the Call Vectoring feature are not partitioned. These announcements are available to every extension partition. VDNs in various extension partitions can terminate to several vectors containing "announcement" steps that all request the same recorded announcement.

Timed Reminder

The Timed Reminder feature does not apply to calls that an attendant extends to a VDN. Once an attendant-extended call enters vector processing, this call is not timed and no reminder will be given to the attendant.

Transfer

Voice terminal users (including ACD agents) can transfer calls to a VDN. When this is done, the transferred-to vector controls call processing for the transferred call.

When an agent adds another agent to an ACD call, an outgoing work-related call, or a personal call, the transferred call is not considered a work-related activity for the second agent unless the second agent was reached by dialing a VDN. If the second agent was not reached by dialing a VDN, the second agent is not removed from the agent queue.

Beginning with Issue 3.0 of DEFINITY Generic 2.1, the operation of the Transfer feature has been changed for an ACD agent (or other voice terminal user) who extends an incoming RLT call to a branch location in a CAS arrangement. Refer to the Centralized Attendant Service interaction and to the User Operations section of this feature description for more information.

Beginning with DEFINITY Generic 2.2, the switch notifies CMS if a measured ACD agent transfers an ACD call.

Trunk Verification—Voice Terminal

Entering a VDN in Procedure 285 as the designated internal extension or the remote maintenance terminal for Trunk Verification is denied. When this is attempted, an administration error will occur.

Unattended Console Service-Preselected Call Routing

Vector processing is not available for incoming calls to the default voice terminal for Unattended Console Service. Entering a VDN in Procedure 275, Word 2 as the default extension number is not allowed. When this is attempted, an administration error will occur.

Entering a VDN as the "Trunk-to-Night Terminal" extension number for a trunk group in Procedure 116, Word 1 or Procedure 150 is not allowed. When this is attempted, an administration error will occur.

An attendant is not allowed to establish a "Trunk-to-Night Terminal" assignment by dialing a VDN as the night-terminal extension. When this is attempted, the switch returns intercept tone.

Beginning with Issue 1.2 of R2 V4, Call Vectoring can indirectly perform night service for the attendant queue. The designated terminal, the common night terminal, or the default night terminal for preselected call routing can activate Call Forwarding—Follow Me to forward all calls to a VDN (Vector Directory Number). When this is done, every call to the night terminal enters the VDN's vector processing and receives the normal treatment programmed within the vector.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, the WCR feature is required to route calls to destinations outside the switch. To route calls over a private network, the "Standard Networking" field (Procedure 276, Word 1) must be assigned.

FRL (Facilities Restriction Level)

The FRL administered to a VDN's class of service in Procedure 010, Word 3 is used to determine whether a call is allowed to route over available network facilities.

Dialing Plan

When the WCR feature is used to route calls over private network facilities, the contents of a vector-group list item for a "route to" step must conform to the dial plan of the network. When System 85s, DEFINITY Generic 2 switches, or DIMENSION FP 8, Issue 3 switches are part of a private network, the dial plan can have one of the following forms:

- RNX (3-Digit Location Code) + XXXX (4-Digit Extension Number)
- RN (2-Digit Location Code) + XXXX (4-Digit Extension Number)
- RNX (3-Digit Location Code) + XXX (3-Digit Extension Number)
- RN (2-Digit location code) + XXX (3-Digit Extension Number).

When calls are routed over public network facilities, the contents of a vector-group list item for a "route to" step must conform to the public network rules for DDD (Direct Distance Dialing). The DDD formats for the public network can have one of the following forms:

- NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- NPA (3-Digit Area Code) + NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- IXC (Interexchange Carrier) + NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- International Telephone Number.

Besides conforming to the dial plan for the network, the vector-group list items for "route to" steps must be prefixed by the appropriate network DAC. For public network routing a prefix digit for toll calls (typically the digit 1) or an international access code may also be required.

Besides conforming to the dialing plan for the network, a pattern must be translated for the destination digits of a "route to" step. When the digits of a "route to" destination are undefined in WCR or translate to the intercept pattern (VNI 0), the "route to" step is treated as having an invalid destination. That is, if the "route to" step is the final effective step in the vector, vector processing treats the step as a "stop" step. Otherwise, vector processing continues with the next sequential step in the vector.

Queuing

Queuing does not apply to calls that are routed outside the switch by a "route to" step. If every preference is busy, vector processing will either retry routing the call at 2-second intervals (if final effective step) or continue vector processing with the next sequential step in the vector.

Bearer Capability Classification

For voice calls, the Bearer Capability Class of Service (BCCOS) is not a significant consideration. This is because voice calls are usually compatible with any carrier facility. However, the WCR feature does check the BCCOS of calls that are routed outside the switch by a "route to" step. Therefore, when applicable, the BCC of the outgoing preference must be compatible with the BCC in the local calling party's COS or the BCC assigned to the incoming trunk group.

Unauthorized Call Control

Calls that route outside the switch by way of a "route to" step can be blocked by unauthorized call control. Whenever a vector-group list item contains a digit string that is marked for call control, vector processing either treats the "route to" step as a "stop" step (if the final effective step) or continues with the next sequential step in the vector.

WCR Toll Restriction

Toll Restriction does not prevent "route to" steps from routing calls to destinations outside the switch. If Toll restriction is assigned to a VDN's class of service, the assignment is ignored.

Restricting Feature Use

Incoming calls to VDNs are not restricted. The following is a summary of these interactions:

- DID restriction is ignored.
- Incoming Attendant Control of Voice Terminal restrictions cannot be activated.
- Incoming Voice Terminal Restrictions are ignored.

The routing of "route to" steps to an external destination is only controlled by the FRL of the VDN's class of service. The following is a summary of these interactions:

- "Route to" steps can only route a call to the private or public network if the VDN's FRL (or Alternate FRL) is high enough to allow access to an available preference in the corresponding network routing pattern.
- ARS / WCR Toll Restriction is ignored.
- Outgoing Attendant Control of Voice Terminal restrictions cannot be activated.
- Code Restrictions do not apply.
- Toll Restriction does not apply.
- Outgoing Voice Terminal Restrictions are ignored.

Hardware Requirements

Beyond the basic hardware chosen to implement other related features [for example, the regular ACD feature, the Music-on-Hold Access feature, the CMS (Call Management System), or recorded announcements there is no additional hardware required to implement the Call Vectoring feature.

Feature Administration

Assignment of the Call Vectoring feature is on a per-trunk group and on a per-vector directory number basis.

On System 85 switches, Call Vectoring is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, Call Vectoring is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — CALL VECTORING

PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the port type as "VDN" (Field 8), and assigns the class of service to a vector directory number.	Yes
010	3	Assigns an FRL to a VDN's class of service.	Yes
011	1	Assigns a vector directory number as the final point in a a coverage path.	Yes
012	1	Assigns a name to a vector directory number or to an incoming trunk group that is controlled by a vector.	Yes
030	1	Assigns the Abbreviated Dialing group-list number to be used for Call Vectoring. Beginning with DEFINITY Generic 2.2, five group lists can be used for Call Vectoring (group list items are used to specify the destinations for "route to" commands).	Yes

ADMINISTRATION PROCEDURES — CALL VECTORING (Contd)			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
030	2	Displays the AUDIX or Message Center adjunct number associated with a vector. Beginning with DEFINITY Generic 2.2, use this procedure to add a new vector or to change, copy, or remove an existing vector. (For System 85 and DEFINITY Generic 2.1, Procedure 030, Word 3 is used to add, change, copy, or remove vectors.)	Yes
030	3	Specifies the desired steps for a vector.	Yes
031	1	Assigns (not SMT) a vector and an ICI display to a VDN. Assigns CMS measurement to the VDN. Specifies whether the return call operation is associated with AUDIX or Message Center, and displays the machine number of the adjunct. Also, assigns the VDN Override option to a VDN.	Yes
031	2	Assigns an automatic trunk group to a VDN. Beginning with Issue 3.0 of DEFINITY Generic 2.1, an RLT trunk group can terminate to a VDN. Use this procedure to specify the VDN to which a trunk group terminates. Beginning with DEFINITY Generic 2.2, if the Expert Agent Selection feature is active, use this procedure to assign a primary, secondary, or tertiary skill to a VDN. Use Procedure 031, Word 3 to specify the VDN to which a trunk group terminates.	Yes*
031	3	Beginning with DEFINITY Generic 2.2, use this procedure to specify the VDN to which a trunk group terminates.	Yes*
032	1	Displays the vectors that queue to or check the queue of a particular split (either main or backup).	Yes
033	1	Assigns a VDN-of-origin announcement to a vector directory number.	Yes
* Display only procedure for the SMT.			

ADMINISTRATION PROCEDURES — CALL VECTORING (Contd)			
PROCEDURE	WORD	PURPOSE	SMT
059	1 & 2	Assigns the group list that is used for Call Vectoring. Beginning with DEFINITY Generic 2.2, five group lists can be used for Call Vectoring (group list items are used to specify the destinations for "route to" commands).	Yes
100	1	Assigns a trunk group as a "Vectoring Announcement" trunk group. The applicable encode is as follows: 90 Vectoring recorded announcement.	No
115	—	Assigns termination to a CAS attendant, Special Services attendant, or ACD split; and CMS measurement to an incoming trunk group. This procedure also displays incoming trunk group termination to a VDN (assigned in Prodedure 031 Word 2.	No
150	—	Assigns the recorded announcement number (16 - 99) (1 - 255 beginning with DEFINITY Generic 2.2) to the equipment location of an SN231 or TN763C recorded announcement trunk, and designates whether the announcement is continuous (for example, city-of-origin, queue-of-origin, or VDN-of-origin).	No
276	1	Assigns Call Vectoring to the feature group class of service.	No
354	1	Assigns blocks of extension numbers to be used as vector directory numbers.	No

Vector Administration for System 85 and DEFINITY Generic 2.1 Using the MAAP, SMT, or Manager II

Beginning with DEFINITY Generic 2.2, Procedure 030, Word 2 and Word 3 are used to administer vectors. Refer to Vector Administration for DEFINITY Generic 2.2 Using the MAAP, SMT, or Manager II for more information.

For System 85 and DEFINITY Generic 2.1, Procedure 030, Word 3 is used to administer vectors (add a new vector or change, copy, or remove an existing vector) with the MAAP, SMT, or Manager II. To make it possible to administer a vector while the switch is processing calls, a "scratch pad" is used during vector administration and then the vector is transferred to permanent memory.

Add (program) a new vector.

To add a new vector, follow these three steps:

1. Clear the scratch pad

   Two entries are required in Procedure 030, Word 3: enter a "0" in Field 1, and a "1" in Field 3. (Dashes must appear in the rest of the fields.) Then, do a "Display Execute" followed by a "Remove Execute."

2. Program each vector step in order.

   Enter the step number (from 1 to 15) in Field 2, a "0" in Field 3,* and the necessary command criteria in Fields 4 through 12. Then, do an "Add Execute" for each vector step.

3. Transfer the new vector (as a complete unit) to permanent memory. †

   Two entries are required in Procedure 030, Word 3: enter the vector number (from 1 through 128) in Field 1, and a "1" in Field 3. (Dashes must appear in the rest of the fields.) Then, do an "Add Execute."

Change an existing vector.

To change an existing vector, follow these three steps:

CAUTION: Step 3 is very important. If Step 3 is skipped, any changes made to the vector (in Step 2) will be lost.

1. Call the vector to be changed into the scratch pad.

   Two entries are required in Procedure 030, Word 3: enter the vector number (from 1 through 128) in Field 1, and a "1" in Field 3. (Dashes must appear in the rest of the fields.) Then, do a "Display Execute."

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* Instead of a "0," a "—" can be entered in Field 3.

† After successfully transferring the vector to permanent memory, Step 1 (clear the scratch pad) does not need to be executed. The scratch pad is automatically cleared.

2. Add, remove, or change a vector step.

   a. Adding a vector step requires the following entries in Procedure 030, Word 3: enter the step number (from 1 through 15) in Field 2, a "0" in Field 3, and the necessary command criteria in Fields 4 through 12. Then, do an "Add Execute."

      NOTE: The step numbers of the steps following the added step are automatically incremented.

   b. Removing a vector step requires two entries in Procedures 030, Word 3: enter the number of the step to be removed (from 1 to 15) in Field 2, and a "0" in Field 3. Then, do a "Display Execute" followed by a "Remove Execute."

      NOTE: The step numbers of the steps following the removed step are automatically decremented.

      CAUTION: Whenever a vector step is either ***added to*** or ***removed from*** a vector, the "go to step" commands in the vector should be carefully examined, and, if necessary, changed. It is quite likely that these "go to step" commands are now passing control to the wrong vector step.

   c. Changing a vector step requires two entries in Procedure 030, Word 3: enter the number of the step to be changed in Field 2, and a "0" in Field 3. Then, do a "Display Execute." Next, change the appropriate entries in Fields 4 through 12. Then, do a "Change Execute."

3. Transfer the vector (as a complete unit) to permanent memory.

   Two entries are required in Procedure 030, Word 3: enter the ***same vector number*** (the vector number entered in Step 1) in Field 1, and a "1" in Field 3. (Dashes must appear in the rest of the fields.) Then, do a "Change Execute."

Copy an existing vector.

To copy an existing vector, follow these two steps:

1. Call the vector to be copied into the scratch pad.

   Two entries are required in Procedure 030, Word 3: enter the vector number (from 1 to 128; for example, 128) in Field 1, and a "1" in Field 3. (Dashes must appear in the rest of the fields.) Then, do a "Display Execute."

2. Write the scratch-pad vector to another vector number.

   Two entries are required in Procedure 030, Word 3: enter a different vector number (for example, 43) in Field 1, and a "1" in Field 3. (Dashes must appear in the rest of the fields.) Then, do an "Add Execute."

Remove an existing vector.

To remove an existing vector:

1. Call the vector to be removed into the scratch pad and remove the vector.

   Two entries are required in Procedure 030, Word 3: enter the number of the vector to be removed (from 1 to 128) in Field 1, and a "1" in Field 3. (Dashes must appear in the rest of the fields.) Then, do a "Display Execute" followed by a "Remove Execute."

The following pages describe the valid field entries for Procedure 030, Word 3 and show examples of the required administration for application vectors B, F, G, and T (presented earlier under the heading "Sample Applications of Vectoring").

PROCEDURE 030, WORD 3

Field 1: This field identifies the ***vector number*** (0, or from 1 to 128). A vector number must be entered when a vector is either called to the scratch pad or written to permanent memory. "0" is also a legal entry in this field. This value is used to open an ***empty*** scratch pad.

Field 2: This field identifies the ***step number*** within a vector (from 1 to 15). A step number must be entered whenever a vector step is added, removed, or changed. A step number cannot be entered if a "1" is entered in Field 3.

Field 3: This field is used to ***access the permanent vector.*** This field is set to "1" to call an entire permanent vector to the scratch pad for modification. Field 3 is also set to "1" to begin the process of transitioning the entire scratch pad vector to the permanent vector. Field 3 must be set to "0" when a vector step is being added, removed, or changed.

Field 4: This field identifies the ***step type.*** The numerical encodes of the nine vector commands are:

1 queue to main split	4 announcement	7 forced disconnect
2 check backup split	5 wait	8 forced busy
3 route to	6 go to step	9 stop

Field 5: This field identifies the ***destination*** of Step Types:

1 split number 1 to 60	3 list item 1 to 95	6 step number 1 to 15
2 split number 1 to 60	4 annct. 16 to 99	7 —, or annct. 16 to 99

Field 6: This field identifies the ***priority level*** (from 0 to 3) of Step Types 1,2, and 6.

Field 7: This field specifies the ***condition*** of Step Types 2,5, and 6.*

Field 8: This field specifies the ***threshold*** (conditional parameter) of the condition specified in Field 7.*

Field 9: This field identifies the ***split number*** referred to in Fields 7 and 8. Legal entries are split numbers 1 through 60 or "—" (the split currently queued to).

Field 10: This field identifies the ***day of week*** in a "time-of-day" branch for Step Type 6. Legal entries are 0 through 7. "0" is everyday, "1" is Monday, and "7" is Sunday.

Field 11: This field identifies the ***hour*** in a "time-of-day" branch for Step Type 6. Legal entries are 0 (midnight) through 23 (11:00 p.m.).

Field 12: This field identifies the ***minutes*** of a "time-of-day" branch for Step Type 6. Legal entries are 0 through 59.

---

* Refer to Procedure 030, Word 3 (prior to DEFINITY Generic 2, Procedure 030, Word 3A) for a complete display of the conditions and thresholds.

Administration of Vector B Using the Manager II Enhanced Mode

```
              ENHANCED MODE - PROCEDURE : 030, WORD: 3
                  CALL VECTORING - PROGRAMMING VECTORS

 1.          Vector Number:  0        START/END TIME OF DAY
 2.            Step Number: --          10.     Day: -
 3. Access Permanent Vector: 1          11.    Hour: --
                                        12.  Minute: --
ACTION
   4.   Step Type: --
   5.  Destination: ---

 6. Priority Level: -

CRITERIA
   7. Condition: -
   8. Threshold: ----

 9. Split Number: --


Connected to CC0 ON-LINE | MAJOR | MINOR | RUN TAPE | BUSY OUT | IN USE | WAIT

enter command:
          |         | F3 DATA |         | F5 HELP | F6 FIELD | F7 INPUT | F8 CMDS
```

dx rx (display execute, remove execute)

```
              ENHANCE MODE - PROCEDURE: 030, WORD: 3
                  CALL VECTORING - PROGRAMMING VECTORS

 1.          Vector  Number: ---      START/END TIME OF DAY
 2.            Step  Number: 1          10.     Day: -
 3.    Access Permanent Vector: 0       11.    Hour: --
                                        12.  Minute: --
ACTION
   4.   Step Type: 1
   5.  Destination: 9

 6. Priority Level: 0

CRITERIA
   7. Condition: -
   8. Threshold: ----

 9. Split Number: --


Connected to CC0 ON-LINE | MAJOR | MINOR | RUN TAPE | BUSY OUT | IN USE | WAIT

enter command:
          |         | F3 DATA |         | F5 HELP | F6 FIELD | F7 INPUT | F8 CMDS
```

ax rs (add execute, reset)

```
                 ENHANCED MODE - PROCEDURE: 030, WORD: 3
                    CALL VECTORING - PROGRAMMING VECTORS

 1.          Vector Number: ---        START/END TIME OF DAY
 2.            Step Number:  2           10.     Day: -
 3.   Access Permanent Vector: 0         11.    Hour: --
                                         12.  Minute: --
ACTION
   4.   Step Type:  6
   5.  Destination:  4

    Priority Level: 0

CRITERIA
   7. Condition: 4
   8. Threshold:  30

    Split Number: --



Connected to CC0 ON-LINE  MAJOR  MINOR  RUN TAPE  BUSY OUT  IN USE  WAIT

enter command: _
                    F3 DATA          F5 HELP  F6 FIELD  F7 INPUT  F8 CMDS
```

ax rs (add execute, reset)

```
                 ENHANCED MODE - PROCEDURE: 030, WORD: 3
                    CALL VECTORING - PROGRAMMING VECTORS

 1.          Vector Number: ---        START/END TIME OF DAY
 2.            Step   Number:  3         10.     Day: -
 3.  Access Permanent Vector: 0          11.    Hour: --
                                         12.  Minute: --
ACTION
   4.   Step Type:  9
   5.  Destination: ---

    Priority Level: -

CRITERIA
   7. Condition: -
   8. Threshold: ----

    Split Number: --



Connected to CC0 ON-LINE  MAJOR  MINOR  RUN TAPE  BUSY OUT  IN USE  WAIT

enter command: _
                    F3 DATA          F5 HELP  F6 FIELD  F7 INPUT  F8 CMDS
```

ax rs (add execute, reset)

```
                ENHANCED MODE - PROCEDURE: 030, WORD: 3
                  CALL VECTORING - PROGRAMMING VECTORS

 1.           Vector Number: ---        START/END TIME OF DAY
 2.             Step Number:  4          10.     Day: -
 3. Access Permanent Vector: 0           11.    Hour: --
                                         12.  Minute: --
ACTION
   4.    Step Type:  4
   5. Destination:  20

    Priority Level: -

CRITERIA
   7. Condition: -
   8. Threshold: ----

 9. Split Number: --

Connected to CC0 ON-LINE   MAJOR  MINOR  RUN TAPE  BUSY OUT  IN USE  WAIT

enter command:
               F3 DATA          F5 HELP  F6 FIELD  F7 INPUT  F8 CMDS
```

ax rs (add execute, reset)

```
                ENHANCED MODE - PROCEDURE: 030, WORD: 3
                  CALL VECTORING - PROGRAMMING VECTORS

 1.           Vector Number:   1        START/END TIME OF DAY
 2.             Step Number: --          10.     Day: -
 3. Access Permanent Vector: 1           11.    Hour: --
                                         12.  Minute: --
ACTION
   4.    Step Type: --
   5. Destination: ---

 6. Priority Level: -

CRITERIA
   7. Condition: -
   8. Threshold: ----

 9. Split Number: --

Connected to CC0 ON-LINE   MAJOR  MINOR  RUN TAPE  BUY OUT  IN USE  WAIT

enter command:
               F3 DATA          F5 HELP  F6 FIELD  F7 INPUT  F8 CMDS
```

ax (add execute)

Administration of Vector F Using the MAAP, SMT, or Manager II Basic Mode

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	0	- -	1	- -	- - -	-	-	- - - -	- -	-	- -	- -	030

Display Execute, Remove Execute

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	1	0	1	6	0	-	- - - -	- -	-	- -	- -	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	2	0	6	5	0	4	1 6	- -	-	- -	- -	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	3	0	6	6	0	4	8	- -	-	- -	- -	030

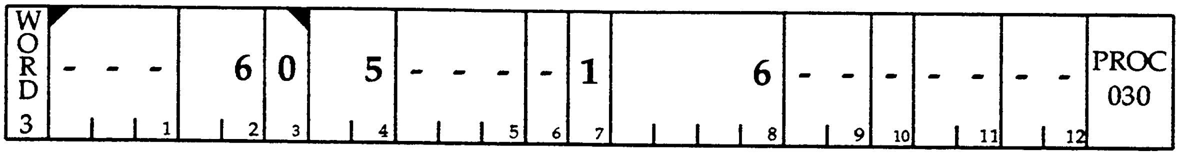
Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	4	0	6	7	-	-	- - - -	- -	-	- -	- -	030

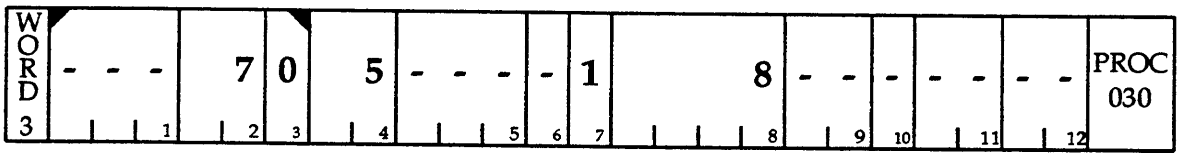
Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	5	0	5	- - -	-	1	6	- -	-	- -	- -	030

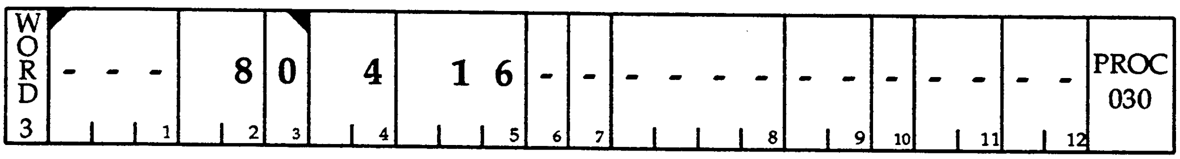
Add Execute, Reset



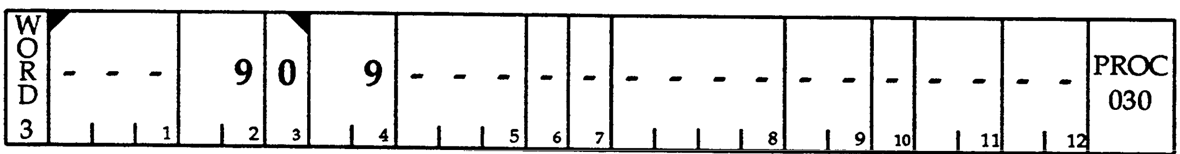
Add Execute, Reset



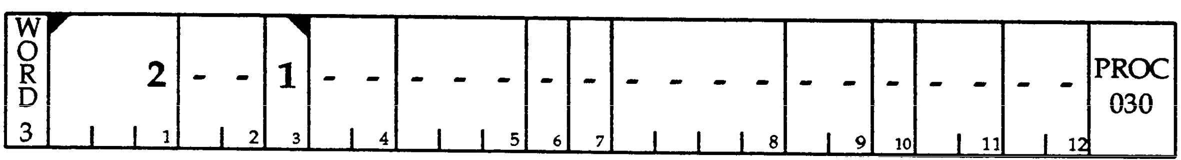
Add Execute, Reset



Add Execute, Reset



Add Execute, Reset



Add Execute

Administration of Vector G Using the MAAP, SMT, or Manager II Basic Mode

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	0	- -	1	- -	- - -	-	-	- - - -	- -	-	- -	- -	

Display Execute, Remove Execute

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1	0	6	8	-	8	- - - -	- -	0	1 7	0	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1	0	6	8	-	9	- - - -	- -	0	7	0	

Add Execute, Reset

(Steps 2 through 7 represent vector processing during work hours.)

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	7	0	9	- - -	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	8	0	3	9 1	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	3	- -	1	- -	- - -	-	-	- - - -	- -	-	- -	- -	

Add Execute

Administration of Vector T Using the W, SMT, or Manager II Basic Mode

WORD	1	2	3	4	5	6	7	8	9	10	11	12	PROC
3	0	- -	1	- -	- - -	-	-	- - - -	- -	-	- -	- -	030

Display Execute, Remove Execute

WORD	1	2	3	4	5	6	7	8	9	10	11	12	PROC
3	- - -	1	0	1	5 9	0	-	- - - -	- -	-	- -	- -	030

Add Execute, Reset

WORD	1	2	3	4	5	6	7	8	9	10	11	12	PROC
3	- - -	2	0	6	1 5	-	8	- - - -	- -	6	0	0	030

Add Execute, Reset

WORD	1	2	3	4	5	6	7	8	9	10	11	12	PROC
3	- - -	2	0	6	1 5	-	9	- - - -	- -	7	0	0	030

Add Execute, Reset

WORD	1	2	3	4	5	6	7	8	9	10	11	12	PROC
3	- - -	3	0	6	1 5	-	8	- - - -	- -	0	1 6	3 0	030

Add Execute, Reset

WORD	1	2	3	4	5	6	7	8	9	10	11	12	PROC
3	- - -	3	0	6	1 5	-	9	- - - -	- -	0	7	3 0	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	4	0	6	1 2	-	4	3 5	- -	-	- -	- -	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	5	0	6	1 0	-	6	3 6	- -	-	- -	- -	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	6	0	5	- - -	-	1	3 0	- -	-	- -	- -	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	7	0	4	1 6	-	-	- - - -	- -	-	- -	- -	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	8	0	5	- - -	-	2	2 0	- -	-	- -	- -	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC
	- - -	9	0	6	7	-	-	- - - -	- -	-	- -	- -	030

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1 0	0	2	1	0	5	1 0	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1 1	0	6	6	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1 2	0	4	1 7	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1 3	0	8	- - -	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1 4	0	9	- - -	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1 5	0	3	9 3	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	1 2 3	- -	1	- -	- - -	-	-	- - - -	- -	-	- -	- -	

Add Execute

(Continuation vector)

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	0	- -	1	- -	- - -	-	-	- - - -	- -	-	- -	- -	

Display Execute, Remove Execute

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	1	0	4	1 8	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	- - -	2	0	7	- - -	-	-	- - - -	- -	-	- -	- -	

Add Execute, Reset

WORD 3	1	2	3	4	5	6	7	8	9	10	11	12	PROC 030
	1 2 4	- -	1	- -	- - -	-	-	- - - -	- -	-	- -	- -	

Add Execute

Vector Administration for DEFINITY Generic 2.2 Using Manager II

Beginning with DEFINITY Generic 2.2, Procedure 030, Word 3 is used to program the steps of a vector and Procedure 030, Word 2 is used to transfer a vector from the scratch pad to permanent memory. Prior to DEFINITY Generic 2.2, Procedure 030, Word 3 was used to program the steps of a vector and to transfer a vector from the scratch pad to permanent memory.

Add (program) a new vector.

To add a new vector, follow these steps:

1. Program the vector steps.

   Using Procedure 030, Word 3, program the vector steps in the desired order. After entering each vector step, do an "Add Execute."

2. Transfer the vector to permanent memory.

   Using Procedure 030, Word 2, enter the number of the vector you want to change an "Add Execute."

Change an existing vector.

To change an existing vector, follow these steps:

> **CAUTION:** Step 3 is very important. If Step 3 is skipped, any changes made to the vector (in Step 2) will be lost.

1. Call an existing vector into the scratch pad

   Using Procedure 030, Word 2, enter the number of the vector you want to change in Field 1. Then, do a "Display Execute."

2. Add, remove, or change vector steps.

   Using Procedure 030, Word 3, add, remove, or change the appropriate vector steps.

   > **NOTE:** The step numbers of the steps following an added or removed step are automatically incremented or decremented.

   > **CAUTION:** Whenever a vector step is either added to or ***removed from*** a vector, the "go to step" commands in the vector should be carefully examined, and, if necessary, changed. It is likely that these "go to step" commands are now passing control to the wrong vector step.

3. Transfer the vector to permanent memory.

   Using Procedure 030, Word 2, enter the number of the changed vector in Field 1. Then, do a "Change Execute."

Copy an existing vector.

To copy an existing vector, follow these steps:

1. Call an existing vector into the scratch pad

   Using Procedure 030, Word 2, enter the number of the vector you want to copy in Field 1 (for example 100). Then, do a "Display Execute."

2. Change the vector number.

   Using Procedure 030, Word 2, enter a different (unused) vector number in Field 1 (for example 200). Then, do an "Add Execute."

Remove an existing vector.

To remove an existing vector:

1. Call an existing vector into the scratch pad and remove the vector.

   Using Procedure 030, Word 2, enter the number of the vector you want to remove in Field 1. Then, do a "Remove Execute."

Refer to DEFINITY Generic 2 Administration Procedures (555-105-506) for more information about Procedure 030, Words 2 and 3.

The following example shows the Manager II screen entries that are required to add Sample Vector A (presented earlier under the heading "Sample Applications of Vectoring") to the system as Vector Number 200.

```
                    ENHANCED MODE - PROCEDURE: 030, WORD: 3
                   CALL VECTORING - PROGRAMMING VECTOR STEPS

 1. Step Number: 1

ACTION
   2.   Step Type: 6
   3.  Destination:  4
 4. Priority Level: 0

CRITERIA
   5.  Condition:  4
   6.  Threshold:    7
 7.  Split/Skill: ---

START/END TIME OF DAY
   8.    Day: -
   9.   Hour: --
  10. Minute: --


Connected to CC0 ON-LINE  MAJOR  MINOR  RUN TAPE  BUSY OUT  IN USE  WAIT

enter command: _
                  3 Form         5 Help  6 Field  7 Input  8 Cmds
```

ax rs (add execute, reset)

```
                ENHANCED MODE - PROCEDURE: 030, WORD:3
                ALL VECTORING - PROGRAMMING VECTOR STEPS

 1.  Step Number:  2

ACTION
   2.   Step Type: 1
   3.  Destination: 15
 4. Priority Level: 0

CRITERIA
   5.  Condition: --
   6. Threshold: ----
 7.  Split/Skill: ---

START/END TIME OF DAY
   8.    Day: -
   9.   Hour: --
  10. Minute: --


Connected to CC0 ON-LINE   MAJOR   MINOR   RUN TAPE   BUSY OUT   IN USE   WAIT

enter ccommand:
                    3 Form              5 Help   6 Field   7 Input   8 Cmds
```

ax rs (add execute, reset)

```
                ENHANCED MODE - PROCEDURE: 030, WORD: 3
                ALL VECTORING - PROGRAMMING VECTOR STEPS

 1. Step  Number: 3

ACTION
   2.  Step  Type:  9
   3.  Destination: ---
 4. Priority Level: -

CRITERIA
   5.  Condition: --
   6.  Threshold: ----
 7.  Split/Skill: ---

START/END TIME OF DAY
   8.    Day: -
   9.   Hour: --
  10.  Minute: --


Connected to CC0 ON-LINE   MAJOR   MINOR   RUN TAPE   BUSY OUT   IN USE   WAIT

enter  command:
                    3  Form             5 Help   6 Field   7 Input   8 Cmds
```

ax rs (add execute, reset)

```
                    ENHANCED MODE - PROCEDURE: 030, WORD: 3
                    ALL VECTORING - PROGRAMMING VECTOR STEPS

 1.  Step Number:  4

ACTION
   2.  Step  Type:   7
   3.  Destination:  17
 4. Priority Level: -

CRITERIA
   5.  Condition: --
   6.  Threshold: ----
 7.   Split/Skill: ---

START/END TIME OF DAY
   8.     Day: -
   9.    Hour: --
  10.  Minute: --


Connected to CC0 ON-LINE ♥  MAJOR   MINOR   RUN TAPE   BUSY OUT   IN USE   WAIT

enter command:
                    3 Form             5 Help   6 Field   7 Input   8 Cmds
```

ax rs (add execute, reset)

```
                    ENHANCED MODE - PROCEDURE: 030, WORD: 2
                    CALL VECTORING - ADMINISTER VECTORS

 1. Vector Number: 200

DISPLAY ONLY
   2.   See Vector Directory Number: -----
   3.              See Vector Number: ---
   4.                See Step Number: --
   5.          AUDIX Machine Number: -
   6. Message Center Machine Number: -


Connected to CC0 ON-LINE ♥  MAJOR   MINOR   RUN TAPE   BUSY OUT   IN USE   WAIT

enter command:
                    3 Form             5 Help   6 Field   7 Input   8 Cmds
```

ax rs (add execute, reset)

Vector Administration Using the CMS (Call Management System)

The "Configuration — Vector Specifications Screen" is used to program vectors with the CMS system. Using this screen, an entire vector can be viewed while programming a new vector or changing an existing vector. Function keys, user prompts, help messages, and limited integrity checks are provided by this screen to simplify vector programming. Also, this screen's user interface is consistent with the rest of the CMS interface.

The four basic programming operations can be carried out using the Vector Specifications screen. These operations include:

- Adding (programming) a new vector
- Changing an existing vector
  - — Adding a vector step
  - — Removing a vector step

    CAUTION: Whenever a vector step is ***either added to*** or ***removed from*** a vector, the "go to step" commands in the vector should be carefully examined, and if necessary, changed. It is quite likely that these "go to step" commands are now passing control to the wrong vector step.

  - — Changing a vector step
- Copying an existing vector
- Removing an existing vector.

The names of three of the nine vector commands have been changed so that the CMS software can recognize a unique entry with fewer key strokes by the user. Using a CMS terminal, the "delay" command has the name "wait." The "forced disconnect" command has the name "disconnect." The "forced busy" command has the name "busy."

A more detailed discussion of vector programming on the CMS system is provided in the ***Vectoring Administration*** guide for Release 2 of CMS (585-215-502).

To briefly illustrate the programming format provided by the CMS system, Vectors B and F are reprogrammed using the Vector Specifications screen.

```
Call Management System                          Switch_Name:Up        Time

                   CONFIGURATION - VECTOR SPECIFICATIONS
Vector :   1 (Service Desk)

STEP              ACTION & CONDITIONS
  1  queue to main split 9 priority low
  2  go to step 4 if calls queued in split 0 > 30
  3  stop
  4  announcement number 20
  5
  6
  7
  8
  9
 10
 11
 12
 13
 14
 15

COPY       DELETE    EDIT         EXIT    PRINT    HELP
                                          SCREEN   KEYS
```

Figure 34-3. Sample CMS Administration of Vector B

NOTE: Boldface type denotes keywords and parameters defined by the user; standard type indicates prompts supplied by the CMS system when a keyword or parameter is typed. Together, keywords and parameters define a vector step.

```
Call Management System                          Switch_Name:Up        Time

                   CONFIGURATION - VECTOR SPECIFICATIONS
Vector :   2 (Claims)

STEP              ACTION & CONDITIONS
  1  queue to main split 6 priority low
  2  go to step 5 if calls queued in split 0 > 16
  3  go to step 6 if calls queued in split 0 > 8
  4  go to step 7 if unconditional
  5  wait time 6 secs hearing ringback
  6  wait time 6 secs hearing ringback
  7  wait time 8 secs hearing ringback
  8  announcement number 16
  9  stop
 10
 11
 12
 13
 14
 15

COPY       DELETE    EDIT         EXIT    PRINT    HELP
                                          SCREEN   KEYS
```

Figure 34-4. Sample CMS Administration of Vector F

Notes:

Call Waiting

Description

The Call Waiting feature allows a call to a busy single-appearance voice terminal to wait on the called voice terminal rather than receiving busy tone. The called party hears a special tone (400 Hz for 0.1 seconds) indicating that a call is waiting. One burst of tone indicates a terminal-to-terminal call, and two bursts of tone indicate an attendant or outside call. The calling party beam call waiting ringback tone.

The Call Waiting feature is assigned through the extension number class of service. This feature is effective only for single appearance voice terminals. That is, Call Waiting does not work for an extension with multiple appearances. For busy multiappearance voice terminals, an incoming call routes to an idle appearance on the terminal or will be redirected by another feature such as Call Forwarding or Coverage. If all appearances of the called extension are busy, and none of the call redirection features are active, the calling party receives busy tone.

The Call Waiting feature is assigned to the terminating extension. That is, if a call is directed toward a busy extension that has Call Waiting assigned to its class of service, the Call Waiting feature applies. However, if the originating extension has Call Waiting assigned and the terminating extension does not, Call Waiting does not apply. This feature was at one time referred to as Call Waiting—Terminating.

Answer/Hold

A special DAC (Dial Access Code), the Call Waiting Answer / Hold access code, is available for use with the Call Waiting feature. The Call Waiting Answer / Hold access code is established using Procedure 350, Word 2 (encode 6). This DAC allows users to answer a waiting call without disconnecting from the currently active call (the current call is placed on hold). Use of the Call Waiting feature essentially doubles the call handling ability of a single-appearance voice terminal.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Answer a Call Waiting Call

Without ending the current 2-way connection:

1. Be sure a burst of tone is heard. [Calling party hears a call waiting ringback tone.]
2. Press **[RECALL]** ,

   or

   Momentarily press the switchhook. [The second party is put on soft hold, and recall dial tone is heard.]
3. Dial the Call Waiting answer / hold access code. [The second party is put on hard hold, and the called party is connected to the waiting party.]

By disconnecting from the current 2-way connection:

1. Be sure a burst of tone is heard. [Calling party hears a call waiting ringback tone.]
2. Go on-hook. [The second party is disconnected, and the switch rings the called party.]
3. Go off-hook. [The called party is connected to the waiting party.]

To Retrieve a Held Call After the Call Waiting Call Is Finished:

1. Press **[RECALL]** ,

   or

   Momentarily press the switchhook. [Recall dial tone]
2. Dial the Call Waiting Answer / Hold access code. [The second party is reconnected to the called party.]

Considerations

Call Waiting Tone

The Call Waiting tone is one or two 100-millisecond, 400-hertz beeps.

One Call to a Terminal

Only one call at a time can be held waiting on a single-appearance voice terminal. The switch returns busy tone to the calling party when another call is already waiting.

Multiappearance Voice Terminals

Calls are not held waiting on extensions with multiple appearances. This is because such calls are usually routed to an idle appearance. If every appearance is busy, and no redirection features are active, the caller receives busy tone.

Application of Call Waiting

The Call Waiting feature only applies to a single appearance voice terminal with an active call in a stable state. A stable state exists when the called station is active on a 2-way talking connection or the called station is connected to an outgoing trunk (once the trunk has been seized).

Companion Feature

The Priority Calling feature (formerly Call Waiting — Originating) is an ***independent*** companion feature to the Call Waiting feature. Priority Calling is assigned to the calling terminal on a class of service basis, and allows the calling party to use distinctive 3-burst ringing or three 400-hertz beeps to alert the called party.

Hard and Soft Processor Swaps

If a call is waiting on a busy single-appearance terminal when a hard processor swap occurs, the waiting call does not endure the hard swap.

The Call Waiting feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Automatic Call Distribution

Call Waiting is denied to an ACD agent while an observer (using agent override) is connected to the agent's call.

Automatic Callback

If a busy party has a call waiting and another party tries to call the busy party, the switch returns busy tone to the calling party. The calling party can now activate Automatic Callback toward the busy party. However, the callback sequence is delayed until there are no calls waiting.

If a calling party activates Automatic Callback toward a busy terminal and then becomes busy with another call, use of the Call Waiting feature by another party toward the calling party is still allowed.

If a party is waiting on a busy line, the busy party with the waiting call cannot place the active call on hold and then activate Automatic callback toward another line.

Bridged Call

Call Waiting is allowed for shared extensions, only under certain circumstances. Specifically, where one of the terminals sharing the extension is a straight line set and there is only one appearance of the share extension on the multiappearance set. In this case, when the straight line set is active on the shared extension, the switch allows an incoming call to wait.

When the multiappearance terminal is active on the shared extension, Call Waiting is denied and busy tone is returned to the calling party.

Busy Verification of Lines

If an attendant attempts to busy verify a terminal line which already has a call waiting busy verification proceeds normally. However, if the attendant attempts to busy verify a terminal line that is waiting for another line, the switch denies the busy verification attempt.

Call Coverage

When a principal has a single-appearance voice terminal with Call Waiting assigned, the call waits on the voice terminal if the principal has Coverage—Don't Answer active and the principal is busy. If any other type of coverage is active, the call goes to coverage normally.

Call Forwarding—Busy and Don't Answer

When Call Forwarding—Busy and Don't Answer is active at the called extension, the forwarding operation takes precedence. Call Waiting at the called extension has no effect. If Call Waiting is active at the forwarded to extension there are several possible cases shown in Table 35-A.

Call Forwarding—Don't Answer

When Call Forwarding—Don't Answer is active at the called extension the forwarding operation takes precedence and Call Waiting has no effect. If Call Waiting is active at the forwarded to extension there are several possible cases shown in Table 35-A.

Call Forwarding—Follow Me

When Call Forwarding—Follow Me is active at the called extension, the forwarding operation takes precedence and Call Waiting has no effect. If Call Waiting is active at the forwarded to extension there are several possible cases shown in Table 35-A.

TABLE 35-A. Call Waiting—Call Forwarding Interactions

Forwarded-To Extension With Call Waiting	Originally Called Extension with Call Forwarding Active	
	Follow Me or Busy	Doesn't Answer
Is a Single Appearance Voice Terminal and is Busy	Call Forwards to and Waits at the Forwarded-To Extension	Call Forwards to and Waits at the Forwarded-To Extension
Is a Single Appearance Voice Terminal and is Idle	Call Forwards to and Rings at the Forwarded-To Extension	Call Forwards to and Rings at the Forwarded-To Extension
Is a Straight Line Set* and is Busy	Call Forwards to and Waits at the Forwarded-To Extension	Call Forwards to and Waits at the Forwarded-To Extension
Is a Straight Line Set* and is Idle	Call Forwards to and Rings at the Forwarded-To Extension	Call Forwards to and Rings at the Forwarded-To Extension
Is a Straight Line Set* and the Single Multiappearance Image is Busy	Busy Tone is Returned	Busy Tone is Returned
Is a Straight Line Set with Multiple Appearances† of the Shared Extension	Call Hunts for and Rings at the First Idle Appearance	Call Hunts for and Rings at the First Idle Appearance
Is a Straight Line Set with Multiple Appearances of the Shared Extension	If there are no Idle Appearances, Busy Tone is Returned	If there are no Idle Appearances, Call Does Not Forward But Continues to Ring at the Called Extension

* These cases assume a single appearance of the extension with two images, one on the straight line set and one on the multiappearance set.

† Note that calls do not wait on extensions with multiple appearances. See Considerations.

Call Park

Call Waiting cannot be used toward a line that is parked.

Call Vectoring

If a single-appearance terminal is the destination of a "route to" command, an incoming call will not wait on the terminal. Instead, if the "route to" command is the last vector step, vector processing will attempt to redirect the call to an idle forwarding destination or coverage point. This final vector step is retried at 2-second internals. If the "route to" command is not the last vector step, vector processing will continue with the next sequential step.

Code Calling Access—Universal

A call is not allowed to wait on a line that has accessed code calling.

Conference—Attendant Five Party

Call Waiting is denied when the called terminal is involved in an attendant established conference.

Conference-Attendant Six Party

Call Waiting is denied when the called terminal is involved in an attendant established conference call.

Conference—Three Party

Call Waiting is denied when the called terminal is involved in a 3-party conference.

Data Protection

The switch denies Call Waiting when the Data Protection feature is active on a call.

DDC (Direct Department Calling)

When Call Waiting is assigned to an individual extension in a DDC group, calls to the terminal are allowed to wait if the called terminal line is busy. The called group member can be connected to the waiting call by going on-hook, whereby the terminal is alerted and connected to the call upon answer. These calls have preference over DDC calls in queue waiting to be answered. The controlling extension number of a DDC group should not have this feature assigned.

EUCD (Enhanced Uniform Call Distribution)

While an observer (using agent override) is connected to an EUCD agent's call, the Call Waiting feature is denied for use by the agent.

Hold

A voice terminal with a call on hold and a call in waiting goes on-hook. The waiting call is connected first. Using the Call Hold access code is the recommended method for returning to a held call.

The switch denies Call Waiting toward a voice terminal that has been placed on hold.

Call Waiting is also denied if the calling voice terminal has a call on soft hold, however, a voice terminal with a call on hard hold can wait.

Hunting

Hunting takes precedence over Call Waiting. That is, if the called terminal is a member of a hunt group, the call will hunt before it will wait. However, if every member of the hunt group is busy, the call will wait on the originally called terminal.

IPA (Interpartition Access)

A partitioned System 85 or Generic 2 allows Call Waiting and provides 1-burst tone for terminal-to-terminal calls inside a partition group. The switch also allows Call Waiting and provides 1-burst tone for terminal-to-terminal calls between an extension partition and Extension Partition 0.

Otherwise, terminal-to-terminal calls are not allowed to cross partition-group boundaries. In order to call a voice terminal in any other partition group, a voice terminal user must dial the appropriate 7-digit number which routes the call over a CO (Central Office) trunk. When this is done, the switch allows Call Waiting and provides 2-burst tone for the called voice terminal.

The switch allows Call Waiting for incoming calls from the public network, and provides 2-burst tone for these calls.

Line Lockout

A call is not allowed to wait on a voice terminal that has been locked out. The switch returns busy tone to the calling party.

Loudspeaker Paging Access

A call is not allowed to wait on a line which has accessed Loudspeaker Paging.

Malicious Call Trace

Calls are not allowed to wait on a line involved in a Malicious Call Trace. Instead, the calling party receives busy tone.

Override

Override is not allowed toward a line which is waiting, but is allowed toward the 2-party call that has a call waiting.

Queuing

The callback sequence associated with Queuing at the local switch is delayed until there are no calls waiting on the terminal. If a callback attempt is made from a tandem switch to a subtending switch, the call appears as an ordinary incoming tie trunk call to the subtending switch. If the called party has call waiting assigned and is busy on another call, the callback attempt notifies the called party. If the callback attempt is answered, the call can be completed while the third party waits on hold. If the called party ignores the tone, the attempt is treated as a "don't answer" case.

Recorded Telephone Dictation Access

A call is not allowed to wait on a terminal that has activated Recorded Telephone Dictation Access.

Restriction—Attendant Control of Voice Terminals

If an Attendant Control of Voice Terminals restriction prevents a call from terminating at a terminal, Call Waiting is also prevented to that terminal.

Restriction—Voice Terminal Restrictions

If a Voice Terminal Restriction prevents a call from terminating at a terminal, Call Waiting is also prevented to that terminal.

Trunk Verification—Attendant and Voice Terminal

If the trunk is being held or answered by a terminal using the answer-hold code of the Call Waiting feature, trunk verification is denied.

Tenant Services

A partitioned System 85 or Generic 2 allows Call Waiting and provides 1-burst tone for terminal-to-terminal calls inside an extension partition. The switch also allows Call Waiting and provides 1-burst tone for terminal-to-terminal calls between an extension partition and Extension Partition 0.

Otherwise, terminal-to-terminal calls are not allowed to cross partition boundaries. In order to call a voice terminal in any other partition, a voice terminal user must dial the appropriate 7-digit number which routes the call over a CO (Central Office) trunk. When this is done, the switch allows Call Waiting and provides 2-burst tone for the called voice terminal. The switch allows Call Waiting for incoming calls from the public network, and provides 2-burst tone for these calls.

System 85 and Generic 2 allow attendant calls to wait on local voice terminals, and provides 2-burst tone for the called voice terminal. When allowed, 2-burst tone is provided for attendant calls directed to a voice terminal using either the extension number or DXS (Direct Extension Selection). Two-burst tone is also provided for attendant calls directed to a local voice terminal using the appropriate 7-digit public-network number (when partitioning requires this method of dialing).

Unattended Console Service—Preselected Call Routing

Call Waiting is not allowed when a call is already waiting on the busy preselected voice terminal, or when the user has placed another call on hold. The next call is handled by the Call Answer From Any Voice Terminal feature, if provided. The incoming queue holds all other calls. When going on-hook with a call waiting and with one or more calls in queue, the user will *probably* be connected to the next call in the queue instead of the waiting call. This occurs because the queue scan runs every 100 milliseconds, whereas the Call Waiting scan runs every 2 seconds. To ensure a connection with the waiting call, the terminal user should press the RECALL button and dial the answer-hold access code. The terminal user is then connected to the waiting party.

UCD (Uniform Call Distribution)

When Call Waiting is assigned to an individual (single appearance) extension in a UCD group, calls to that terminal are allowed to wait if the called terminal line is busy. The called terminal is connected to the waiting call by going on-hook, whereby the terminal is alerted and connected to the call when answered. These calls have preference over UCD calls in queue. The controlling extension number of a UCD group should not have the Call Waiting feature assigned.

Hardware Requirements

None.

Feature Administration

Assignment of the Call Waiting feature is on a per-system and per–terminal class of service basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer Call Waiting using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature

On Generic 2 switches, this feature is administered using the DEFINITY Manager II

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — CALL WAITING			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the extension number class of service to an extension number.	Yes
010	1	Assigns Call Waiting to an extension number class of service.	Yes
275	1	Assigns Call Waiting to the system class of service.	Yes
350	1	Assigns the first digit of the dial access codes (if required).	No
350	2	Assigns the feature dial access codes. The applicable encode is: 6 Call Waiting - Answer Hold.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — CALL WAITING	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns Call Waiting to an extension number class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Call Work Codes

Description

A CWC (Call Work Code) is a customer-defined code such as an account number, a call activity code, a credit card number, or a social security number. An ACD agent using a DCP voice terminal can enter a CWC during or after an ACD call. The agent must be measured by and logged into CMS (Call Management System). The DEFINITY Generic 2 collects the CWC and sends it to a CMS adjunct for storage; the switch does **not** store the CWC.

Feature History and Development

This feature was first available with DEFINITY Generic 2.2.

Required Feature

The Automatic Call Distribution feature must be assigned to the DEFINITY Generic 2. Only ACD agents who are measured by and logged into CMS can use the CWC feature.

Activating the CWC Feature

Pressing the CWC feature button on a DCP voice terminal activates the CWC feature. Activation of the CWC feature is allowed if:

- The CWC feature is assigned to the system (Procedure 276, Word 1).
- The ACD agent is measured by and logged into CMS.
- The agent is off-hook on an ACD call or in the after-call-work state. (When using Manual-In mode, an agent is automatically placed in the after-call-work state after disconnecting from an ACD call.)
- The CWC feature is not already active for the extension. (If the CWC feature has already been activated by this voice terminal user, pressing the CWC button again deactivates the feature. If the CWC feature is active and another voice terminal user with a bridged appearance of the extension attempts to activate the feature, the switch ignores the feature activation.)

If activation of the CWC feature is successful, the status lamp associated with the CWC button lights steadily, signaling the agent that a CWC can be entered. If the voice terminal has display capabilities, a "C:" is displayed to prompt the user to enter a CWC. The flash rate of the status lamp is set to broken flutter for 2 seconds if activation of the CWC feature is denied.

Entering a CWC

After activating the CWC feature, the agent uses the touch-tone pad to enter as many as 16 digits (0 through 9). If the voice terminal has display capabilites, the digits are displayed as they are entered. An agent can enter a CWC while handling an ACD call without interrupting the call.

If the agent enters the wrong CWC or makes a mistake while entering a CWC, pressing the **[*]** key on the touch-tone pad clears the entry (and the display).

To accommodate CWCs of different lengths, an agent must press the **[#]** key on the touch-tone pad (end-of-dialing character) after entering a CWC.

While entering a CWC, pressing a call-appearance or feature button or going on-hook before pressing the **[#]** key deactivates the CWC feature and cancels the CWC entry. Furthermore, the switch treats the button that was pressed to deactivate the CWC feature as if the CWC feature were not active. For example, if a call-appearance button is pressed while entering a CWC, the CWC feature is deactivated, the CWC entry is canceled, and if the agent is off-hook on an ACD call, the call is disconnected.

> **NOTE:** Manual-In mode is recommended for agents who use the CWC feature. Using Manual-In mode is especially important for agents who receive short-duration ACD calls. If an agent is using Auto-In mode to receive ACD calls and the caller disconnects while the agent is entering a CWC, the entry will be canceled.

If the CWC feature is deactivated while entering a CWC, the status lamp associated with the CWC button is turned off and the display is cleared (if the terminal is equipped with a display). If the CWC feature is deactivated accidently, the agent should press the CWC button and enter the CWC again (or a different CWC).

> **NOTE:** Although unlikely, it is possible to temporarily lock up a voice terminal by entering CWC digits extraordinarily fast. If this happens, the lamp associated with the CWC button remains lit after the **[#]** key is entered. The agent should wait until the voice terminal returns to normal operation (usually about 15 seconds), press the CWC button, and enter the CWC again.

Call Management System

After the CWC and the **[#]** key are entered, the switch collects the CWC and sends it to the CMS adjunct. If the CWC cannot be sent to the CMS adjunct, the switch sets the flash rate of the status lamp associated with the CWC button to broken flutter. If the agent's voice terminal has a display, the CWC remains displayed so that it can be recorded manually. The CWC remains displayed and the lamp continues to flash until the agent deactivates the CWC feature, for example by pressing a feature button or by going on hook.

CMS stores all CWCs an agent enters for an ACD call.

The Forced-Entry Option

An ACD split that is measured by CMS can be assigned the forced-entry option (Procedure 026, Word 2). Agents in forced-entry splits should use the Manual-In mode to receive ACD calls and must enter a CWC or press a stroke-count button before receiving another ACD call. A CWC or stroke count can be entered during or after an ACD call. Refer to the ACD feature description for information about the stroke-count button.

> **NOTE:** The forced-entry requirement can also be satisfied by pressing the CWC button followed by the **[#]** key (without entering a CWC), by pressing the AUTO-IN button, by pressing the AUX-WORK button, by logging out, or by unstaffing.

When using the Manual-In mode, agents press the MANUAL-IN button to receive each ACD call. After disconnecting from a Manual-In call, an agent is unavailable to receive another ACD call until the MANUAL-IN button is pressed again. This unavailable period is called the after-call-work state and it allows agents to finish call-dated paper work or do other follow-up work such as entering a CWC.

While an agent is in the after-call-work state, the lamp associated with the MANUAL-IN button flashes. For agents in forced-entry splits, the lamp continues to flash until the forced-entry requirement is satisfied and the MANUAL-IN button is pressed.

The multiple-call-handling and forced-entry options cannot both be assigned to the same ACD split. Likewise, the automatic-available and forced-entry options cannot both be assigned to the same split.

User Operation

To Enter a Call Work Code

To enter a CWC, an ACD agent must be measured by and logged into CMS and must be off-hook on an ACD call or off-hook in the after-call-work state.

1. If not active on an ACD call, go off-hook.
2. Press the CWC button.
3. Enter the CWC (0 to 16 digits).
4. Press the **[#]** key on the touch-tone pad (indicates end of dialing). [Status lamp associated with the CWC button is turned off]

To Clear a Call Work Code During Entry

If an agent makes a mistake while entering a CWC or enters the wrong CWC, the entry can be cleared if the agent finds the mistake before pressing the **[#]** key. To clear a CWC entry (and the display), press the **[*]** key on the touch-tone pad. Then enter the correct CWC.

To Change a Call Work Code After It is Sent to the CMS Adjunct

After entering a CWC and pressing the **[#]** key, an agent can change (overwrite) the CWC by entering a different CWC. (More than one CWC can be entered for an ACD call, but CMS only stores the last-entered CWC.)

Considerations

A CWC button can only be assigned to a DCP (Digital Communications Protocol) voice terminal. That is, a CWC button cannot be assigned to an analog, a hybrid, or a BRI set. Furthermore, the switch does not recognize digits sent from earlier versions of 7403D, 7404D, and 7405D voice terminals.

Only one CWC button can be assigned per voice terminal.

Interactions With Other Features

Abbreviated Dialing

The Abbreviated Dialing feature cannot be used to enter a CWC.

While entering a CWC, pressing an Abbreviated Dialing button deactivates the CWC feature and cancels the CWC entry. However, unlike other feature buttons, pressing an Abbreviated Dialing button does not initiate that function. That is, the switch does not select an idle call appearance and dial the stored number if an Abbreviated Dialing button is pressed while entering a CWC.

Automatic Call Distribution

If agents use the Auto-In mode to receive ACD calls, a CWC entry will be canceled if the caller disconnects before the agent completes the entry. Consequently, Manual-In mode is recommended for agents who use the CWC feature and receive short-duration calls. When using Manual-In mode, an agent can enter a CWC during or after an ACD call. Furthermore, a CWC entry is not canceled if the caller disconnects before the agent completes the entry.

The multiple-call-handling and forced-entry options cannot both be assigned to the same ACD split. Likewise, the automatic-available and forced-entry options cannot both be assigned to the same split.

ACD splits that are administered as automatic available usually distribute calls to ports on an adjunct processor such as a voice response unit. For this type of split, a CWC can only be entered during an ACD call, because the ports on the adjunct processor are not put into the after-call-work state.

An agent can enter a CWC while being monitored by Service Observing, however, the observer cannot activate the CWC feature or enter a CWC for the agent that is being monitored.

While entering a CWC, pressing a call-appearance or ACD feature button or going on hook before pressing the **[#]** key deactivates the CWC feature and cancels the CWC entry.

Bridged Call

The CWC feature can be activated from a bridged appearance, however, if the CWC feature is already active for the extension, another attempt to activate the feature is ignored.

Conference—Three Party

Pressing the Conference button while entering a CWC deactivates the CWC feature and cancels the CWC entry. However, an agent can enter a CWC before establishing a conference.

If two agents, both in splits assigned the forced-entry option, are involved in a conference call, only the agent who established the conference has to satisfy the forced-entry requirement. The agent who is added to the conference does not have to satisfy the forced-entry requirement because the call is not considered an ACD call for that agent.

Display Voice Terminal

Pressing a display-mode button while entering a CWC deactivates the CWC feature, cancels the CWC entry, and initiates the requested display mode.

Hold

Pressing the Hold button while entering a CWC deactivates the CWC feature and cancels the CWC entry.

Transfer

Pressing the Transfer button while entering a CWC deactivates the CWC feature and cancels the CWC entry. However, an agent can enter a CWC before transferring a call.

If two agents, both in splits assigned the forced-entry option, are involved in a call transfer, only the agent who transferred the call has to satisfy the forced-entry requirement. The agent to whom the call was transferred does not have to satisfy the forced-entry requirement because the call is not considered an ACD call for that agent.

Hardware Requirements

An adjunct processor running Release 3 CMS software and a DCIU link connecting the switch to the adjunct processor are required to stem call work codes.

Feature Administration

Assignment of the CWC feature is on a per-system basis.

On DEFINITY Generic 2, this feature is administered using DEFINITY Manager II.

This feature can also be administered using CSM (Centralized System Management).

Only ACD agents who are measured by and logged into CMS can use the CWC feature. Consequently, ACD, CMS, and a DCIU (Data Communications Interface Unit) link connecting the switch to the CMS adjunct must be administered. Refer to the ACD feature description for information about administering ACD and to Appendix H for information about administering a DCIU. Furthermore, CMS (Procedure 275, Word 4) and DCIU (Procedure 275, Word 1) must be enabled in the system class of service.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
CALL WORK CODES

PROCEDURE	WORD	PURPOSE	SMT
026	2	Assigns the forced-entry option to an ACD split.	Yes
054	1	Assigns the CWC button to a DCP voice terminal.	Yes
276	1	Assigns the Call Work Codes feature to the feature group class of service.	No

Centralized Attendant Service

Description

This feature enables private-network attendants (and other call-answering positions) to be centrally located at one switch in the network. The attended system is called a CAS main and each unattended system is called a CAS branch.

Consolidating answering positions at one location provides the following advantages:

- Traffic peaks at branch locations are more evenly distributed.
- Supervising and training one group could be easier.
- Working together can improve morale and reduce space requirements.

At branch locations, calls requiring attendant services route by way of RLTs (Release Link Trunks) to the main location (see Figure 37-l). After speaking with the calling party, the attendant (or other call-answering position) at the main transfers the call back to the branch location that originally received the call.

When an RLT is used in a CAS arrangement, the RLT is released when the call is transferred back to the original branch location. In this way, the RLT can be used to service another call at the CAS main while the call that used it previously is still active (at the branch). If a conventional tie trunk where use is this way, the tie trunk that carried the call to the main, plus another tie trunk used to transfer the call back to the branch, would be tied up for the duration of the call.

Feature History and Development

This feature was first available on System 85 in Release 1.

Beginning with R2 V3, Extension Number Steering must be used to route DID (Direct Inward Dialing) calls to the CAS queue.

Beginning with Issue 3.0 of DEFINITY Communications System Generic 2.1, an RLT can terminate to an ACD split, VDN (Vector Directory Number), or attendant console. Before this enhancement, an RLT could only terminate to an attendant console.

For RLTs that terminate to ACD splits or VDNs, any voice terminal with a Conference or Transfer button can be used to transfer incoming RLT calls from the CAS main to a branch location.

> **NOTE:** When RLTs terminate to ACD splits or VDNs, some attendant functions can be replaced by ACD agents or other voice terminal users. However, if attendants at the main perform CAS and non-CAS functions, it may not be possible to eliminate all attendant positions. Refer to the Considerations section in this feature description for more information.

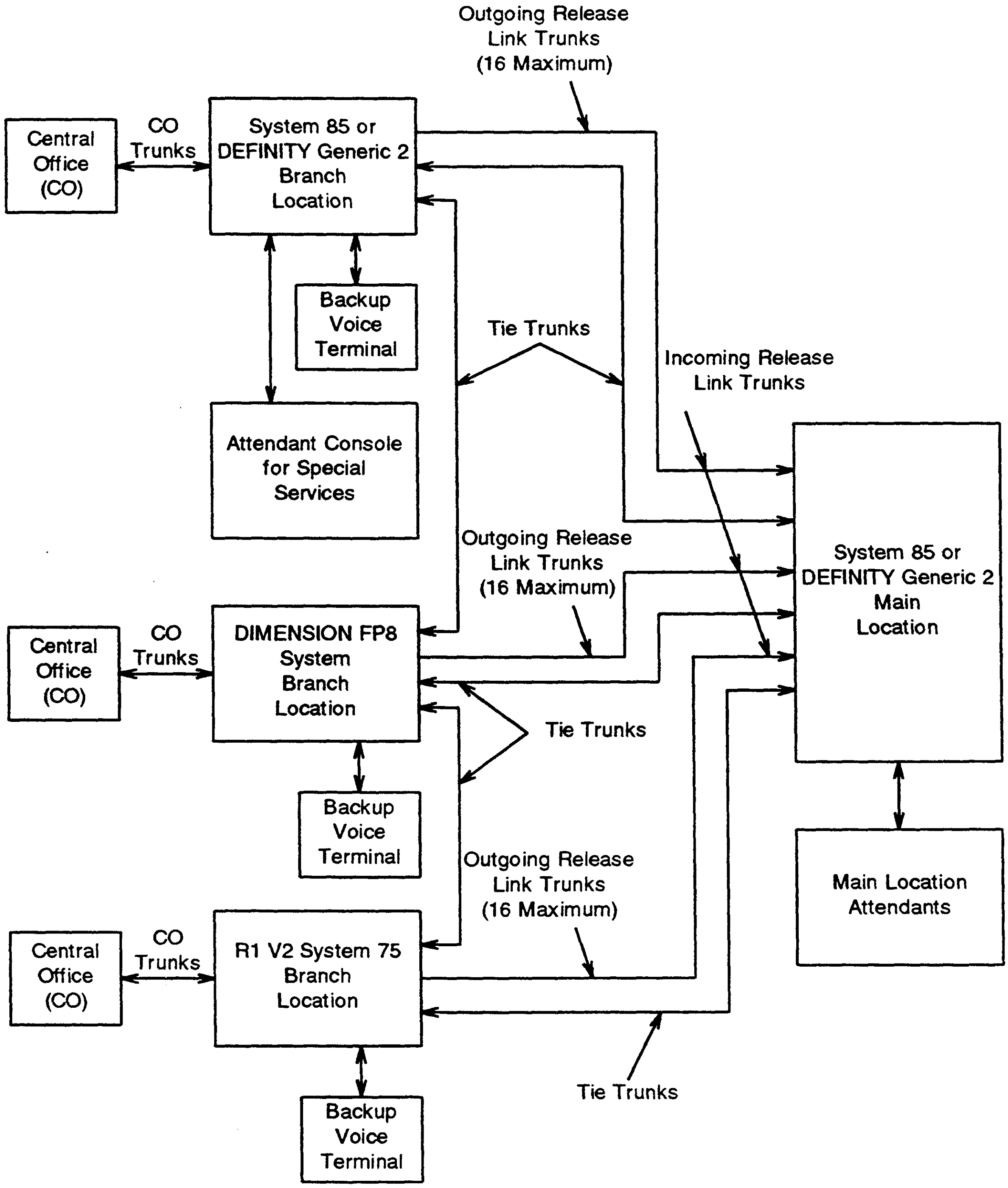


Figure 37-1. Centralized Attendant Service—Block Diagram

Call Identification Tones

An attendant or other answering position may receive call identification tones before connecting to a call. (See the Considerations section for information about Call Identification Tones, Zip Tones, and Recorded Announcements.) A distinctive tone is provided for each of the following types of calls.

TABLE 37-A. Attendant Information Tones From a Branch Location

Call Type	Information Tone
Incoming Trunk Call	In sequence: 480-hertz (for 100 msec), 440 Hertz (for 100 msec), 480-hertz (for 100 msec).
Dial Access to Attendant (Dial "0") call	In sequence: 440-hertz (100 msec ON, 100 msec OFF, 100 msec ON).
Forwarded Call on Don't Answer, ACD (Automatic Call Distribution), or EUCD (Enhanced Uniform Call Distribution) Overflow	Ringback for 300 msec followed by connection to normal ringing cycle at any point in the cycle.
Recall on Code Calling Access	Ringback for 300 msec followed by connection to normal ringing cycle at any point in the cycle.
Recall on call Waiting	440-hertz (for 100 msec).
Redirected Call From the Remote Hold Feature	A series of four to six cycles: 440-hertz (for 50 msec) and silence (for 50 msec).

Attendant Console and Voice Terminal Displays

For an RLT that terminates to an attendant console, the attendant console can be administered to display the source of incoming RLT calls (the CAS branch identification is administered in Procedure 204, Word 1). A similar capability can be administered for RLTs that terminate to ACD splits or VDNs. Using Procedure 012, Words 1 and 2, a name-data-base entry can be assigned to an incoming RLT trunk group (for ACD splits) or a VDN. When this is done, answering positions equipped with voice terminals with display capabilities can receive information about the source (branch identification) and type (RLT or non-RLT) of incoming call. This information is important because, for ACD agents and other voice terminal users, the operation of the Conference and Transfer features is different for incoming RLT calls. See the User Operations section of this feature description for more information about transferring RLT calls.

Branch Location Attendants

System 85 or Generic 2 branch locations can also have local attendant positions. Local attendants can provide services that centralized attendants cannot provide. The following are examples of some such services:

- Alternate Facilites Restriction Levels
- Attendant Control of Trunk Group Access*
- Attendant Interposition Calling and Transfer
- Busy Verification of Lines*
- Conference—Attendant Six Party
- Malicious Call Trace
- Restriction—Attendant Control of Voice Terminals
- Serial Calls
- Trunk Verification by Attendant*
- Unattended Console Service—Preselected Call Routing.

Voice terminal users at a branch location can reach a local attendant by dialing the attendant dial access code. The centralized attendants handle all other calls requiring attendant help.

Branch Modes

In a CAS arrangement, each System 85 or Generic 2 branch location can operate in one of three modes:

- Regular (CAS active)
- Backup Extension Mode

  In this mode, extensions (at the branch) provide service for the RLTs.
- CAAVT (Call Answer From Any Voice Terminal

  In this mode, calls that would otherwise route to a centralized attendant are identified by a bell. Any unrestricted voice terminal user (at the branch) can answer the call by dialing the CAAVT access code.

These modes are invoked at the branch location via dial access code. When the regular CAS mode is not active, the branch location can use either the backup voice terminal mode or the CAAVT mode. In either of the backup modes, calls are not referred to the main over RLTs.

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* With the DCS (Distributed Communications System) feature, these services can be provided by CAS attendants.

System Status Monitoring

If provided at the main or branch locations, SSI (System Status Indicator) lamp panels monitor the busy/idle status of RLTs and the following system status information:

- CONTROL — Indicates that the system is in Regular (CAS active) mode or either backup extension mode or CAAVT mode.
- OVERLOAD — Indicates that the queue threshold has been reached or exceeded.
- MAJOR — Indicates that the system has a major alarm.
- MINOR — Indicates that the system has a minor alarm.

User Operations

The following are the user operating procedures for this feature.

To Transfer (Extend) an Incoming RLT Call to a Branch Location:

An Attendant Console user should:

1. Press the appropriate loop button.

   or

   Press **[ANSWER]** . [PA lamp goes out] (Calling party is connected.)

2. After receiving the calling party's instructions, press **[START]** . [Dial tone] (Calling party is placed on hold.)

3. Dial the requested number.

4. If ringback (or Call Waiting ringback) is heard, press **[RLT RELEASE]** . (The RLT is released.)

   NOTE: If the called party does not answer before the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main.

   or

   If busy tone is heard, press **[CANC]** (Attendant is reconnected to the calling party.) and inform the calling party that the extension is busy.

   If the calling party does not want to wait for the called party to answer, press **[RLT RELEASE]** . (The RLT is released.)

   or

   If the calling party wants to wait for the called party to answer, press **[START]** . [Dial Tone] (Calling party is placed on hold at the main.)

   Dial the Remote Hold dial access code [Confirmation tone] (Calling party is placed on hold at the branch.)

Press **[RLT RELEASE]** . (The RLT is released.)

NOTE: If the calling party is a station at a branch location, Remote Hold will be denied.

NOTE: When the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main. The attendant can attempt to complete the call again.

An ACD agent (or other voice terminal user) should:

1. After receiving the calling party's instructions, press **[TRANSFER]** or **[CONFERENCE]** .* [Dial tone] (Calling party is placed on hold.)
2. Dial the requested number.
3. If ringback (or Call Waiting ringback) is heard, press **[DISCONNECT]** , **[RELEASE]** ,

   or

   Go on-hook. (The RLT is released.)

   NOTE: If the called party does not answer before the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main.

   or

   If busy tone is heard, press **[TRANSFER]** or **[CONFERENCE]** (Voice terminal user is reconnected to the calling party.) and inform the calling party that the extension is busy.

   If the calling party does not want to wait for the called party to answer, press **[DISCONNECT]** , **[REALEASE]** ,

   or

   Go on-hook. (The RLT is released.)

   or

   If the calling party wants to wait for the called party to answer, press **[TRANSFER]** or **[CONFERENCE]** . [Dial tone] (Calling party is placed on hold at the main.)

   Dial the Remote Hold dial access code. [Confirmation tone] (Calling party is placed on hold at the branch.)

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* For incoming RLT calls, the Conference button works the same as the Transfer button That is, pressing the Conference button transfers an incoming RLT call back to the originally called branch location. Pressing the Conference button does not set up a 3-party conference.

Press **[DISCONNECT]** , **[RELEASE]** ,

or

Go on-hook. (The RLT is released.)

NOTE: If the calling party is a station at a branch location, Remote Hold will be denied.

NOTE: When the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main. The voice terminal user can attempt to complete the call again.

To Transfer (Extend) an Incoming RLT Call to an ACD or EUCD Split at a Branch Location:

An Attendant Console user should:

1. Press the appropriate loop button.

   or

   Press **[ANSWER]** . [PA lamp goes out] (Calling party is connected.)
2. After receiving the calling party's instructions, press **[START]** . [Dial tone] (Calling party is placed on hold.)
3. Dial the number associated with the desired split at the branch. [Ringback tone]
4. Press the **[RLT RELEASE]** button ***within 4 seconds.*** [The call enters the split's queue.]

   NOTE: If the CAS attendant does not release within 4 seconds, the call is treated as a direct attendant call. Attendant calls to an ACD or EUCD split do not enter the split's queue. Instead, System 85 or Generic 2 scans the split for an available agent, and if found, completes the call to that agent. If an available agent is not found, there are two possible switch responses. First, if Attendant Call Waiting is assigned at the branch location, the call waits on the split supervisor's voice terminal. Second, without Attendant Call Waiting, the switch returns busy tone to the CAS attendant.

An ACD agent (or other voice terminal user) should:

1. After receiving the calling party's instructions, press **[TRANSFER]** or **[CONFERENCE]** . * [Dial tone] (Calling party is placed on hold.)

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* For incoming RLT calls, the Conference button works the same as the Transfer button. That is, pressing the Conference button transfers an incoming RLT call back to the originally called branch location. Pressing the Conference button does not set up a 3-party conference.

2. Dial an associated extension number for the desired split at the branch. [Ringback tone]

3. When ringback is heard, press **[DISCONNECT]** *within 4 seconds.* [The call enters the split's queue.]

    NOTE: If an ACD agent (or other answering position) at the CAS main does not disconnect within 4 seconds, the call is treated as a direct attendant call (because the call came in on an RLT). An attendant call to an ACD or EUCD split does not enter the split's queue. Instead, System 85 or Generic 2 scans the split for an available agent, and if found, completes the call to that agent. If an available agent is not found, there are two possible switch responses. First, if Attendant Call Waiting is assigned at the branch location, the call waits on the split supervisor's voice terminal. Second, without Attendant Call Waiting, the switch returns busy tone to the agent at the main.

To Call a CAS Attendant from a Branch Location

A voice terminal user should:

1. Go off-hook. [Dial Tone]
2. Dial the Call to CAS Attendant access code.* [Ringback tone]

To Call a CAS Attendant from the Main Location

A voice terminal user should:

1. Go off-hook. [Dial tone]
2. Dial the attendant access code.* [Ringback tone]

To Test the System Status Indicator Lamps:

An attendant-console user (at the CAS main) or a backup voice terminal user (at a CAS branch) should:

1. Press an idle loop button.
2. Press **[START]** . [Dial tone]
3. Dial the lamp test access code. [Status lamps light.]
4. Scan the status lamps to verify that all are lighted.

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* The CAS attendant access code and the attendant access code are two different dial access codes. The CAS attendant access code is used only from a branch location to reach an attendant at the CAS main. The attendant access code can be used at either a CAS branch or a CAS main. When used at a CAS branch the attendant access code calls a local attendant if one is assigned. When used at the CAS main, the attendant access code also calls a local attendant, who in this case is also a CAS attendant.

To Return the System Status Indicator Lamps to Normal Operation:

An attendant-console user (at the CAS main) or a backup voice terminal user (at a CAS branch) should:

1. Be sure you're in the test mode.
2. Press **[CANC]** .
3. Press **[START]** . [Dial tone]
4. Dial the lamp test access code. [Status lamps go out.]
5. Press **[RELEASE]** . [PA lamp lights.]

To Activate Control by a Backup Voice Terminal

The backup voice terminal user should:

1. Go off-hook. [Dial tone]
2. Dial the Activate Control of CAS Backup Terminal access code. [Confirmation tone]
3. Go on-hook.

To Return CAS Control From the Branch to the Main Location

The backup voice terminal user should:

1. Go off-hook. [Dial tone]
2. Dial the Activate CAS Control access code. [Confirmation tone]
3. Go on-hook.

Considerations

Main Capacities

As many as 40 attendant consoles and 40 incoming RLT trunk groups can be assigned at the CAS main. Each RLT trunk group can contain up to 16 trunks. If an SSIs (System Status Indication) are used to monitor RLTs, as many as 110 RLTs can be monitored by SSIs. If RLTs are not monitored by SSIs, the number of RLTs that can be assigned at the main is limited to 16 times the number of branch locations or a maximum of 640 (16 RLTs X 40 RLT trunk groups).

Branch Capacities

Each branch location can be assigned one outgoing RLT trunk group with up to 16 RLTs.

Department Numbering Plans

If the CAS arrangement is not part of a DCS, to simplify call-handling procedures, identical departments at every branch location should be assigned the same extension number. In this way, one extension number can be used to transfer a call to a specific department at any location. When several departments at a branch location share the same voice terminal, several extension numbers can be assigned to the terminal.

RLT (Release Link Trunk) Functions

RLTs function as l-way outgoing circuits at CAS branch locations and 1-way incoming circuits at the CAS main. At a branch location, incoming calls that require attendant services route by way of RLTs to the main location. A CAS attendant (or other call-answering position) transfers the call back to the branch location that originally received the call and then releases the RLT. Calls originated by a CAS attendant (or voice terminal user at the main) cannot be routed to a branch location over an RLT. Therefore, to allow normal voice traffic, tie trunks and RLTs should be provided between CAS locations.

Type 36 (Procedure 100) Tie Trunks

Type 36 (2-way dial repeating both ways) tie trunks, connecting each pair of switches in the CAS arrangement, provide calling mobility for centralized attendants and other answering positions at the main location. Without these tie trunks, an answering position at the main can ***only*** transfer a CAS call back to the originally called branch. When tie trunks are provided, an answering position can transfer calls ***through*** the originally called branch to ***any*** desired switch in the CAS arrangement.

Tie Trunk Numbering

To simplify call-handling, uniform numbering of the type 36 tie trunks throughout the CAS arrangement is recommended. That is, every tie trunk that terminates to a specific switch in the CAS arrangement is assigned the same dial access code at every originating switch. In this way, one access code is used to transfer a call to a specific switch through any switch in the arrangement.

Backup Extensions (Voice Terminals)

At a branch location (backup extensions are not used at a main), each RLT can be assigned an individual backup extension (optional), or one extension can serve as the backup for all RLTs at a particular branch. The number of backup extensions cannot exceed the number of RLTs (maximum 16). Since the backup mode provides call identification tones, backup voice terminals should be equipped with turnkey operation, which allows the user to place the headset or handset near the ear before going off-hook.

For RLT calls processed on the console switch loops, the BUSY, RING, and ANS lamps are inactive.

ETN (Electronic Tandem Networking)

An ETN can overlay a CAS arrangement. If a switch in an ETN is also translated as a branch location in a CAS arrangement, both features will work properly. (The tie trunks in the network will not lock up.)

Routing DID Calls From a Branch to the CAS Queue

Procedure 115 does not allow DID trunk groups at a branch to terminate to the CAS queue at the main. Instead, Extension Number Steering provides digit-oriented routing to the CAS queue for these DID calls. Extension Number Steering is assigned in Procedure 354, Word 2. To enable DID routing from a branch to the CAS queue, assign the desired extension number to the Call to CAS Attendant dial access code (encode 49).

Hard and Soft Processor Swaps

Stable CAS calls that have already been transferred to a branch location will endure a hard processor swap at either the main or the branch location.

When a hard swap occurs at either the main or a branch location, a CAS call that has been answered (but not yet transferred to a branch location) cannot be transferred to the branch. The calling party should hang up and retry the call in 40 to 60 seconds.

The attendant queue at the main location is stored in a status portion of switch memory. Therefore, if a hard processor swap occurs at the main location, the attendant queue is cleared.

The CAS queue at each branch location is stored in a status portion of switch memory. Therefore, when a hard processor swap occurs at a branch location, the CAS queue at that location is cleared.

ACD queues at the CAS main or branch locations are not cleared if a hard processor swap occurs. However, agents must log in before calls are distributed.

The CAS feature operates normally during a soft processor swap at either the main or a branch location.

Considerations For Issue 3.0 of Generic 2.1 or Later

The following considerations apply to Generic 2.1, Issue 3.0 or later. Beginning with Generic 2.1, Issue 3.0, an RLT can terminate to an ACD split, a VDN (Vector Directory Number), or an attendant console.

Reduced Equipment Cost

One advantage of Terminating RLTs to ACD splits or VDNs is reduced equipment cost. A multiappearance voice terminal with display capabilities is less expensive than an attendant console and provides similar functionality in a CAS arrangement.

Replacing Attendant Positions

When RLTs terminate to ACD splits or VDNs, some attendant functions can be replaced by ACD agents or other voice terminal users. However, if attendants at the main perform CAS and non-CAS functions, it may not be possible to eliminate all attendant positions. For example, be swam that some features at the main such as Dial Access to Attendant, Intercept Treatment—Attendant, Attendant Control of Trunk Group Access, Alternate FRLs and Manual Time-of-Day Plan Changes cannot be used if all attendant positions are eliminated.

CMS (Call Management System) Measurements

Another advantage of terminating RLTs to ACD splits or VDNs is that ACD splits, ACD agents, and VDNs can be monitored by CMS. Attendant positions cannot be monitored by CMS.

Call Identification Tones, Zip Tones, and Recorded Announcements

For RLTs that terminate to ACD splits or VDNs, an answering position (at the CAS main) can receive one form of call identification; either Call Identification Tones, or zip tones, or recorded announcements. For example, if recorded announcements (VDN-of-Origin, Queue-of-Origin, or City-of-Origin) are provided at the main, Call Identification Tones or Zip Tones are not provided.

Agent Voice Terminals

For RLTs that terminate to ACD splits or VDNs, answering positions (at the CAS main) should be equipped with voice terminals that have a Conference or Transfer button. Furthermore, answering positions (at the CAS main) should be equipped with voice terminals that have display capabilities so that the user can distinguish between RLT and non-RLT calls. This is because the user operation for the Conference—Three Party and Transfer features is different for incoming RLT calls. The Conference—Three Party and Transfer features work normally for non-RLT calls. See the "User Operations" section of this feature description for more information about transferring incoming RLT calls.

Routing RLT Calls to Locations Outside of the CAS Arrangement

For RLTs that terminate to ACD splits or VDNs, incoming calls should not be routed to a destination outside of the CAS arrangement (by way of Call Forwarding, Look-Ahead Interflow, or Call Vectoring "route to" steps). This type of routing disables the dropout and reuse capabilities that make RLTs desirable.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

An answering position at the main cannot use the Abbreviated Dialing feature to transfer an incoming RLT call to a branch location. (The reason is that the Abbreviated Dialing lists are stored at the main, and digit collection and routing are performed by the branch.) A number that is stored within a voice terminal (not on the switch) can be used to transfer an incoming RLT call to a branch location. One example of a voice terminal that stores numbers is the 7103A Programmable voice terminal. A backup voice terminal user cannot transfer a call using the Abbreviated Dialing feature.

Attendant Auto—Manual Splitting

Manual splitting is inactive. The SPLIT lamp and button do not function on RLT calls.

Attendant Call Waiting

The Attendant Call Waiting feature is compatible with the CAS feature. When a CAS attendant transfers a call to a branch location and Attendant Call Waiting is assigned, the transferred call is allowed to wait (with 2-burst waiting tone) on an active single-appearance voice terminal at the branch.

Attendant Direct Extension Selection With Busy Lamp Field

When a console at a Centralized Attendant Service main location is handling a call from a branch location, the BLF gives no indication of busy/idle status of branch location voice terminals. An attendant's DXS (Direct Extension Selection) buttons cannot be used to call voice terminals at the branch.

The Attendant Direct Extension Selection With Busy Lamp Field feature is not available for backup voice terminals at branch locations.

Attendant Interposition Calling and Transfer

A call originated to a centralized attendant by way of an RLT cannot be transferred to another attendant using the Attendant Interposition Calling and Transfer feature.

Attendant Recall

Attendant Recall does not apply to calls originated by stations at branch locations. Also, if an attendant (or other answering position) transfers an incoming RLT call to a station at a branch location and the call goes to Coverage, Attendant Recall does not apply.

Automatic Call Distribution

The following ACD interactions apply to Issue 3.0 or later of Generic 2.1.

A centralized attendant, ACD agent, or other answering position at a CAS main can transfer (extend) an RLT call to an ACD split at a branch location. When ringback tone is heard, the answering position should release the call within 4 seconds. This allows the call to enter the split's queue.

For RLTs that terminate to ACD splits or VDNs, an answering position (at the CAS main) can receive either Call Identification Tones or zip tones, but not both. The same is true for recorded announcements. If recorded announcements (VDN-of-Origin, Queue-of-Origin, or City-of-Origin) are provided at the main, Call Identification Tones cannot be provided. However, an answering position that is equipped with a display voice terminal can be given information about the source of an incoming call. The combined information an answering position receives from zip tones, recorded announcements, and a display voice terminal is often a suitable substitute for call Identification Tones. Refer to *ACD From the Agent's Perspective* in the ACD feature description for more information about zip tones, recorded announcements, and display capabilities.

For RLTs that terminate to ACD splits or VDNs, answering positions (at the CAS main) should be equipped with display voice terminals so that the user can distinguish between RLT and non-RLT calls. The reason is that the user operation for the Conference—Three Party and Transfer features is different for RLT calls. The Conference—Three Party and Transfer features work normally for non-RLT calls. See the "User Operations" section of this feature description for more information about transferring incoming RLT calls.

While an observer, using Agent Override, is connected to an agent's call (at a CAS main), the agent cannot use the Conference—Three Party, Transfer, Call Waiting, or Hold features except to transfer an RLT call to a branch location.

Call Coverage

When a backup voice terminal in a CAS arrangement is assigned coverage, direct calls to that extension are redirected normally. However, attendant-seeking calls to a backup voice terminal do not redirect to coverage.

CDR (Call Detail Recording)

The CDR feature is provided by the CAS branch for calls transferred by a CAS attendant, backup voice terminal, or other answering position. However, an attendant cannot activate CDR trunk-group recordings of RLT trunk groups.

Call Vectoring

Vector processing is not available for incoming calls to backup voice terminals at a branch location. Entering a VDN in Procedure 211, Word 2 as the extension number of a backup voice terminal is not allowed. When this is attempted, an administration error will occur.

The following Call Vectoring interactions apply to Issue 3.0 or later of Generic 2.1.

A centralized attendant or other answering position at the CAS main can extend (transfer) an RLT call to a VDN at a branch location. If the answering destination specified by the vector is an ACD split, then when ringback tone is heard, the answering position should release (disconnect from) the call within 4 seconds. This allows the call to enter the split's queue.

A centralized attendant or other answering position at the CAS main cannot originate calls to a branch location by way of an RLT (Release Link Trunk). When this is attempted, the switch returns intercept tone. An answering position at the main can originate a call to a VDN at the branch by way of a tie trunk.

For RLTs that terminate to ACD splits or VDNs, an answering position (at the CAS main) can receive either Call Identification Tones or zip tones, but not both. The same is true for recorded announcements. If recorded announcements (VDN-of-Origin, Queue-of-Origin, or City-of-Origin) are provided at the main, Call Identification Tones cannot be provided. However, an answering position that is equipped with a display voice terminal can be given information about the source of an incoming call. The combined information an answering position receives from zip tones, recorded announcements, and a display voice terminal is often a suitable substitute for call Identification Tones. Refer to *ACD From the Agent's Perspective* in the ACD feature description for more information about zip tones, recorded announcements, and display capabilities.

For RLTs that terminate to ACD splits or VDNs, answering positions (at the CAS main) should be equipped with display voice terminals so that the user can distinguish between RLT and non-RLT calls. The reason is that the user operation for the Conference—Three Party and Transfer features is different for RLT calls. The Conference—Three Party and Transfer features work normally for non-RLT calls. See the "User Operations" section of this feature description for more information about transferring incoming RLT calls.

Conference—Three Party

Beginning with Issue 3.0 of DEFINITY Communications System Generic 2.1, the operation of the Conference—Three Party feature has been changed for an ACD agent (or other voice terminal user) who transfers an incoming RLT call to a branch location in a CAS arrangement. Refer to the User Operations section of this feature description for more information about transferring RLT calls.

DCS (Distributed Communications System)

Attendants serving several DCS nodes can all work at one location and still provide a wide range of services. For attendant features to work transparently, there must be at least one direct tie trunk between the attendants' location and each unattended node.

FRL (Facilities Restriction Level)

Centralized attendant positions (and other answering positions at the main) as well as tie trunks and RLTs are assigned FRLs. To achieve satisfactory performance, assign an appropriate FRL for these answering positions.

FADS (Force Administration Data System)

As an option, the FADS feature can be used with CAS to monitor attendant-related traffic data. The FADS feature is assigned to the CAS main location.

ISDN—PRI (Primary Rate Interface)

An ISDN—PRI trunk group can be used to provide tie trunk service between a CAS main to a CAS branch location. However, RLT trunk types (57, 66) cannot be assigned to an ISDN—PRI trunk group.

Look-Ahead Interflow

If a receiving switch is also the main location in a CAS arrangement, the receiving Look-Ahead Interflow vector can contain a "route to" step with the Attendant Dial Access code (Encode 8) as the destination. This "route to" step (usually used as an ***alternate*** destination) will unconditionally deliver calls to the local attendant queue. In the case of a C.AS main, the local attendant queue is also the CAS queue. The receiving vector can also be programmed to send calls to an ACD split or other answering position only if certain conditions are met. However, Look-Ahead Interflow calls must use ISDN—PRI trunks. Since ISDN—PRI trunks cannot be RLTs, these calls will not come in on RLTs. Refer to the Call Vectoring and Look-Ahead Interflow feature descriptions for more information about vector programming.

If a receiving switch is also a branch location in a CAS arrangement, the receiving Look-Ahead Interflow vector can contain a "route to" step with the Call to CAS Attendant dial access code (Encode 49) as the destination. This "route to" step (usually used as an ***alternate*** destination) will unconditionally deliver calls to the CAS queue at the main location.

Loudspeaker Paging Access

An answering position at the main has only 10 seconds to make a page, and then the paging circuit is released. This prevents the RLTs from being tied up.

Main/Satellite/Tributary

A Main / Satellite complex cannot use the Centralized Attendant Service feature. If a satellite location in a Main / Satellite arrangement is also translated as a branch location in a CAS arrangement, neither feature will work properly. (Tie trunks in the network will lock up.)

For Main / Satellite configurations, a centralized attendant capability can be provided by using the Extended Trunk Access function of the Main / Satellite feature.

MCT (Malicious Call Trace)

At a Branch Location

A malicious call placed to a branch location in a CAS arrangement cannot be controlled by an answering position at the main location. However, when the MCT feature is assigned at a branch location, malicious calls can be controlled by a special services attendant at the branch location if one is assigned.

At the CAS Main

A malicious call placed to the main location in a CAS arrangement can be controlled by a centralized attendant.

Override

When a CAS backup extension is handling an RLT call, an attempt to inter the conversation by another terminal using the Overrride feature is denied.

Precedence Calling

The Precedence Calling feature is compatible with the Centralized Attendant Service feature. That is, these features can be coresident in the same network. When this is done, the interface to the AUTOVON or DSN switch (the gateway switch) must be the same switch as the CAS main.

However, RLTs between branch locations and the CAS main cannot be precedence capable trunks, nor can RLTs be used to connect to the AUTOVON or DSN switch (incompatible trunk types required). While these two features can coexist in the same networking arrangement, they must coexist independently of one another. That is, they cannot be used together, except that the CAS attendant group can provide attendant services for the Precedence Calling feature.

Restriction — Attendant Control of Voice Terminals

This feature cannot be used by an attendant at the CAS main to control (restrict) voice terminals at a branch location. It does, however, work normal for extensions at the CAS main, and if an attendant position is assigned at the branch location, that attendant position can use the Attendant Control of Voice Terminals feature to control extensions at that branch.

Tenant Services

A partitioned System 85 or Generic 2 can function as a main location in a CAS arrangement. However, incoming RLTs (Release Link Trunks) are not assigned to specific partitions, so centralized attendants must reside in Attendant Partition 0.

A partitioned System 85 or Generic 2 cannot serve as a branch location in a CAS arrangement.

Through Dialing

When a switch is a branch location in the CAS arrangement, the CAS attendant can allow terminal users at the branch to through dial calls at the branch.

Timed Reminder

The CAS feature uses a recall timing process that is differenct from that provided by the Timed Reminder feature. The timed reminder interval for CAS is assignable in 2-second intervals up to a maximum of 62 seconds (Procedure 211, Word 1). The Timed Reminder feature functions for incoming attendant seeking calls extended to a local extension and the recall internal is 30 seconds.

Another difference between timed reminder for CAS and the Timed Reminder feature is that when the timed reminder interval expires at a branch location, the branch seizes an RLT and sends the call back to an attendant (or other answering position) at the main.

Transfer

Beginning with Issue 3.0 of Generic 2.1, the operation of the Transfer feature has been changed for an ACD agent (or other voice terminal user) who transfers an incoming RLT call to a branch location in a CAS arrangement. See the "User Operations" section of this feature description for more information about transferring RLT calls.

Unattended Console Service-Call Answer From Any Voice Terminal

When the branch switch is in the Call Answer From Any Voice Terminal mode of operation, incoming tie trunk and Remote Access calls are sent to the Call Answer From Any Voice Terminal queue.

Hardware Requirements

The CAS feature requires the following special hardware.

- Each RLT requires:
  - — One circuit of an SN233 or TN760C circuit pack (four circuits per SN233 or TN760C)

    or

    One channel of an ANN11 or TN767 (DS1) circuit pack (24 channels per ANN11 or TN767).
- Additional hardware is required to monitor Release Link Trunks. This equipment includes:
  - — 30A8 SSI (System Status Indicator) lamp panels (8 lamps per panel). As many as 15 SSIs can be assigned at the CAS main, as many as 3 SSIs can be assigned at a CAS branch.
  - — SN241 Contact Interface circuit pack (eight circuits per circuit pack).

    NOTE: The SN241 can be used only on a traditional module. There is no equivalent circuit pack for universal modules.

The branch location communications systems may require the following additional hardware.

- An SN253 or TN768 auxiliary tone plant per module
- Specially equipped backup voice terminal(s) (optional).
  - — 2514BM backup voice terminal sets
  - — 60A or KS-20778 headsets
  - — G15A handsets.

Feature Administration

The CAS feature is assigned on a per-system basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal), the FM (Facilities Management) feature, or the TCM (Terminal Change Management) feature.

On DEFINITY Communications System Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

For RLTs that terminate to ACD splits or VDNs, refer to the ACD and Call Vectoring feature descriptions for information about administering these features.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — CENTRALIZED ATTENDANT SERVICE
Common Administration for Main and Branch Locations

PROCEDURE	WORD	PURPOSE	SMT
001	1	Assigns the associated extensions to the primary extension for uniform numbering.	Yes
100	1	Assigns a dial access code and trunk type to a trunk group. The applicable trunk-type encodes are: 36 2-way tie trunk dial repeating both ways 57 CAS release link trunk l-way outgoing from branch 65 SN241 Contact Interface 66 CAS release link trunk l-way incoming at main.	No
101	1	Administers trunk-group characteristics for the trunk groups administered in Procedure 100, Word 1.	No
115	1	Assigns incoming trunk group termination to a CAS attendant, Special Services attendant, or ACD split. This procedure also displays incoming trunk group termination to a VDN (assigned in Procedure 031 Word 2).	No
150	1	Assigns the SN233 or TN760C equipment location of an RLT trunk to its trunk-group number.	No
155	1	Assigns the equipment location of an SN241 contact interface trunk to its trunk-group number.	No
275	3	Assigns the DCS node number of the main location.	Yes
286	1	Assigns Remote Access to Attendant for the CAS attendant.	Yes

ADMINISTRATION PROCEDURES — CENTRALIZED ATTENDANT SERVICE Administration for Main Locations			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
012	1, 2	Beginning with Issue 3.0 of Generic 2.1, an RLT trunk group can terminate to an ACD split or a VDN. Use this procedure to assign a name-data-base entry to an incoming RLT trunk group (for ACD splits) or a VDN.	
031	2	Beginning with Issue 3.0 of Generic 2.1, an RLT trunk group can terminate to a VDN. Use this procedure to specify the VDN to which a trunk group terminates.	Yes*
100	1	Assigns the dial access code and trunk type of the incoming RLT trunk groups. The applicable encode is: 66 CAS RLT — Incoming at main; 1-way automatic in.	No
101	1	Administers trunk-group characteristics for the trunk groups administered in Procedure 100, Word 1.	No
203	1	Assigns the RLT RELEASE button to the CAS consoles. The applicable encode is: 29 RLT RELEASE Button.	No
204	1	Designates the alphanumeric display for incoming calls to the CAS attendants. The applicable encodes are: R2 V1 to R2 V3: 320 CAS branch identification R2 V4 and later: 2320 CAS branch identification.	No
212	1	Administers the CAS main RLT trunk groups and the type of branch.	No
212	2	Administers lamp status for incoming RLTs to the main.	No
350	1	Assigns the first digit of the feature or trunk-group dial access codes (if required).	No
350	2	Assigns the CAS dial access codes. The applicable encodes are: 8 Attendant Dial Access Code 48 CAS attendant remote hold 64 CAS main lamp test.	No
* Display only procedure for the SMT.			

ADMINISTRATION PROCEDURES — CENTRALIZED ATTENDANT SERVICE Administration for a Branch Location			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Administers CAS backup terminal equipment location and class of service.	Yes
000	2	Administers hunt-to assignments for the CAS backup terminal.	Yes
100	1	Administers the special queue trunk group and the outgoing, RLT trunk group. The applicable encodes are 6 Special queue 57 CAS release link trunk l-way outgoing from branch.	No
101	1	Administers trunk-group characteristics for the trunk groups administered in Procedure 100, Word 1.	No
211	1	Administers CAS branch characteristics timed reminder interval, LDN tone, queue trunk group, and queue overflow level.	Yes
211	2	Administers the CAS branch backup extension associated with an RLT and start pulse activation.	Yes
252	1	Administers a tone plant (TN768) to a universal module for use with CAS.	No
252	2	Administers an auxiliary tone plant (SN253) to a traditional module for use with CAS.	No
275	4	Assigns Trunk-to-Trunk Transfer to the system class of service.	Yes
350	1	Assigns the first digit of the feature or trunk-group dial access codes (if required).	No
350	2	Assigns the CAS dial access codes. The applicable encodes are: 8 Attendant dial access code (for local attendant console, if provided) 45 Activate CAS control 46 Activate control of CAS backup terminal 47 Activate CAAVT (Call Answer From Any Voice Terminal) for CAS backup 48 CAS attendant puts remote call in hold state 49 Call to CAS attendant 50 CAS branch lamp test.	No

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — CENTRALIZED ATTENDANT SERVICE	
PATH NAME	**PURPOSE**
terminal-change extensions attributes	Assigns the class of service for the CAS backup terminal. Also, this screen administers the hunt-to extension number to the CAS backup terminal and assigns the associated extension numbers to the primary extension for uniform numbering.

The following is the applicable FM path name used with the AP 16.

FM SCREEN — CENTRALIZED ATTENDANT SERVICE	
PATH NAME	**PURPOSE**
facilities-mgmt system-parameters centralized-attnd	Specifies the timed reminder interval, sets the queue overflow level, and assigns the branch backup extension to an RLT.

Notes:

Code Calling Access

There are two types of Code Calling Access: Code Calling Access — Traditional and Code Calling Access — Universal. Code Calling Access — Universal is an implementation of the Code Calling Access feature and is available only on DEFINITY Communications System Generic 2 switches with one or more universal modules.

The Code Calling Access — Traditional feature is available on either System 85 switches or on Generic 2 switches that have one or more traditional call processing modules. Systems that have traditional modules and universal modules can use either implementation of the Code Calling Access feature or both.

The following table lists the most significant differences between Code Calling Access — Traditional and Code Calling Access — Universal.

	TRADITIONAL	UNIVERSAL
Code Calling ID	2 or 3 digits	3 digits
Tones	892-hertz	860-hertz
Zone chiming	Not available	Uses zone paging trunk circuits (Loudspeaker Paging).
User Operation	Paging party dials a called party code and hears chimeback tone.	Paging party dials the extension number of the paged party followed by a paging zone code. The paging party does not hear chimeback tone.
Hardware	SN253 Auxiliary Tone plant 89A Control Unit with or 2012D Transformer	Tone source is provided by the standard TN768 Tone/Clock generator circuit pack. 89A Control Unit with 2012D Transformer 278A Adapter with -24 Volt Power supply
Administration	Unchanged	The association between an extension number and a Code Calling ID must be administered (Procedure 000 Word 3).

Feature History and Development

Code Calling Access — Traditional was first available on System 85 in Release 1.

An option to have music while waiting for an answer-back was added for Release 2, Version 3.

Code Calling Access — Universal was first available in Generic 2.

Notes:

Code Calling Access — Traditional

Description

Code Calling Access — Traditional allows attendants, voice terminal users, and tie trunk users to page a specific individual using audible coded signals. The coded signals consist of chimes distributed by a loudspeaker system. The called party answers the page by dialing an answer-back code from any voice terminal within the switch.

This feature is especially useful for alerting people who are frequently away from their desk such as supervisors, security, or maintenance personnel. This feature also provides an option to have music while waiting for an answer-back.

Signaling

The code calling chime signals are 892-hertz tones. To resemble chiming, the volume of these tones is gradually reduced to silence.

Code Calling ID

One or more tone bursts are combined sequentially to represent a single digit of the 2- or 3-digit Called Party code. The interval between successive tone bursts in a digit signal is approximately 0.5 seconds. However, this interval can be adjusted (using switch settings on the SN253) to provide a different rate of chiming or to optimize the chiming rate for customer-provided equipment. The interval between successive digits of the Called Party code is approximately 1.5 seconds. The Called Party code is sounded three times. The interval preceding each repetition of the Called Party code is approximately 4.5 seconds.

After the Called Party code has sounded the third time, the associated code calling trunk becomes idle in 5 seconds. It is then ready for the next service request.

User Operations

The following are the user operating procedures for this feature.

To Page Someone Using Code Calling Access

From a voice terminal with the user remaining off-hook:

1. Go off-hook. [Dial tone]
2. Dial the Code Calling Access code. [Second dial tone]
3. Dial the 2- or 3-digit called party code. [Confirmation tone is heard followed by chimeback tone. Ringback tone or music is heard while waiting for the answer-back.]
4. User waits off-hook until the paged party answers the page. [The paging and paged are connected. Ringback tone or music is removed, and the called party code circuit is released.]

From a voice terminal with the user going on-hook:

1. Go off-hook. [Dial tone]
2. Dial the Code Calling Access code. [Second dial tone]
3. Dial the 2- or 3-digit called party code. [Confirmation tone is heard followed by chimeback tone. Ringback tone or music is heard while waiting for the answer-back.]
4. The user goes on-hook to wait for a call back. Ringback tone or music is removed, and the called party code is released.]

From the attendant console with the attendant remaining on the loop:

1. Press an idle loop button. [ATND lamp lights. If lit, the PA lamp goes out]
2. Press **[START]** . [Dial tone]
3. Dial the Code Calling Access code. [Second dial tone]
4. Dial the 2- or 3-digit called party code. [Confirmation tone is heard followed by chimeback tone. Ringback tone or music is heard while waiting for the answer-back. RING lamp lights.]
5. Attendant stays on the loop until the paged party answers the page. [The attendant and paged party are connected. Ringback tone or music is removed, and the called party code is released. ANS lamp lights.]
6. When finished with the paged party, press **[RELEASE]** . [ATND and ANS lamps go out. PA lamp lights.]

From the attendant console when the attendant releases the loop:

1. Press an idle loop button. [ATND lamp lights. If lit, the PA lamp goes out]
2. Press **[START]** . [Dial tone]
3. Dial the Code Calling Access code. [Second dial tone]
4. Dial the 2- or 3-digit called party code. [Confirmation tone is heard followed by chimeback tone. Ringback tone or music is heard after chimeback tone. RING lamp lights.]
5. Press **[RELEASE]** to wait for a return call. [The ATND and RING lamps go out, the PA lamp lights, and the called party code is released.]

To Answer a Coded Call Page When the Paging Party Remains Off-Hook:

1. Go off-hook. [Dial tone]
2. Dial the Code Calling answer-back access code. [Second dial tone]
3. Dial the 2- or 3-digit called party code. [The called party is connected to the calling party. Confirmation tone is heard by both parties.]

To Answer a Coded Call Page When the Paging Party Goes On-Hook:

1. Go off-hook. [Dial tone]
2. Dial the paging party's number. [Ringback tone]

Considerations

Answer-Back Maximum

At any one time, as many as six voice terminal users can wait for Answer-Back.

Busy Tone

Busy tone is heard when another party has seized the code calling circuit. Busy tone is also heard if the code dialed is awaiting callback.

Called Party Codes

The Called Party codes can consist only of a combination of the digits 1 through 5. The switch returns intercept tone when the digits 6 through 9 or 0 are dialed. Three-digit codes provide a maximum of 125 Called Party codes. Two-digit codes provide up to 25 Called Party codes.

Calling Back the Paging Party

The paged party must be aware of the paging party's number if the paging party goes on-hook before being called by the paged party.

Intercept Tone

Intercept tone is heard when:

- An invalid Code Calling access code is dialed.
- An invalid called party code *is* dialed.
- The tone generating circuit is out of service.
- An attendant console attempts to answer a page.
- The calling party did not remain off-hook while waiting for a response from the called party.

Shared Equipment — Common Zones

When Code Calling Access — Traditional and Loudspeaker Paging Access are both available on the same switch, customer provided amplifier and speaker equipment can be shared between the two features in common paging zones.

Whether equipment is shared or not, when both features serve common paging zones, care must be taken to ensure that both features are not active for the same paging zones at the same time. This is accomplished by connecting the seizure indication leads (CBS1 & 2) and the busy-out input leads (COS1 & 2) of the 89A Control Units serving the common paging zones. This will ensure that neither feature can be activated in a zone where the other feature is already active. These connections are shown in Figure 38-1.

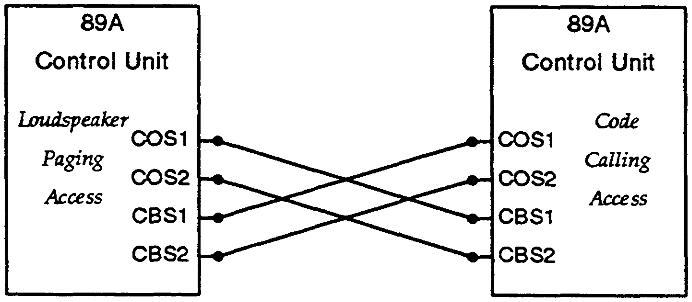

Figure 38-1. Lockout For 89A Control Units Serving Common Zones

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Call Park

Enabling Code Calling Access also enables Call Park.

Loudspeaker Paging Access

The Code Calling Access and Loudspeaker Paging Access features may use either common amplifiers and speakers or separate amplifiers and speakers. At all times, whether common or separate equipment is used, take care that both features are not used at the same time. Simultaneous use produces interference between the two audible signals. To prevent this interference, the lockout option on the 89A control unit can be set. This option allows only one feature at a time to access the amplifier and speaker equipment.

Music-on-Hold Access

When Music-on-Hold is implemented, music can be provided for a call on hold in an answer-back channel.

Remote Access

The Code calling Access feature and the Remote Access feature are not compatible. A caller attempting to use Code Calling Access over a Remote Access trunk will receive intercept tone.

Restriction—Toll Restriction

Terminal lines with Toll Restriction in the line class of service cannot use the number "1" as a first or second digit of the Code Calling Access code or answer-back code.

Restricting Feature Use

Code Calling Access can be busied-out by making a switch closure between COS1 and COS2 of the 89A control unit (see Figure 38-l).

Hardware Requirements

The Code Calling Access feature requires the following additional or special hardware.

- An SN253 auxiliary tone plant
- An 89A control unit
- A 2012D transformer to supply -48 Volt power to the control unit
- Customer-provided equipment to provide audible tones [an amplifier and speaker(s)].

Feature Administration

Assignment of the Code Calling Access feature is on a per-system basis.

On System 85 switches, this feat-me is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Communications System Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

To provide music while waiting for answer back, the Music-on-Hold feature must also be assigned.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — CODE CALLING ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to an extension number.	Yes
010	3	Assigns a Miscellaneous Trunk Restriction group to a voice terminal class of service.	Yes
100	1	Assigns the dial access code and trunk type of the Code Calling Access trunk group. The applicable trunk-type encode is as follows: 53 Code calling interface.	No
102	1	Assigns Code Calling Access to a Miscellaneous Trunk Restriction group.	Yes
202	1	Administers the Direct Trunk Group Selection button for the attendant console.	No
252	2	Administers the SN253 auxiliary tone plant for use with chime paging.	No
275	1	Assigns Code Calling Access to the system class of service and assigns the music-on-hold option (Field 7).	Yes
275	4	Specifies the number of Code Calling Access digits for system class of service.	Yes
350	1	Assigns the first digit of the dial access code (if required).	No
350	2	Assigns the answer-back dial access code. The applicable encode is: 18 Code Call answer-back.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — CODE CALLING ACCESS	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns a Miscellaneous Trunk Restrictions group to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Code Calling Access — Universal

Description

Code Calling Access — Universal allows attendants, voice terminal users, and tie trunk users to page a specific individual using coded chime signals, which are broadcast over a loudspeaker system. The paged party answers the page by dialing his or her extension number or a predesignated answer-back extension number from any voice terminal within the switch.

This feature is especially useful for alerting people who are frequently away from their work area, for example, supervisors, security, or maintenance personnel.

As an option, the paging party can listen to music while waiting off-hook for the paged party to call back.

Code Calling ID

The loudspeaker system broadcasts a 3-digit Code Calling ID. Each digit is represented by a corresponding number of chime signals (tones). For example, the Code Calling ID 123 is broadcast as one tone followed by two tones followed by three tones.

The chime signals are 860-hertz tones. To resemble chiming the volume of the tone is gradually reduced to silence.

The interval between successive tones in a digit signal is approximately 0.5 seconds. The interval between successive digits of the Code Calling ID is approximately 1.5 seconds. The Code Callin ID is repeated three times. The interval preceding each repetition of the Code Calling ID is approximately 4.5 seconds.

After the third repetition of the Code Calling ID, the associated paging trunk(s) becomes idle in approximately 5 seconds and is ready for the next service request.

Zone Chiming

The chime signals can be broadcast over one of 18 individual paging zones (areas) or over paging zones 1 through 5, which is called all-zones paging. The user dials a paging zone code to access one or more paging zones. Individual paging zones are numbered 1 through 18 and have corresponding paging zone codes. The paging zone code for all-zones paging is 0 (zero). If the system has nine or fewer paging zones, the paging zone codes are single-digit numbers (0-9). If the system has 10 or more paging zones, 2-digit paging zone codes must be dialed (00, ... 08, 09, 10, 11, etc.).

User Operations

The following are the user operating procedures for this feature.

To Page Someone Using Code Calling Access

From a voice terminal if the user remains off-hook:

1. Go off-hook. [Dial tone]
2. Dial the Code Calling access code. [Second dial tone]
3. Dial the extension number of the person you want to page followed by a paging zone code. [Ringback tone or music, whichever is administered]
4. Wait off-hook until the paged party answers the page.

From a voice terminal if the user goes on-hook:

1. Go off-hook. [Dial tone]
2. Dial the Code Calling access code. [Second dial tone]
3. Dial the extension number of the person you want to page followed by a paging zone code. [Ringback tone or music, whichever is administered]
4. Go on-hook and wait for a call back.

From the attendant console if the attendant remains on the loop:

1. Press an idle loop button. [ATND lamp lights. If lit, the PA lamp goes out]
2. Press **[START]** . [Dial tone]
3. Dial the Code Calling access code. [Second dial tone]
4. Dial the extension number of the person you want to page followed by a paging zone code. [Ringback tone or music, whichever is administered, is heard. RING lamp lights.]
5. Attendant stays on the loop until the paged party answers the page. [ANS lamp lights.]
6. When finished with the paged party, press **[RELEASE]** . [ATND and ANS lamps go out. PA lamp lights.]

From the attendant console if the attendant releases the loop:

1. Press an idle loop button. [ATND lamp lights. If lit, the PA lamp goes out]
2. Press **[START]** . [Dial tone]
3. Dial the Code Calling access code. [Second dial tone]
4. Dial the extension number of the person you want to page followed by a paging zone code. [Ringback tone or music, whichever is administered, is heard. RING lamp lights.]

5. Press **[RELEASE]** and wait for a call back. [The ATND and RING lamps go out, and the PA lamp lights.]

To Answer a Code Call Page if the Paging Party Remains Off-Hook:

1. Go off-hook. [Dial tone]
2. Dial the Code Calling answer-back access code. [Second dial tone]
3. Dial your extension number.

To Answer a Code Call Page If the Paging Party Goes On-Hook:

1. Go off-hook. [Dial tone]
2. Dial the predesignated answer-back extension number.

Considerations

Answer-Back Maximum

As many as six voice terminal users can wait for Answer-Back.

Code Calling ID

The digits of the Code Calling ID can be any number from 1 to 5. As many as 125 3-digit Code Calling ID can be assigned.

Calling Back the Paging Party

The paged party must know the answer-back extension number if the paging party goes on-hook while waiting for the paged party to call back.

Shared Equipment

The Code Calling Access and Loudspeaker Paging Access features share common tone generating circuitry, paging trunks, 89A control units or 278A Adapters, and loudspeaker equipment. With Code Calling Access — Universal, connections to common equipment are switched and the switch can detect and reject attempts by either feature to use equipment that is already engaged by the other feature.

Intercept Tone

Intercept tone is heard if:

- An invalid Code Calling access code is dialed.
- An invalid paged party extension number is dialed.
- An invalid paging zone code is dialed.

- The tone generating circuit is out of service.
- An attendant console attempts to answer a page.

Reorder Tone

Reorder tone is heard if the paging trunk(s) is already in use or an intercom record is not available.

Interactions With Other Features

The following Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

A call is not allowed to wait (via Attendant Call Waiting) on a line that has accessed code calling.

Busy Verification of Lines

Busy verification is denied on a voice terminal line that has accessed code calling. Attempts to busy verify a voice terminal line in this state results in the attendant hearing reorder tone.

CDR (Call Detail Recording)

An attendant with an incoming tie trunk call can dial the CDR access code and account code and then use code calling. The call record shows the call not the page.

Call Park

Enabling Code Calling Access also enables Call Park.

Call Waiting

A call is not allowed to wait (via Call Waiting) on a line that has accessed code calling.

Loudspeaker Paging Access

The Code Calling Access — Universal and Loudspeaker Paging Access features share the same loudspeaker equipment. Simultaneous use of these features over shared equipment is blocked by the switch.

Music-on-Hold Access

If Music-on-Hold is implemented, the paging party hears music if he or she remains off-hook while waiting for answer-back.

Override

Override is denied to a voice terminal line that has accessed code calling.

Priority Calling

A call is not allowed to wait (via Priority Calling) on a single-appearance voice terminal that has accessed code calling.

Tenant Services

The paging (chiming) zones for the Code Calling Access feature are not partitioned. Attendants in any attendant partition and voice terminal users in any extension partition have equal access to each paging zone.

Remote Access

The Code Calling Access and Remote Access features are not compatible. Callers attempting to use Code Calling Access via Remote Access trunks will receive intercept tone.

Restriction—Toll Restriction

Terminal lines with Toll Restriction in the line class of service cannot dial the digit "1" as the first or second digit of the Code Calling Access code or Code Calling answer-back access code.

Restricting Feature Use

Access to code calling can be restricted by assigning miscellaneous trunk restrictions to a voice terminal by way of the line class of service.

Hardware Requirements

The new Code Calling Access feature can only be implemented on a system that has at least one universal module. The feature requires the following additional or special hardware

- An auxiliary trunk circuit, TN763 for each paging zone.
- Customer-provided loudspeaker equipment to broadcast the chime signals.
- An 89A Control Unit, with a 2012D Power Transformer (-48 Volts). Each 2012D Power Transformer can support two 89A Control Units.

  or

  A 278A Adapter, with a -24 Volt Power Supply. The 2012D Power Transformer can be modified to provide -24 Volts using Parts Kit D18132.
- A 2012D power transformer to supply power to the 89A or 278A.

NOTE: If the Loudspeaker Paging Access feature is implemented, some or all of the equipment listed above may already be in place.

Feature Administration

Assignment of Code Calling Access — Universal is on a per-system basis. The Loudspeaker Paging Access feature must also be assigned to the system and zone paging trunks must be administered.

Code Calling Access — Universal is administered using DEFINITY Manager II.

To provide music while waiting for answer-back, the Music-on-Hold feature must also be assigned.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — CODE CALLING ACCESS		
PROCEDURE	**WORD**	**PURPOSE**
000	1	Assigns a class of service to an extension number.
000	3	Associates an extension number with a Code Calling ID.
010	3	Assigns a Miscellaneous Trunk Restriction group to a voice terminal class of service.
100	1	Assigns a dial access code, trunk type, and signaling type to the Loudspeaker Paging Access trunk group. The applicable trunk-type encode is: 54 Loudspeaker Paging interface.
102	1	Assigns a trunk-group dial access code to a Miscellaneous Trunk Restriction group.
150	1	Assigns equipment locations to trunks in the paging trunk group and associates a paging zone number with each trunk circuit.
275	1	Assigns Loudspeaker Paging Access/Code Calling Access to the system class of service and specifies whether users hear ringback tone or music (Field 7).
350	1	Assigns the first digit of the dial access code (if required).
350	2	Assigns the feature and answer-back dial access codes. The applicable encodes are: 105 Universal Code Calling/Chime Paging - access 106 Universal Code Calling/Chime Paging - answer back.
350	3	Assigns the feature encodes for the fixed-feature buttons on 71-series voice terminals. The applicable encodes are: 105 Universal Code Calling/Chime Paging - access 106 Universal Code Calling/Chime Paging - answer back.

Notes:

Conference—Attendant Five Party

Description

The Conference—Attendant Five Party feature is designed for a switch that has only universal modules (DEFINITY Generic 2). The Conference—Attendant Six Party feature should be used if the switch has one or more traditional call processing modules. If the Conference—Attendant Five Party feature is assigned to a switch that has traditional modules and universal modules, poor transmission quality could occur if trunks or voice terminals assigned to a traditional module are part of the conference. This section describes the Conference—Attendant Five Party feature. For a description of the Conference—Attendant Six Party feature, refer to the feature description in this manual.

The following table lists the most significant differences between the Conference Attendant Six Party feature and the Conference—Attendant Five Party feature.

	Six Party	Five Party
Number of conferees	Six conferees plus the attendant, for a total of seven.	Five conferees plus the attendant, for a total of six.
Bridged Call	Each conferee can have another party bridged onto the conference call. That is, a conference could have an attendant, six conferees, and six bridge-on parties.	A user can bridge onto a conference call if there are four or fewer conferees on the call. If a conference already has five conferees, bridge-on is denied.
Hardware	SN254 conference circuit pack (maximum 13 per switch).	No hardware is required. The maximum number of simultaneous attendant conferences is still 13.

The Conference—Attendant Five Party feature allows an attendant to set up a conference for up to five conferees, plus the attendant. Conferees do not have to be on the same switch. External conferees can be included in a conference call.

Once the conference has been established, the RELEASE button on the Attendant Console removes the attendant from the conference. The HOLD button holds the conference on the attendant console. An internal conferee (a conferee on the local switch) can recall the attendant for assistance or to add conferees.

User Operations

The following are the user operating procedures for this feature.

To Establish a Conference or Add a Conferee:

1. At the attendant console, after receiving instructions from the party requesting the conference, press the **[CONF]** key. (Requesting party is connected to the conference.)
2. Call each successive conferee by using the DXS and DTGS features, or by pressing the **[START]** key and dialing the applicable trunk access code and [or] extension number.
3. Announce the conference.
4. Press the **[CONF]** key. (New party is added to the conference.)
5. Repeat steps 2 through 4 until all conferees are included.
6. Press the **[RELEASE]** button to exit the conference, [PA lamp lights]

   or

7. Press the **[HOLD]** button if attendant assistance will be needed later. [Held loop lamp lights, and PA lamp lights.]

To Recall the Attendant Into the Conference (Local Conferee)

For terminals without a RECALL button:

Momentarily press the switchhook.

For terminals with a RECALL button:

Press the **[RECALL]** button.

To Reenter the Conference (Attendant):

At the attendant console, press the appropriate loop button. (Attendant is connected to the conference.)

Considerations

Attendant Service to Multiple Conferences

An attendant can be connected to only one conference at a time. To start another conference, the attendant can hold the first conference on the console or release from the conference call.

Trunks Only Conferences

If an attendant releases a conference connection with only trunk conferees, the trunk circuits are released. To avoid this, place the conference on hold.

Transmission Quality With Trunk Connections

Transmission quality deteriorates if more than two CO (Central Office) trunks or more than one CO trunk and two tie trunks are connected to a conference.

System Limits

The maximum number of simultaneous attendant conferences is 13.

Administrable Recall Buttons

Some multiappearance voice terminals do not have a fixed RECALL button. If RECALL buttons are needed for these voice terminals, use Procedure 054, Word 1 to assign the button.

Interactions With Other Features

The following features affect or are affected by the operation of this feature.

Attendant Call Waiting

Attendant Call Waiting is denied when the attendant is attempting to add a busy called party to an attendant established conference.

Attendant Display

The Attendant Display feature is denied when the attendant is connected to a conference.

Bridged Call

For voice terminals with bridge-on capability, it is possible to have the bridged-on party involved in an attendant established conference, if the conference limit (5) has not already been reached.

Busy Verification of Lines

Busy Verification of Lines is denied when attempted toward a line connected to a Conference—Attendant Five Party call.

CDR (Call Detail Recording)

The attendant can dial the CDR access code and account code before adding a voice terminal or a trunk to a conference. The CDR feature records the trunk or extension number.

Call Waiting

Call Waiting is denied when the called terminal is involved in an attendant established conference.

Conference—Attendant Six Party

The Conference—Attendant Five Party and Conference—Attendant Six Party features are mutually exclusive. That is, these two features cannot both be used on the same switch.

Dial Access to Attendant

The requesting party (first member of the conference) can use Dial Access to Attendant and be connected to the conference. However, once an attendant conference has been started, a voice terminal user cannot use Dial Access to Attendant and be added to an active conference. The attendant must originate the call to add a party to an active conference.

Extension Number Portability

Because the attendant may not know which extension numbers are local and which ones have been ported (ported extensions connect to the local switch via a tie trunk), all conferences should be held on the console until completion. If the attendant releases a conference from the console when only trunks are involved, the switch drops the conference to prevent it from locking up.

Hold

Voice terminal users who are participating in an attendant conference cannot use the Hold feature.

Loudspeaker Paging Access

When establishing a Conference—Attendant Five Party call, an attendant cannot connect the conference to the paging loudspeakers.

Override

Any attempt to use Override toward a terminal that is connected to a attendant conference is denied.

Precedence Calling

If a conferee in an attendant 5-party conference is preempted by the Precedence Calling feature, all parties hear preemption warning tone. Then, all conferees must go on hook, and if necessary, the attendant reestablishes the conference.

Priority Calling

Priority Calling is denied when the called terminal is involved in an attendant established conference call.

Privacy—Attendant Lockout

When the attendant reenters a conference connection in answer to a recall, the attendant connects to all conferees. Privacy is denied.

Remote Access

A remote access user can be a member of an attendant established conference only as the conference requester (first party added to the conference).

Restriction—Voice Terminal Restrictions

An Inward restricted terminal may be added to an established conference involving an incoming trunk.

Serial Calls

The Serial Calls feature is denied when the called party is involved in an attendant established conference.

Tenant Services

The Conference—Attendant Five Party feature can be accessed by an attendant in any attendant partition.

After a voice terminal or trunk user (the first conferee) calls an attendant (in a partition other than Attendant Partition 0 to establish a conference, the attendant can add conferees in the same partition by dialing the conferee's extension number. Voice terminal users in other Extension Partitions can also be added, but the attendant must dial the 7-digit private network number. Furthermore, transmission quality may deteriorate if more than two CO trunks or more than one CO trunk and two tie trunks are in the connection.

An extension in Extension Partition 0 can be included in any attendant conference.

An attendant in Attendant Partition 0 is allowed to add conferees from any partition in the switch or from any trunk group to an attendant conference.

An attendant recall by a conferee is usually directed to the attendant partition associated with that conferee's extension partition. However, when the establishing attendant has placed the conference on hold, an attendant recall by any of the conferees is directed to that attendant.

Timed Recall on Outgoing Calls

The Timed Recall on Outgoing Calls feature does not apply to an attendant conference connection.

Trunk Group Busy/Warning Indicators to Attendant

The trunk-group warning indicator lamp lights when five conferees are connected to the conference. The trunk-group warning indicator lamp goes out when the attendant releases from the conference. The trunk-group warning indicator lamp lights only on the console that is controlling the conference.

Trunk Verification—Attendant and Voice Terminal

Trunk Verification may not be used on a trunk that is involved in an attendant conference connection.

Unattended Console Service—Preselected Call Routing and Call Answer From Any Voice Terminal

If the attendant releases a Conference—Attendant Five Party call and then changes to an Unattended Console Service feature, attendant recall is denied. The conference must be held on the console prior to starting the Unattended Console Service feature to provide attendant service. When attendant recall is used from a conference held on the attendant console, the recall is activated toward the attendant console where the call is held even if Unattended Console Service is in effect.

Hardware Requirements

The Conference — Attendant Five Party feature should only be assigned to a switch that has only universal modules.

Feature Administration

Assignment of the Conference—Attendant Five Party feature is on a per-system basis. This feature is administered using DEFINITY Manager II.

ADMINISTRATION PROCEDURES CONFERENCE—ATTENDANT FIVE PARTY		
PROCEDURE	WORD	PURPOSE
054	1	Assigns the RECALL button to a multiappearance voice terminal without a fixed RECALL button. The applicable encode is: 27 Recall.
100	1	Administers a dummy dial access code and a trunk type. The applicable trunk-type encode is: 5 Attendant conference.
202	1	Assigns the CONF (Direct Trunk Group Selection) button to the attendant console(s).
204	1	Assigns the desired alphanumeric display for an attendant conference call.
275	3	Assigns the type of attendant conference to the system class of service. The applicable encode is: 1 Five-party.
350	1	Assigns the first digit of the dial access code for the attendant conference (if required).

Notes:

Conference—Attendant Six Party

Description

This feature allows an attendant to set up a conference for up to six conferees, plus the attendant. Conferees do not need to be on the same switch. External conferees can be included in a conference call.

Once the conference has been established, the RELEASE button on the Attendant Console removes the attendant from the conference. The HOLD button holds the conference on the attendant console. An internal conferee (a conferee on the host switch) can recall the attendant for assistance or to add conferees.

Feature History and Development

This feature was first available on System 85 in Release 1. An administrable recall button was first provided for R2 V4 and was also retrofitted to the R2 V2 and R2 V3 software packages.

User Operations

The following are the user operating procedures for this feature.

To Establish a Conference or Add a Conferee:

1. At the attendant console, after receiving instructions from the party requesting the conference, press the **[CONF]** button. (Requesting party is connected to the conference circuit.)
2. Call each successive conferee individually using the DXS and DTGS features, or by pressing the **[START]** button and dialing the applicable trunk-group access code and [or] extension number.
3. Announce the conference.
4. Press the **[CONF]** button. (New party is added to the conference circuit.)
5. Repeat the above steps until all conferees are included.
6. Press the **[RELEASE]** button to exit the conference completely, PA lamp lights]

   or

   Press the **[HOLD]** button if attendant assistance will be needed later. [Held loop lamp lights, and PA lamp lights.]

To Recall the Attendant Into the Conference (by Any Local Conferee)

For terminals without a RECALL button:

Momentarily press the switchhook.

For terminals with a RECALL button:

Press the **[RECALL]** button.

To Reenter the Conference (as an Attendant):

At the attendant console, press the appropriate loop button. (Attendant is connected to the conference circuit.)

Considerations

Attendant Service to Multiple Conferences

An attendant can be connected to only one conference at a time. To start another conference, the attendant can hold the first conference on the console or release from the conference call.

Trunks Only Conferences

If an attendant releases a conference connection with only trunk conferees, the conference circuit goes idle, and the trunk circuits are released. To avoid this, place the conference on hold.

Transmission quality deteriorates if more than two trunks are connected to a conference circuit.

System Limits

As many as 13 conference circuit packs (SN254) can be installed on the System 85 or the DEFINITY Generic 2 switch with a traditional module, providing for up to 13 simultaneous conferences.

A single attendant console has only one CONF button, regardless of the number of conference circuits provided in the switch.

Administrable Recall Buttons

Some multiappearance voice terminals do not have a fixed RECALL button. If RECALL buttons are needed for these voice terminals, RECALL buttons can be assigned to the terminals using Procedure 054, Word 1.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

Attendant Call Waiting is denied when the attendant is attempting to add a busy called party to an attendant established conference.

Attendant Display

The Attendant Display feature is denied to an attendant when the attendant is connected to a conference circuit.

Bridged Call

For voice terminals with bridge-on capability, it is possible to have the bridged-on party involved in an attendant established conference though not actually connected to the conference circuit. However, transmission quality may degrade.

Busy Verification of Lines

Busy Verification of Lines is denied when attempted toward an extension connected to a Conference—Attendant Six Party call.

CDR (Call Detail Recording)

If the attendant wants to add a terminal or a trunk to a conference, the CDR access code and account code could be dialed before the trunk or terminal number is dialed and added to the connection. The trunk or terminal is recorded.

Call Waiting

Call Waiting is denied when the called terminal is involved in an attendant established conference call.

Conference—Attendant Five Party

The Conference—Attendant Six Party and Conference—Attendant Five Party features are mutually exclusive. That is, these two features cannot both be used on the same switch.

Dial Access to Attendant

The requesting party (first member of the conference) can use Dial Access to Attendant and be connected to the conference circuit. However, once an attendant conference has been started, a station user cannot use Dial Access to Attendant and be added to an active conference circuit. The attendant must initiate the contact to add a party to an active conference circuit.

Extension Number Portability

Only the attendant console and local voice terminals can provide the disconnect supervision required to release a conference circuit once the conference is finished. Since the attendant may be unaware of which extension numbers are local and which ones have been ported (ported extensions connect to the local switch via a tie trunk), all conferences should be held on the console until completion. If the attendant releases a conference from the console when only trunks are involved, the switch drops the conference automatically to prevent the conference from locking up.

Hold

Voice terminal users who are participating in an attendant conference connection are denied access to the Hold feature.

Loudspeaker Paging Access

When establishing a Conference—Attendant Six Party call, an attendant cannot connect the conference connection to the paging loudspeakers.

Override

Any attempt to use Override toward a terminal that is connected to a attendant conference is denied.

Precedence Calling

If a tie trunk being used on an attendant conference is preempted by the Precedence Calling feature, all parties to the conference will hear preemption warning tone. The affected trunk will then be made idle, and the conference call will remain in effect minus the preempted trunk circuit.

Priority Calling

Priority Calling is denied when the called terminal is involved in an attendant established conference call.

Privacy—Attendant Lockout

When the attendant reenters a conference connection in answer to a recall, the attendant connects to all conferees. Privacy is denied.

Remote Access

A remote access user can be a member of an attendant established conference only as the conference requester (first party added to the conference circuit).

Restriction—Voice Terminal Restrictions

An Inward restricted terminal may be added to an established connection involving an incoming trunk.

Serial Calls

The Serial Calls feature is denied when the called party is involved in an attendant established conference.

Tenant Services

The SN254 attendant conference circuits (as many as 13) are not partitioned. These trunks, when provided, are equally accessible to an attendant in any attendant partition needing them.

After a voice terminal or trunk user (the first conferee) calls an attendant (in a partition other than Attendant Partition 0 to establish a conference, the attendant is only allowed to add conferees that would otherwise have access to the first conferee who is not a member of Extension Partition 0.

An extension in Extension Partition 0 can be included in any attendant conference.

An attendant in Attendant Partition 0 is allowed to add conferees from any partition in the switch or from over any trunk group to an attendant conference.

An attendant recall by a conferee is usually directed to the attendant partition associated with that conferee's extension partition. However, when the establishing attendant has placed the conference on hold, an attendant recall by any of the conferees is directed to that attendant.

Timed Recall on Outgoing Calls

The Timed Recall on Outgoing Calls feature does not apply to an attendant conference connection.

Trunk Group Busy/Warning Indicators to Attendant

When Conference—Attendant Six Party is provided, the associated trunk-group warning indicator is lighted when six conferees are connected to the conference circuit. The trunk-group warning indicator is extinguished when less than six conferees are connected to a conference circuit and when the attendant releases from the conference. The trunk-group warning indicator lights only on the console that is controlling the conference circuit.

Trunk Verification—Attendant and Voice Terminal

Trunk Verification may not be used toward a trunk that is involved in an attendant conference connection.

Unattended Console Service—Preselected Call Routing and Call Answer From Any Voice Terminal

If the attendant releases a Conference—Attendant Six Party call and then changes to an Unattended Console Service feature, attendant recall is denied. The conference must be held on the console prior to starting the Unattended Console Service feature to provide attendant service. When attendant recall is used from a conference held on the attendant console, the recall is activated toward the attendant console where the call is held even if Unattended Console Service is in effect.

Hardware Requirements

The Conference—Attendant Six Party feature requires the following hardware for a traditional module.

- One SN254 conference circuit pack per conference (maximum of 13 conference circuit packs per switch).

Feature Administration

Assignment of the Conference—Attendant Six Party feature is on a per-system basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES CONFERENCE—ATTENDANT SIX PARTY			
PROCEDURE	WORD	PURPOSE	SMT
054	1	Assigns the RECALL button to a multiappearance voice terminal without a fixed RECALL button. The applicable encode is: 27 Recall.	Yes
100	1	Administers the Dummy dial access code and the trunk type. The applicable trunk-type encode is as follows: 5 Conference—Six Party.	No
150	—	Assigns the SN254 equipment location of an attendant conference circuit pack to its trunk-group number.	No
202	1	Assigns the CONF (Direct Trunk Group Selection) button to the attendant console(s).	No
204	1	Assigns the desired alphanumeric display for an attendant conference call.	No
275	3	For Generic 2, assigns the type of attendant conference to the system class of service. The applicable encode is: 0 Six-party.	N/A
350	1	Assigns the first digit of the dial access code for the attendant conference circuit (if required).	No

Notes:

Conference—Three Party

Description

This feature allows voice terminal users to set up 3-party conferences without attendant assistance. This is done by adding the third party to an established 2-party connection. Participants can include stations that are external to the switch.

Feature History and Development

This feature was first available on System 85 in Release 1.

Beginning with Issue 3.0 of DEFINITY Communication System Generic 2.1, an RLT (Release Link Trunk) can terminate to an ACD split, a VDN (Vector Directory Number), or an attendant console. This enhancement changes the operation of the Conference—Three Party feature. For RLTs that terminate to ACD splits or VDNs, any voice terminal with a Conference or Transfer button can be used to transfer incoming RLT calls from the CAS main to a branch location.

User Operations

The following are the user operating procedures for this feature.

To Establish a 3-Party Conference From a 2-Party Connection

For terminals without CONFERENCE, RECALL, or TRANSFER buttons:

1. Momentarily press switchhook. [Recall Dial Tone] (Connected party is put on soft hold.)
2. Dial the third party. Wait for answer.
3. Momentarily press switchhook. (3-way conference)

For terminals with a RECALL button but not CONFERENCE and TRANSFER buttons:

1. Press the **[RECALL]** button. [Recall Dial Tone] (Connected party is put on soft hold.)
2. Dial the third party. Wait for answer.
3. Press the **[RECALL]** button. (3-way conference)

For terminals with CONFERENCE and TRANSFER buttons (non-RLT call):

1. Press the **[CONFERENCE]** or **[TRANSFER]** button. [Dial Tone] (Connected party is put on hard hold.)
2. Dial the third party. (To cancel conference request, press the appearance button of held call.)

   During ringback, or after the third party answers:

3. Press the **[CONFERENCE]** button. (3-way conference)

For terminals with CONFERENCE and TRANSFER buttons (RLT call):

1. After receiving the calling party's instructions, press **[TRANSFER]** or **[CONFERENCE]** .* [Dial tone] (Calling party is placed on hold.)
2. Dial the requested number.
3. If ringback (or Call Waiting ringback) is heard, press **[DISCONNECT],** **[RELEASE],**

   or

   Go on-hook. (The RLT is released.)

   > **NOTE:** If the called party does not answer before the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main.

   or

   If busy tone is heard, press **[TRANSFER]** or **[CONFERENCE]** (Voice terminal user is reconnected to the calling party.) and inform the calling party that the extension is busy.

   If the calling party does not want to wait for the called party to answer, press **[DISCONNECT]** , **[RELEASE]** ,

   or

   Go on-hook. (The RLT is released.)

   or

   If the calling party wants to wait for the called party to answer, press **[TRANSFER]** or **[CONFERENCE]** . [Dial tone] (Calling party is placed on hold at the main.)

   Dial the Remote Hold dial access code. [Confirmation tone] (Calling party is placed on hold at the branch.)

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* For incoming RLT calls, the Conference button works the same as the Transfer button That is, pressing the Conference button transfers an incoming RLT call back to the originally called branch location. Pressing the Conference button does not set up a 3-party conference.

Press **[DISCONNECT]** , **[RELEASE]** ,

or

Go on-hook. [Dial tone] (The RLT is released.)

NOTE: If the calling party is a station at a branch location, Remote Hold will be denied.

NOTE: When the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main. The terminal user can attempt to complete the call again.

To Drop the Third Party (while on 3-party connection, or with two parties ringing a third Party)

For terminals with a DROP button:

Press the **[DROP]** button. (You are mconnected to the original party.)

For terminals with a RECALL button but not a DROP button:

Press the **[RECALL]** button. (You are reconnected to the original party.)

For terminals without a DROP or RECALL button:

Momentarily press switchhook. (You are reconnected to the original party.)

Using Meet-Me Conference to Converse With Two Calling Parties On a Multiappearance Terminal:

1. Be sure there is an active call on one appearance.
2. Receive another call on an idle appearance. [Voice terminal rings.]
3. Press the **[HOLD]** button to place the active call on hold. [Green status lamp flutters.]
4. Select the ringing appearance. [The new call is now active.]
5. Press the **[CONFERENCE]** button for the newly active call. [An idle appearance is automatically selected.]
6. Select the held appearance. [The original call is again active.]
7. Press the **[CONFERENCE]** button for this active call. [You and the two calling parties are now in a 3-way connection.]

Considerations

Trunk Connections

If the conference host (the System 85 or Generic 2 terminal user) releases from a 3-party conference and both of the remaining conferees are on trunk circuits (stations external to the switch), the switch may disconnect the remaining parties. This would occur if the trunk-to-trunk transfer option is not assigned, or if at least one of the trunks is not handling ***an incoming call*** to the switch. The Trunk-to-Trunk Transfer option is assigned in the system class of service (see Feature Administration).

Hard and Soft Processor Swaps

A stable 3-party conference will endure a hard processor swap.

During a hard swap, a voice terminal user cannot establish a 3-party conference.

The Conference—Three Party feature operates normally during a soft processor swap.

Voice Terminals That Cannot Be Used to Transfer RLT Calls

Voice terminals that are not equipped with a Conference or Transfer button cannot be used to transfer an incoming RLT call from the CAS main to a branch location.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

ACD (Automatic Call Distribution)

When an ACD agent adds another agent to the agent's call, the resulting conference is not considered a work-related activity for the second agent. The second agent is not removed from the agent queue.

The Conference—Three Party feature is denied to an agent while an observer (using agent override) is connected to the agent's call.

A voice terminal user on a 2-party call is allowed to add the recipient of an ACD call to a 3-party conference. However, the second button press (of the switchhook, RECALL button, or CONFERENCE button) in the conferencing operation is ignored until the receiving agent has actually answered the call. (That is, the second button press is ignored while the call is queued.)

Bridged Call

Bridging is allowed on an appearance that is being held for conference.

For multiappearance voice terminals while bridging is active, the CONFERENCE button is inoperable during a bridged appearance call for both the controlling terminal (extension that originated the bridged call) and the bridged terminal. A user can bridge on after a conference is established.

For a straight line set while bridging is active, pressing the RECALL button (if provided) and momentarily pressing the switchhook are ineffective for both a controlling terminal and a bridged terminal. A user can bridge on after a conference is established.

Busy Verification of Lines

Busy Verification of Lines is denied when attempted toward an extension connected to a Conference—Three Party call, unless the extension appears on a multiappearance terminal with more than one appearance. In this case, the verification attempt routes to an idle appearance of the extension; if any.

Call Coverage

After the call goes to coverage, when the covering user presses the CONFERENCE button, the temporary bridged appearance is removed from the principal's terminal.

A user attempting to activate the Conference—Three Party feature from a multiappearance terminal during the caller response interval of a coverage call is denied. The button press is ignored.

CDR (Call Detail Recording)

The CDR feature records tht last voice terminal connected to a 3-party conference call as the called number.

Call Vectoring

A multiappearance voice terminal user on a 2-party call is allowed to add the recipient of a VDN call to a 3-party conference. However, the second press of the CONFERENCE button in the conferencing operation is ignored until the recipient of the VDN call has actually answered the call. (That is, the second button press is ignored during vector processing.)

A single-appearance voice terminal user on a 2-party call is also allowed to add the recipient of a VDN call to a 3-party conference. However the controlling party must execute the second button press (of the switchhook or the RECALL button) after the recipient of the VDN call has actually answered the call. If the controlling party executes the second button during vector processing, the soft held party is reconnected to the controlling party, and the conference attempt does not succeed.

Call Waiting

Call Waiting is denied when the called terminal is involved in a 3-party conference.

When attempting to use Call Waiting toward a busy third party (to be added to a conference), if the second party is on soft hold, busy tone will be returned. Call Waiting is denied toward a busy third party if the calling terminal has another party on soft hold.

Centralized Attendant Service

The following CAS interaction applies to Issue 3.0 or later of Generic 2.1.

If RLTs terminate to answering positions other than attendant consoles, the answering positions should be equipped with display voice terminals so that the user can distinguish between RLT and non-RLT calls. The reason is that the user operation for the Conference—Three Party feature is different for RLT calls. The Conference—Three Party feature works normally for non-RLT calls. For more information, refer to the "User Operations" section of this feature description.

EUCD (Enhanced Uniform Call Distribution)

When an EUCD agent adds another agent to an EUCD call, the resulting conference is not considered work-related activity for the second agent. The 106B status indicator shows the second agent as engaged in non-EUCD activity.

While an observer (using agent override) is connected to an EUCD agent's call, the Conference—Three Party feature is denied for use by the agent.

Hold

Used in conjunction with hard hold, the Conference—Three Party feature allows the calling voice terminal user to transfer a second call while holding the the first call. After transferring the second call, the calling voice terminal user flashes the switchhook and dials the Call Hold access code to return to the call on hold.

IPA (Interpartition Access)

A voice terminal user (in a partition other than Extension Partition 0) is allowed to establish 3-party conferences that include participants that the user is otherwise allowed to call. If the user tries to add a conferee to the conference by dialing an extension number in another partition group, the switch returns intercept treatment to the user.

A voice terminal user in Extension Partition 0 is allowed to establish 3-party conferences with any voice terminal or over any trunk in the switch.

LWC (Leave Word Calling) in a DCS (Distributed Communications Service)

In a DCS environment without an Applications Processor and after a 3-party internode conference is established, the switch denies activation of Leave Word Calling.

Look-Ahead Interflow

At a receiving switch, the Look-Ahead Interflow feature and the 3-party Conference feature are compatible. The answering voice terminal user at the receiving switch can normally conference a Look-Ahead Interflow call with a third party inside or outside the receiving switch.

Malicious Call Trace

When Malicious Call Trace is activated during a 3-party conference, the malicious caller's identity in the trace information becomes uncertain. The contents of Items 2 and 5 and of Items 3 and 6 in the controlling attendant's display can be reversed.

If there is one (and only one) party in the 3-party conference who is connected to the switch over an incoming trunk. The MCT software presumes that this party is the malicious caller. This party is identified as the malicious caller in Items 2 and 3 of the controlling attendant's display.

Sometimes, this ambiguity can be avoided with human intervention. Since the malicious caller usually initiates a malicious call, the originally called party can press the DROP button to disconnect the third party in the conference, verify that the malicious caller is still on the connection, and then quickly activate the trace. (However, when this is done, the third party cannot be readded to the conference.)

Music-on-Hold Access

Music is provided to the first call put on hold when establishing a Conference—Three Party call. When an established Conference—Three Party call has been put on hold and one of the parties goes on-hook, music is not given to the remaining party on hold.

Override

Override is denied when directed toward a extension connected to a Conference—Three Party call, unless the extension appears on a multiappearance terminal with more than one appearance assigned. In this case, the override attempt routes to an idle appearance of the extension, if any.

Precedence Calling

If a tie trunk being used on a 3-party conference is preempted by the Precedence Calling feature, all parties to the conference will hear preemption warning tone; the affected trunk will then be made idle, and the conference call will revert to an otherwise normal 2-party connection.

Priority Calling

When calling a single-appearance voice terminal in a conference created by the Conference—Three Party feature using the Priority Calling feature, the call is denied.

When the Priority Calling feature is activated toward a multiappearance terminal in a Three Party Conference, the call completes to an idle appearance. If no appearance is idle, the priority call is denied.

Priority Calling is denied when the calling party (party attempting to set up a conference call) is holding a second party in the soft hold state. If the held party is in the hard hold state, Priority Calling is allowed.

Recorded Telephone Dictation Access

The Conference—Three Party feature cannot be used in conjunction with the Recorded Telephone Dictation Access feature. Any attempt to access a recorded telephone dictation trunk from a 2-party connection is denied.

Remote Access

A Remote Access user can be a member of a 3-party conference but cannot establish the conference. To be a member of a 3-party conference, the remote access user first calls a local station user, and then that local station user must establish the conference by adding the third party.

Restriction—Miscellaneous Trunk Restrictions

A terminal restricted from accessing a trunk group due to the Miscellaneous Trunk Restrictions feature can be added to a conference, via Conference—Three Party, involving the restricted trunk group. However, the restricted terminal cannot originate a conference, via Conference—Three Party, involving the restricted trunk group.

Restriction—Voice Terminal Restrictions

An Inward restricted terminal may be added to an established connection involving an incoming trunk via the Conference—Three Party feature.

An Outward restricted terminal can access a public-network trunk if the restricted terminal calls an unrestricted terminal. From here the unrestricted terminal could use the Conference—Three Party feature to connect the restricted voice terminal to the public network.

An origination-restricted single-appearance terminal may originate a call from a 2-party connection by first accessing the Conference—Three Party feature.

Serial Calls

For terminals without CONFERENCE or TRANSFER buttons, when Serial Calls is in effect, pressing the RECALL button at a local terminal or momentarily pressing the switchhook, recalls the attendant. Therefore, these terminal users cannot use the Conference—Three Party feature during a serial call.

Tenant Services

A voice terminal user (in a partition other than Extension Partition 0) is allowed to establish 3-party conferences that include participants that the user is otherwise allowed to call. If the user tries to add a conferee to the conference by dialing an extension number in another partition, the switch returns intercept treatment to the user.

A voice terminal user in Extension Partition 0 is allowed to establish 3-party conferences with any voice terminal or over any trunk in the switch.

Timed Recall on Outgoing Calls

With timed recall, a trunk group is assigned a recall interval at which time a call using that trunk group is routed to the attendant. The attendant can then determine whether the call should be allowed to continue or should be terminated. This prevents users from tying up critical trunk facilities. A 3-party conference call could involve up to two such trunks. The first trunk to time-out receives recall treatment (that is, times out and is routed to the attendant) even after the addition of a second trunk to the call. After the first timed recall, the trunk with the shorter recall time controls the recall treatment.

Trunk Verification—Attendant and Voice Terminal

Trunk Verification is denied when attempted toward a trunk connected to a Conference—Three Party call.

Unattended Console Service—Preselected Call Routing

When the Preselected Call Routing feature is in effect, the Conference—Three Party feature is automatically provided to the preselected voice terminal regardless of the extensions class of service. However, a single-appearance preselected voice terminal must have Conference—Three Party assigned to its extension class of service to enable the user to direct incoming calls to trunks. All multiappearance voice terminals have Conference—Three Party capabilities.

Restricting Feature Use

While the basic voice terminal restrictions apply, they do not specifically restrict this feature. Outside the context of class of service, Miscellaneous Trunk Restrictions, and Attendant Control of Voice Terminals restrictions (already mentioned), this feature is not specifically restricted.

Hardware Requirements

None.

Feature Administration

Assignment of the Conference—Three Party feature is on a per-extension class of service basis for single-appearance terminals. Multiappearance terminals are always provided with the Conference—Three Party feature (not administrable). When required, the trunk-to-trunk transfer option is administered on a per-system basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Communications System Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — CONFERENCE—THREE PARTY			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the voice terminal class of service to an extension number.	Yes
010	1	Assigns the Conference—Three Party feature to a voice terminal class of service.	Yes
275	4	Assigns the trunk-to-trunk transfer to the system class of service.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — CONFERENCE—THREE PARTY	
PATH NAME	PURPOSE
terminal-change class-of-service attributes	Assigns the Conference—Three Party feature to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Data Call Setup

Description

Data Call Setup provides access to System 85 and DEFINITY Generic 2 data services from most types of data endpoints. Data endpoints may be located on-premises or off-premises and may use either analog or digital interfaces. Standard data rates supported are:

- Low (below 300 bps)
- 300 bps
- 1200 bps
- 2400 bps
- 4800 bps
- 9600 bps
- 19.2 Kbps.

Other data rates can be supported for special applications such as ACCUNET Service Interface, DMI (Digital Multiplexed Interface), and ISDN (Integrated Services Digital Network). For these applications, data rates of 56 Kbps and 64 Kbps are available as required.

Call Setup Methods

Four basic methods are available for setting up data calls:

- Analog data call setup
- Digital Voice terminal dialing
- Keyboard dialing which includes:
  - — Basic terminal dialing
  - — Data hot line
  - — Default terminal dialing
  - — Mnemonic dialing.
- Host computer dialing.

Each of these methods is described in detail later in this section.

Data Call Connectivity

On-Premises Connections

On-premises data call setup connections can be either analog or digital. Either type of connection works satisfactorily. However, because both the System 85 and DEFINITY Generic 2 are digital switches, certain advantages can be derived from the full digital connection option.

While on-premises data calls can be either analog or digital, they are generally expected to be compatible. That is, an on-premises data call starting at an analog station is expected to go to another analog station. A data call that starts on a digital station is expected to terminate to another digital station. The data features have enough flexibility to support these expectations. If, for some reason, it is necessary to convert on-premises data calls between analog and digital formats, "conversion resources" must be provided. These conversion resources are special hardware configurations not feature supported. The Modem Pooling feature does not normally support on-premises data call setup.

Off-Premises Connections

The Data Call Setup feature can also be used for off-premises data calls to or from local (on-premises) data end points. Off-premises data connections are made using trunks or off-premises terminal lines. Several different arrangements are available for off-premises data call setup, and the switch offers several features that support these arrangements. Different arrangements available include:

- Calling from the analog public network via DID (Direct Inward Dialing)
- Calling over an analog private network connection
- Calling from a digital public network service (ACCUNET Service or ISDN)
- Calling over a digital private network (using the DS1 Interface feature or ISDN facilities)
- Calling to or from an Off-Premises Data Only station.

Like on-premises data calls, off-premises data calls can use either analog or digital interfaces. However, with off-premises connections, there is no assumption of uniformity. That is, it is assumed that incoming analog calls will attempt to reach digital interfaced end points, and that digital interfaced stations on the switch will need to place outgoing calls over analog trunks. The Modem Pooling feature is provided to handle these requirements. Modem Pooling supports off-premises data calls (both incoming and outgoing). Modem Pooling is described in detail separately in this manual.

Terminal Equipment Used

The following terminal equipment configurations can be used for data call setup:

Analog Voice Terminal Configurations

- A single-line analog voice terminal equipped with the Transfer feature

  These terminals can provide dial access to data services for data terminals. The data terminal may or may not be associated with the voice terminal. This arrangement is most likely to be used by the attendant who has a voice terminal located next to the attendant console. Restricted data users can be routed to the attendant for call completion. But, attendant-extended calls (via the console) are not provided with Data Protection or Modem Pooling support. The adjacent voice terminal can be used to extend the call with Data Protection active and to obtain Modem Pooling support when required.

- A single-line analog voice terminal equipped with a modem

  This arrangement is the traditional dedicated voice terminal. This is the same configuration that is typically used with analog switches. A voice terminal and modem is dedicated to the data terminal for data calling. The voice terminal can be used for voice calling when the data terminal is not in use. However, this arrangement does not support simultaneous voice and data service.

Digital Voice Terminal Configurations

- A DCP (Digital Communication Protocol) voice terminal (e.g., 7404D, 7405D, etc.), a data module or data stand, and an assigned EIA data terminal [e.g., 513 BCT (Business Communications Terminal)]

  These digital voice terminals can be assigned one or more voice and data extensions and be equipped with DATA buttons. The DATA buttons are used to transfer a data call from the voice extension to a specific data extension. The DATA button also provides enhanced features used with data calling. The Transfer feature can be used for Data Call Setup if a DATA button is not provided. The data terminal can be administered to perform dialing from the keyboard, if desired.

- An ISDN—BRI (Basic Rate Interface) voice terminal (such as a, 7505, 7506, or 7507), equipped with an ADM-T (Asynchronous Data Module with T Interface) terminal adapter, and an assigned data terminal (such as a, 513 BCT or AT&T 4425 terminal).

  These BRI voice terminals can be assigned a separate data extension. Unlike DCP terminals, they cannot be assigned DATA buttons as feature buttons. A data button on a BRI terminal is a *call appearance button* rather than a feature button. The data terminal can perform dialing from the keyboard.

Data Terminal Only Configuration

- A data terminal without an associated voice terminal

A data terminal can use a stand-alone data module such as a PDM (Processor Data Module) or 7400A or 7400B* Data Module for DCP interface or the 7500 Data Module for ISDN—BRI interface to the switch without having a directly associated voice terminal. With the 7400A or 7400B Data Module (or the PDM), data calls can be set up from a DCP voice terminal using the Transfer feature, One Button Transfer (from an assigned DCP voice terminal), or data calls can be set up using Keyboard Dialing techniques (including Default Dialing or Hot Line) from the data terminal itself. For ISDN—BRI terminals with the 7500 Data Module in a stand alone configuration, voice terminal data call setup is not available.

Hybrid Voice Terminal Configurations

- A hybrid voice terminal (such as the 7205H) with an associated data module and EIA data terminal

This terminal can be assigned a voice extension and data extension(s) and be equipped with DATA buttons. The DATA button is used to transfer a data call from the voice extension to a specific data extension. The voice extension is used to dial the data service extension or access code. The hybrid voice terminal's DATA button also provides enhanced features used specifically with data calls *(see **Data Button Functions***). The Transfer feature can be used if a DATA button is not provided.

NOTE: The 7405H Hybrid Voice Terminal was Manufacture Discontinued after January, 1987, and is available only while supplies last.

3270 System Configurations

- IBM† 3270 System Terminals and Cluster Controllers

IBM 3270 System equipment can be used with System 85 or DEFINITY Generic 2 switches. For this configuration, special 3270 Series Data Modules perform conversion and necessary buffering. This configuration provides digital connections between terminals and the host computer for 3270 type equipment, without a need for coaxial cabling (uses standard building wiring). With this configuration, the cluster controller views the switch ports as terminals. This arrangement is shown in Figure 42-1.

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* The 7400B Data Module can function in either the stand alone configuration (data terminal only) or with an associated (linked) voice terminal. A switch is provided (internally) to set the configuration

† Registered trademark of IBM Corporation

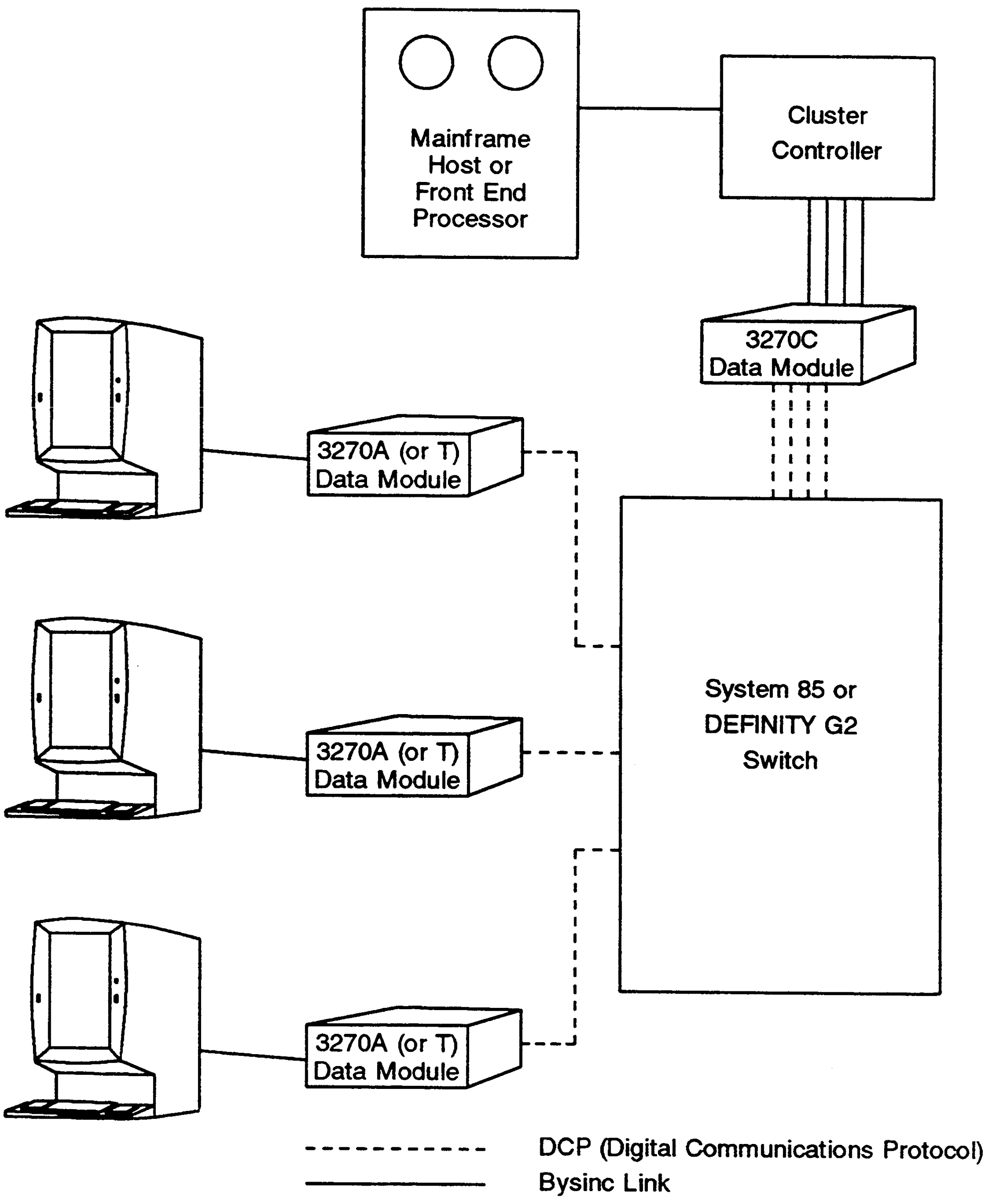

Figure 42-1. 3270 Equipment Configuration

Personal Computer Configurations

- The PC Interface feature provides several conjuration groups that can be for Data Call Setup.

  This arrangement allows a compatible PC (Personal Computer), such as the AT&T PC6300, to connect to the switch using either DCP or ISDN—BRI. These interfaces permit the PC to take advantage of the data communications features of the System 85 or DEFINITY Generic 2 switch. The PC Interface feature with its associated configurations is described in detail in its own chapter of this manual.

Integrated Voice/Data Station Configurations

- An integrated voice / data station (such as the AT&T Personal Terminal 510D or the 515 BCT)

  These voice / data terminals are equipped with an integral voice terminal and DCP data module for data call setup operations. Dialing can be performed using the terminal keyboard or from the voice extension.

NOTE: The term ***Voice/Data Station*** is used to describe an arrangement where a voice terminal and a data terminal share the same interface. Examples of Voice / Data Stations include a 7405D (DCP) voice terminal with a DTDM (Digital Terminal Data Module) and its associated data terminal; or a 7505 (BRI) voice terminal with an ADM-T and its associated data terminal.

Analog Data Call Setup

Analog connections require the use of a dedicated modem (modulator demodulator). This is the traditional arrangement that is commonly associated with analog switches. The modem converts digital (EIA) signals from a terminal or other digital device to an analog signal. This analog signal is usually a ***frequency shift keyed*** format that uses a (relatively) high frequency analog signal for a bit and a low frequency analog signal to represent a no bit. The specific analog format used is generally switch independent, and is relevant only to the modems at either end of the analog connection (this is the general case; there are, as always, exceptions).

Analog data calls work well at low data rates, however, because they are carried over voice band circuits and the band widths used for standard voice range frequencies is narrow, this technique is limited in the data rates it can support. Most modems support data rates of 1200 bps or less. Some modems are available that can support data rates of up to 4800 bps, however, these modems are generally more expensive. Figure 42-2 shows the basic analog arrangements available with System 85 or DEFINITY Generic 2 switches.

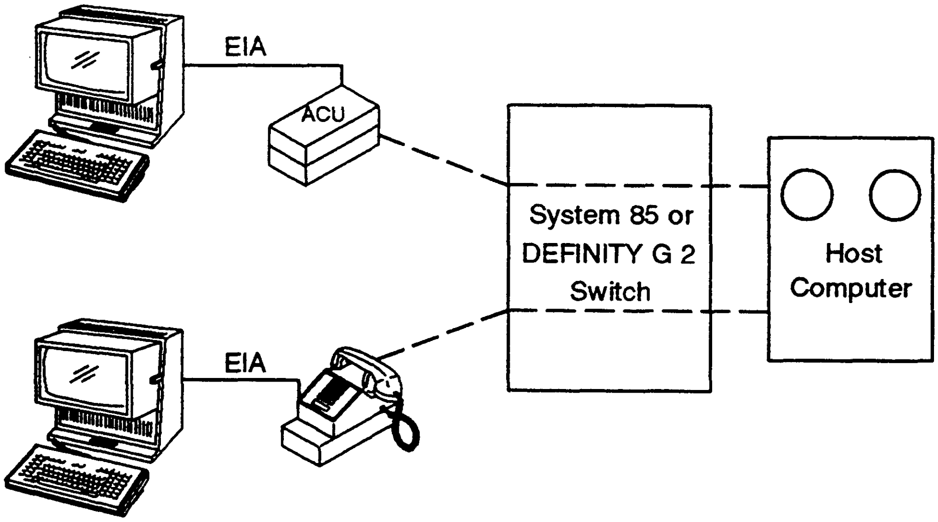

Figure 42-2. Data Call Setup Analog Arrangements

Analog Call Setup Methods

Analog data call setup uses either an analog voice terminal and an associated modem or an ACU (Automatic Calling Unit). With either method, the dialing istrument is assigned to the modem and cannot be used separately while a data call is in-progress. For example, with an analog voice terminal the user dials the number for the desired data port and listens to the call-progress tones. When the modem at the data port returns ***ready tone,*** the user presses a data button that transfers control of the local end of the call to the modem. The telephone used to setup the data call cannot now be used to setup or receive any other calls as long as the data call is in progress (no dial tone locally and any incoming calls receive busy tone).

With analog interfaces at both ends of the connection, two alternatives are available for connecting the host computer. One is to use a line appearance and modem set up like an analog interfaced terminal, as shown in Figure 42-2. The other is to use the DCA (Data Communications Access) feature. The DCA feature also uses modems; however, this feature uses trunk appearances rather than line appearances. The DCA feature is described in detail in its own chapter of this manual.

Digital Data Call Setup

BRI (Basic Rate Interface) or DCP (Digital Communications Protocol) and Data Modules

Digital data connections on both the System 85 and DEFINITY Generic 2 switches use a data module in place of a modem. Data modules convert the EIA signals from data terminals and other digital devices to the internal digital protocol DCP (on either System 85 or DEFINITY Generic 2 switches) or to ISDN protocol in the case of ISDN—BRI terminals on DEFINITY Generic 2 switches. Both DCP and ISDN—BRI use standard building wiring (4-wire metallic conductor) to provide two communications channels and a signaling channel. Using either DCP or ISDN—BRI, one interface can support both a digital data terminal and a digital voice terminal and also provide signaling services for other features such as the Display — Voice Terminal feature. Figure 42-3 shows some typical on-premises digital data call setup connections.

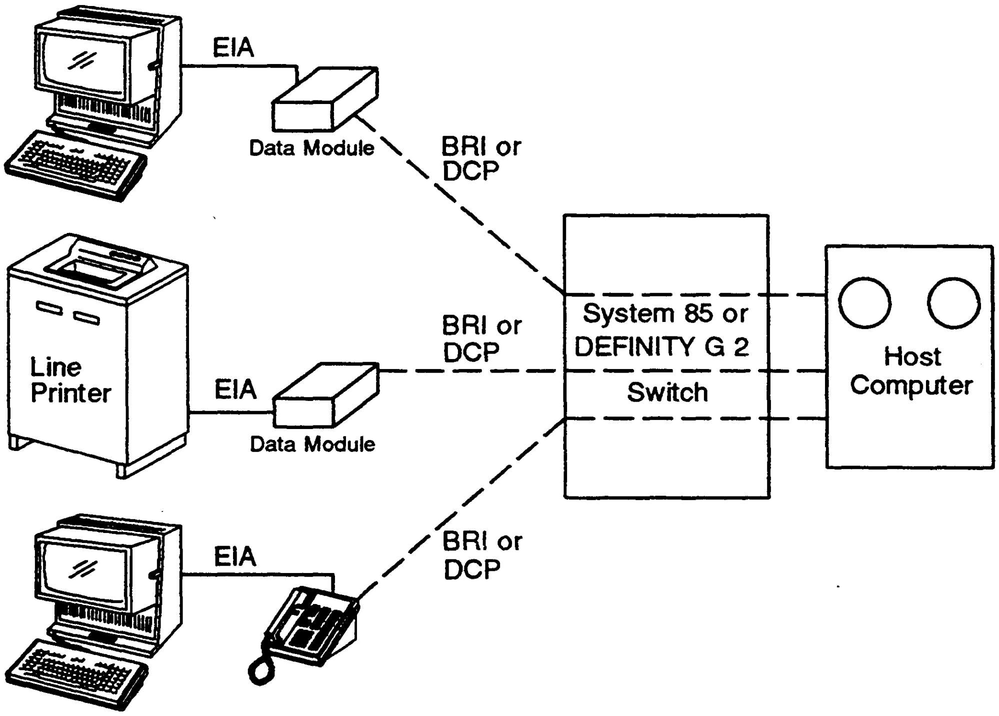

Figure 42-3. Data Call Setup Digital Arrangement

Digital Voice Terminal Dialing

With a digital voice terminal and an associated data module, the voice terminal is used in much the same way as for analog call setup. The principle difference is that once the data call is passed to the data terminal, the digital voice terminal is returned to useful service and can be used to setup additional data calls or establish and receive voice calls.

With DCP Voice Terminals

There are two methods of establishing a digital data call with a DCP voice terminal.

- Voice terminal users that do not have a DATA button can use the Transfer feature.
- Voice terminal users with a DATA button use the DATA button to set up calls for the assigned data endpoint(s). They can also use the Transfer feature to access other data endpoints.

Using the Transfer Feature

Voice terminal dialing and the Transfer feature can be used to originate a data call for another user or a data end point without an associated voice terminal. The call can be originated by the transferring user or by one of the data end points to be connected. The transferring user uses the same steps that would be used to transfer a voice call. The call can be between two on-premises data end points or between a local (on-premises) data end point and a remote (off-premises) data end point.

Using an Assigned Data Button

Voice terminals with a DATA button can establish a data call for the associated data terminal by simply dialing the data extension or access code from the voice terminal and pressing the DATA button when the called data endpoint responds. A login prompt appears on the data terminal's CRT.

The Data Button

Any administrable feature button on a 7400D (Digital) series voice terminal can be assigned as a DATA button. The DATA button controls the following data communications functions for the associated data end point.

- Data preindication
- One button transfer
- Return-to-voice.

Multiple DATA buttons can be assigned to a single DCP voice terminal. This voice terminal can then set up data calls for all of the associated data terminals. This can be useful for operations such as centralized data entry points or FAX centers. Also, a single data endpoint can be associate with DATA buttons on several different voice terminals.

Data Button Functions

Data Preindication

Data preindication notifies the switch that a data call is being set up. This allows the switch to reserve the associated data module, and if needed, to reserve a Modem Pooling conversion resource before the outgoing data call is dialed. If the conversion resource is not available, the call is denied before completion and prevents toll charges from being incurred. If data preindication is not used, the call is completed to the remote data endpoint, and toll charges are assessed. Without a conversion resource, the data call may fail when the switch attempts to transfer the connection to a data endpoint.

Preindication is important in a DCS (Distributed Communications System). With a DCS, it is not always apparent to the user when a conversion resource is needed. The dialed data endpoint may or may not be located on-premises, and trunk facilities used for calls between nodes may or may not be capable of supporting end-to-end digital connections. By preindicating the call, a conversion resource is reserved in case it is needed. If not needed, the conversion resource is released by the switch. With Keyboard Dialing (discussed later), preindication is automatic.

One Button Transfer

One button transfer is the basic procedure for establishing the data connection. It allows the user to transfer a call to the associated data module simply by pressing the DATA button.

Return-to-Voice

Return-to-voice is a procedure that can be used to terminate a data call. By going off-hook on the voice extension and pressing the DATA button, the data call is transferred back to the voice extension. The user then has three options:

a. Tear down the data connection by going on-hook.

b. Establish a new data connection to a different data endpoint by pressing a different DATA button.

c. Establish a new data connection to a different data endpoint using the Transfer feature.

With a BRI Voice Terminal

There is one method of setting up a data call from an ISDN—BRI voice terminal. Unlike the DCP terminals, the Data button on a BRI voice terminal is the ***call appearance button*** for the associated data terminal.

When placing a data call from a BRI voice terminal, the data call appearance button is selected and the voice terminal is used to dial the appropriate address digits. Because the call appearance is used, no transfer to a data line is needed. Voice call appearances are not effected. More detailed information is provided in the ***ISDN—BRI feature*** chapter of this manual.

Keyboard Dialing

Keyboard Dialing allows a user to set up and break down data calls directly from the data terminal. This includes the ability to send on-hook/off-hook signals to the switch and to dial extension numbers. Data Preindication is automatically provided for DCP terminals when Keyboard Dialing is used to setup a data call. The user can place on- and off-premises public and private network calls where such connections are available.

Keyboard Dialing With DCP Terminals

Terminal Requirements

Keyboard Dialing requires the use of terminals that provide ASCII (American Standard Code for Information Interchange), 10-bit start/stop signaling. Data rates from 110 bps to 64 Kbps can be used. When Modem Pooling is involved, the data rates available are determined by the Modem Pooling conversion resource. Calls are originated by pressing the BREAK key on the keyboard or the ORIGINATE/DISCONNECT button on the associated data module.

System Messages

Standard system messages are displayed on the data terminal CRT. These messages tell the user the call's state and expected responses (*see* Call Progress Monitoring and Control). The user may need to respond to multiple requests for dialing (for example, with the ISN Interface feature or when an access or authorization code is used to restrict access to a trunk facility or data endpoint).

In Process Corrections

While entering digits, errors can be corrected using editing functions. Users wishing to transmit at extra low speeds (for example 110 bps) must dial at 300 bps or more and then switch to the low speed when dialing has been completed. If a printer is involved in the data call setup, it prints everything that appears on the CRT.

Modem Pool Identification

When keyboard dialing is used, one of the call progress messages gives the user information that can be used to identify the Modem Pooling conversion resource that has been reserved for the call. This is the "RINGING XXX XXX" message. The two numbers (XXX XXX) represent the trunk-group number and the trunk number (for the digital member of the conversion resource assigned to the call) within that group. These numbers are used to obtain the dial access code used with the Trunk Verification—Voice Terminal feature to check trunk and conversion resource condition *(see* the Trunk Verification—Voice Terminal feature for these procedures).

Queue Position Feedback

If the data call is placed in a queue, the call's place in queue is displayed on the CRT, and this display can be updated as the queue is served. The display is updated by typing "q" followed by a carriage return.

Receiving Calls

The data terminal can also be used to answer incoming data calls. The message INCOMING CALL is displayed on the CRT. When a DTDM is used, the call is answered automatically. For MPDMs and MTDMs, auto-answer is an administrable option.

Call Progress Monitoring and Control

Call progress monitoring and control is an essential element to data terminal dialing. This attribute provides the data terminal user with the ability to receive and respond to call progress messages while a call is being placed. Call Progress Monitoring and Control is essential when placing a call that requires responses to switch signals, such as network or remote access calls or while on a call that has been placed in a queue.

Call progress monitoring and control permits the user to respond to access code or authorization code requests. The system generates these requests when needed circuits are under access control. This also permits the data terminal dialing user to enter feature access codes and respond to second dial tone states. Table 42-A lists some of the messages provided to the terminal dialing user.

The switch also alerts the user to special states or conditions, as when a data terminal has Call Forwarding—Follow Me active. In this case, when a call is forwarded the display message `"FORWARDED"` and the ASCII *Bell Ring* character are sent to the terminal to alert the user that Call Forwarding is active.

A Keyboard Dialing string can contain up to 31 characters (for example, in response to the system messages). Cosmetic characters such as spaces, dashes, and parentheses are permitted. Special control characters, such as the dial tone delimiter "+" and pause "%," are available to facilitate Keyboard Dialing through networks and multiple switching nodes.

Table 42-A. Message Set for Data Call Setup Terminal Dialing

Message Format	Application	Meaning
DIAL:	Placing Call	Enter extension number or feature dial access code.
TRY AGAIN	Placing Call	Reorder tone is returned.
BUSY	Placing Call	Busy tone is returned.
DENIED	Placing Call	Intercept tone is returned.
DENIED TRY AGAIN	Placing Call	Busy tone is returned.
CONFIRMED	Activating a feature	Confirmation tone returned.
RINGING XXX XXX	Placing Call	Ringing tone is returned. The numbers (XXX XXX) show the trunk group and trunk numbers of the Modem Pooling conversion resource.
CALLBACK	Receiving a callback call	A previously placed Auto Callback call is ringing.
ABANDONED	Receiving a call	The calling party has gone on-hook.
PLEASE ANS-	Receiving a call	Equivalent to ringing, or one button transfer is in use.
FORWARDED	Call Forwarding	A call to the terminal has been forwarded.
DISCONNECTED-TRANSFER	Return to Voice	A call has been terminated using Return to Voice.
INCOMING CALL-	Receiving an incoming call	Equivalent to ringing.
DISCONNECTED-OTHER END	Call termination	A call has been terminated; other terminal has gone on-hook.
ANSWERED	Placing Call	Called terminal has gone off-hook.
CHECK OPTIONS	Placing Call	Handshaking failed between data modules.
LIST:	Abbreviated Dialing	Request for list number to be programmed.
INDEX:	Abbreviated Dialing	Request for list index number to be programmed.
ENTRY:	Abbreviated Dialing	Request for digit string.
STORED	Abbreviated Dialing	Confirmation: number has been entered into list.
WAIT,	Placing Call	Call setup delay.
WAIT, ## IN QUEUE	Placing Call	Call position in queue.
PROCEEDING	Placing Call	Call in queue is being served.
CODE:	Placing Call	Recall dial tone received. Authorization code requested.

The form of these messages may vary depending on the method and terminal used. For example, with the AT&T 5620 terminal, if the BRK key is used to start dialing or end a call, the messages displayed are " **D** " rather than " **DIAL:** " or " **DISCONNECTED:** ."

Forms of Keyboard Dialing

Four forms of keyboard dialing are available. These are basic keyboard dialing, default keyboard dialing data hot line service, and mnemonic dialing.

Basic Keyboard Dialing

Basic keyboard dialing can be used on any switch that allows keyboard dialing. With basic keyboard dialing, the user simply enters the dialing string (numeric and special characters) on the data terminal keyboard. Entries are in response to prompt messages that appear on the data terminal screen.

Default Terminal Dialing

Default terminal dialing places a data call to a predesignated data end point by simply entering a "null" character. For the purposes of default terminal dialing, a null charater is a space followed by a carriage return, or simply a carriage return in response to the dial prompt. For a given data module, the default destination address is administered on the switch in:

- Procedure 059, Word 1 and 2, for System 85, Release 2, Version 3
- Procedure 059, Word 4, for System 85, Release 2, Version 4, and DEFINITY Generic 2.

The maximum length of a default dialing number is 20 characters. When default dialing is used, the switch retrieves this number and uses it to place the call. Default terminal dialing is especially useful when a clear majority of calls placed from a given terminal are placed to a single data end point (for example, an assigned host computer).

Data Hot Line Service

A data module, associated with a data terminal, and administered for ***default dialing,*** can be assigned as a Hot Line station on Version 4 switches. This is done using Procedure 000, Word 3. Once assigned as a data Hot Line, the user simply presses the BREAK key or the Originate / Disconnect button on the data module and the default number is automatically dialed by the switch.

Mnemonic Dialing

Mnemonic dialing is a form of keyboard dialing that allows the user to enter an ***alphanumeric string*** rather than a strict numeric to place a data call.

The principle advantage to mnemonic dialing is that for most users, names are easier to remember than numbers For example, a user who frequently accesses several different host computers can use the name of each host as a mnemonic dialing entry rather than having to remember and associate separate numbers with each host name. Mnemonic dialing can be used to access local extensions, data terminals on remote private network nodes, or data terminals on the public network (including ISDN).

Mnemonic dialing is much like Abbreviated Dialing with the system list. There is one mnemonic dialing list that is maintained by the switch administrator and is available on a system-wide basis. A maximum of 1000 mnemonic dialing "names" can be stored on the mnemonic dialing list (300 in Release 2, Version 3). Mnemonic dialing names and their associated numbers must conform to the following

- A mnemonic dialing entry can be up to ten characters in length.
- The first character must be an alpha character (letter).
- A distinction is made between cases (upper and lower) for alpha characters. That is, "george" and "George" are treated by the switch as two separate and distinct mnemonic names.
- Each mnemonic entry represents a number that can be up to 20 characters in length.
- Numbers associated with mnemonic dialing names can contain the same special function characters used with Abbreviated Dialing except for the Suppress Character. The suppress function is not needed with keyboard dialing. The following special functions are available:
  - Pause
  - Wait
  - Mark
  - Wait for dial tone
  - Manual digit entry (a specified number of digits, up to 15).

Keyboard Dialing With ISDN—BRI

Data terminals using ISDN—BRI can also use keyboard dialing. The same functionality available to DCP data terminals is also available to BRI data terminals. The actual operations of the *Basic Keyboard Dialing* form is somewhat different for BRI data terminals. With BRI, a special Command Mode is used. The command mode and its specific use for Data Call Setup are described in detail in the User Operations section of this chapter and in the ***ISDN—BRI Feature*** chapter of this manual.

Host Computer Connectivity

As with data terminals, several methods can be used to connect host computers to the switch. These include analog connections, digital end-to-end arrangements, and a combination of analog and digital interfaces. Each host computer interface feature is discussed in detail in its own chapter of this manual. They are briefly discussed here for continuity purposes.

Analog Host Computer Interface Features

Host computer interface ports can be administered as individual line appearances. These can be provided with modems or ACUs (Automatic Calling Units) depending on their intended use. When this is done, each port is view by the switch as a separate data terminal. Calls to or from these ports work exactly like calls to or from a data terminal.

The System 85 and DEFINITY Generic 2 switches offer one analog computer interface feature that is specifically designed for use with a host computer. That is the Data Communications Access feature.

Digital Host Computer Interface Features

Specific digital host computer interface features available on the either the System 85 or DEFINITY Generic 2 switches include the following:

- Host Computer Access feature
- DMI* (Digital Multiplexed Interface) feature
  - — BOS (Bit-Oriented Signaling)
  - — MOS (Message-Oriented Signaling).

Each of these host interface features is described in detail in its own chapter of this manual.

Analog and Digital Interfaces on the Same System

The same switch can support both analog and digital interfaced data end points at the same time.

Separate Hosts

Separate and distinct analog data systems and digital systems can be supported. However, the two systems are not directly compatible with one another. Figure 42-4 shows an arrangement that supports both analog and digital data calling (to separate host computers). This arrangement shows both an analog interfaced host computer and a digital interfaced host computer.

Dual Interfaced Host Computer

It is also possible for a single host computer to use both analog and digital interface arrangements at the same time. This is represented in Figure 42-4 by the dotted box. With this arrangement, the same host uses both the DCA (Data Communications Access) feature (analog) and the HCA (Host Computer Access) feature (digital) for system interfaces (on separate ports).

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* The DMI feature was originally available in only one version. The DMI BOS and DMI MOS are System 85, Release 2, Version 4 enhancements of the DMI feature. Both BOS and MOS differ from the original version which used the DS1 AVD format.

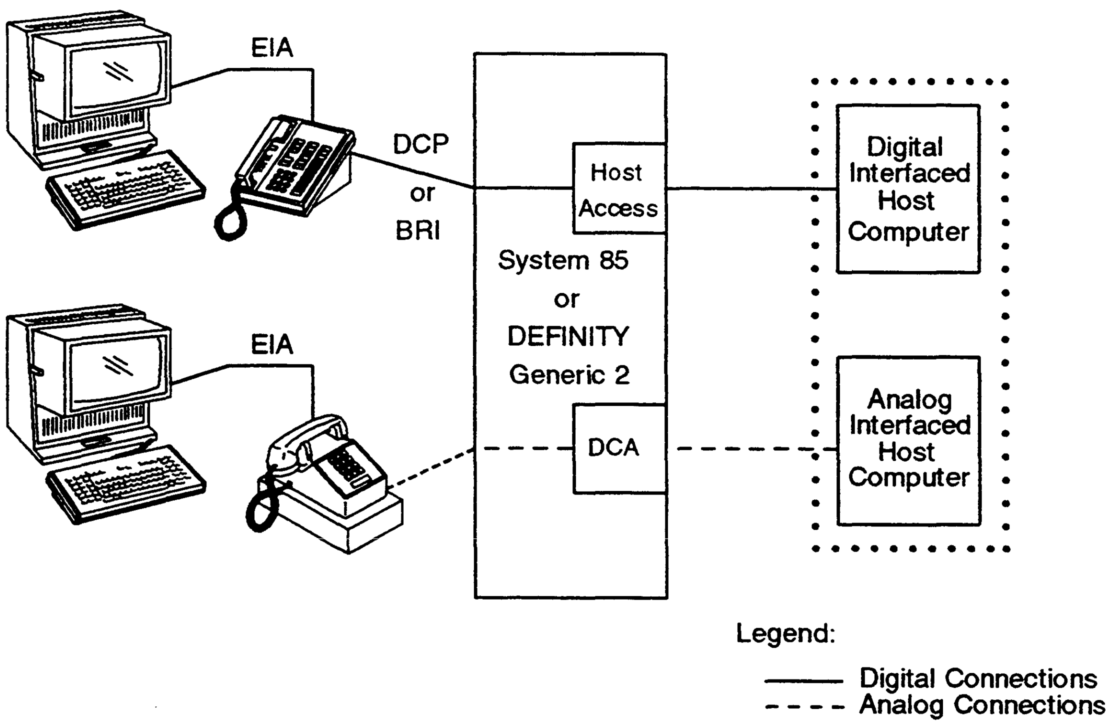

Figure 42-4. Data Call Setup With Dual Interface Arrangements

Other Analog to Digital Connections

It is possible for analog and digital interfaced end points to communicate with each other in both the System 85 and the DEFINITY Generic 2 environment. Figures 42–5 and 42-6 show this being done with off-premises connections. The Modem Pooling feature makes this connectivity possible.

Modem Pooling: The Modem Pooling feature provides the ability for both analog-interfaced and digital-interfaced facilities to be used effectively on the same switch. In simple terms, the Modem Pooling feature provides the switch with the ability to automatically insert a ***conversion resource*** into a connection when needed. The conversion resource converts data signals between analog and digital data signal formats.

It is important to note that the Modem Pooling feature is associated with off-premises data calling. That is, it supports calls from local stations to remote data end points or from remote stations to local data end points. It does not support calls between local analog interfaced facilities and local digital interfaced facilities. In order for this type of call to be supported, one end of the connection must appear to the switch to be a remote facility. This could be achieved by the initiating station placing a call to the local CO (Central Office) and then coming back into the switch via a separate DID CO trunk or a remote access trunk. As with the other supporting data features, the Modem Pooling feature is described in detail in its own chapter of this manual.

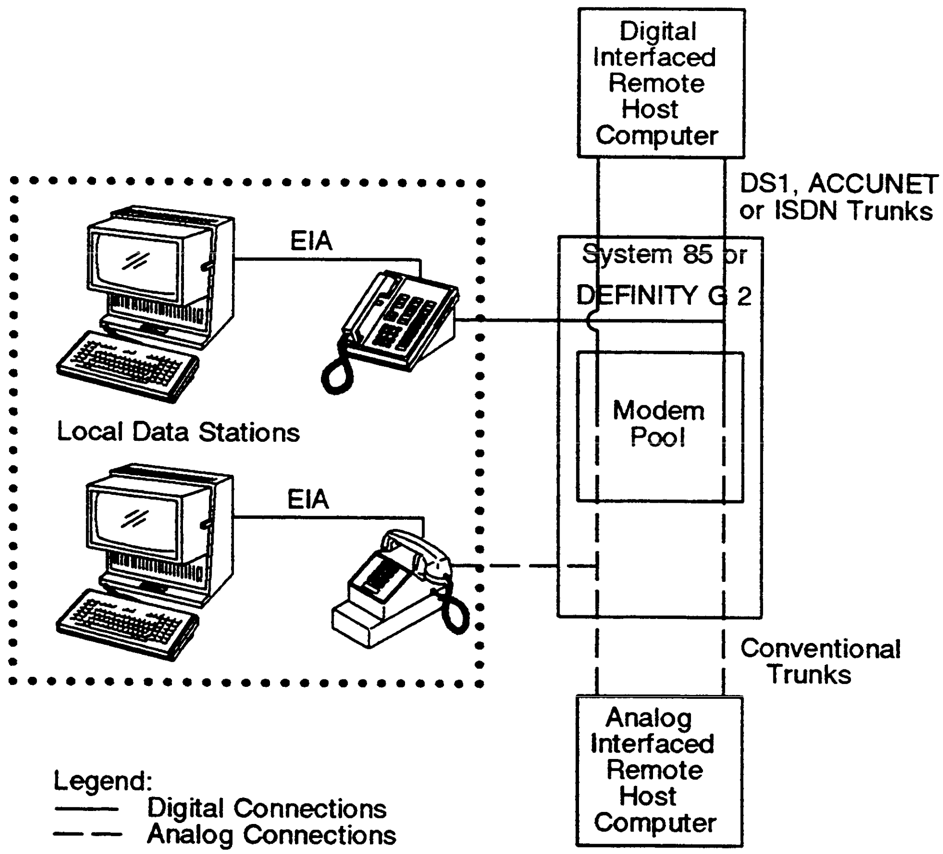

Figure 42-5. Off-Premises Data Calls From Local Stations to Remote Hosts

Host Computer Dialing

It is possible for host computers to dial data endpoints. Remote computers must be provided with the proper equipment to access a local host computer. Local hosts, interfaced to the switch through EIA ports can use an automatic calling unit and modem to dial other data end points. Hosts interfaced to switch via MPDMs and MTDMs can originate data calls when the data modules are equipped with the terminal dialing option. This option allows the data module to recognize dialing messages. The off-hook / on-hook and dialing instructions are generated from software provided by the customer. Detailed information on this (host computer dialing) option is given in Appendix D.

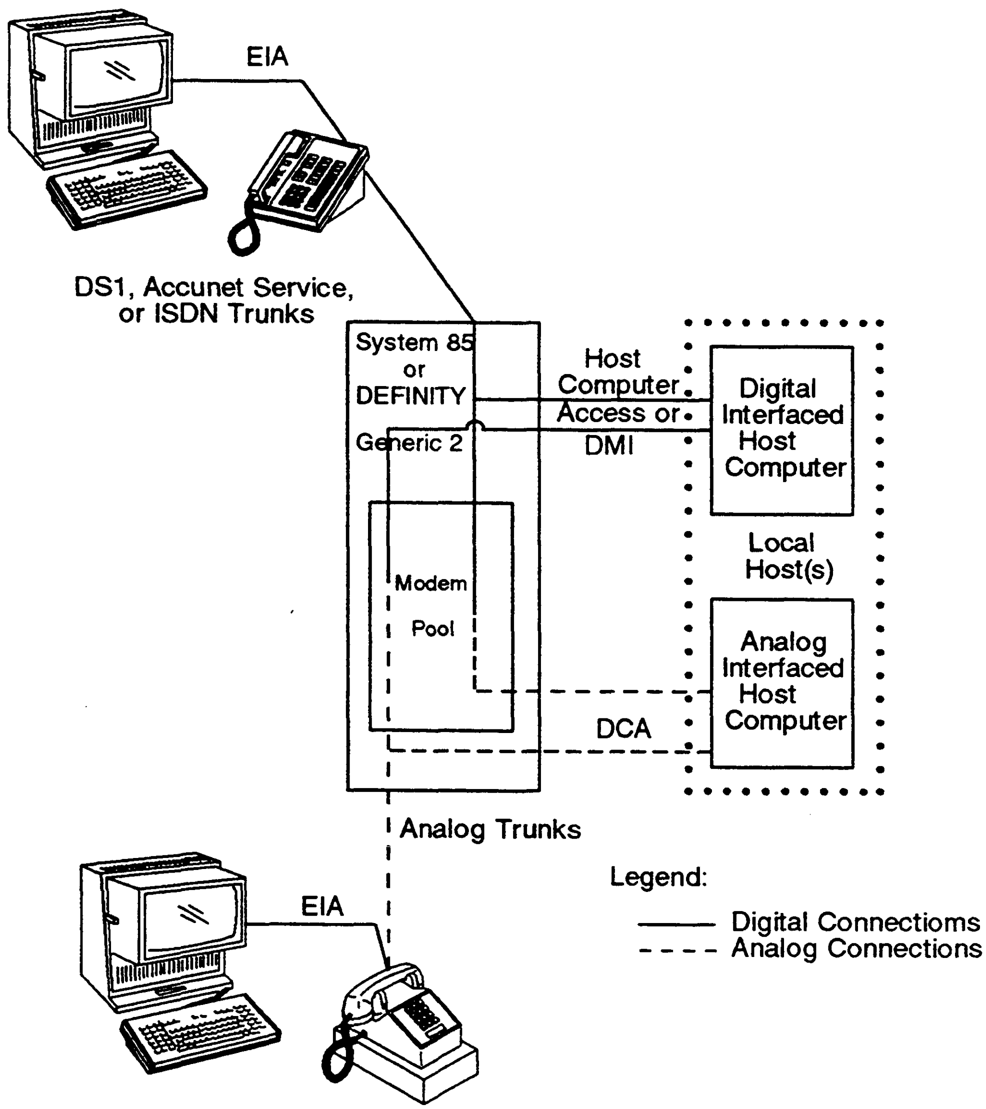

Figure 42-6. Off-Premises Data Calls From Remote Stations to Local Hosts

Feature History and Development

- The Data Call Setup feature was first available on System 85 in Release 1.
- The Mnemonic Dialing and Default Terminal Dialing capabilities were added in System 85, Release 2, Version 3.
- The Mnemonic Dialing capability was enhanced to provide up to 1000 mnemonic names in System 85, Release 2, Version 4 (from 300 in Release 2, Version 3).
- The ISDN—BRI featuree, introduced in DEFINITY Generic 2, provides a new data capable interface and family of terminals (7500 series) that can be used with the Data Call Setup feature.

User Operations

The following are the basic user operations for the Data Call Setup feature:

Analog Voice Terminals:

User operations for analog interfaced data terminals with voice terminal data call setup may vary depending on the type of modem used. The following procedures are based on use of the DATAPHONE® Modem 300/1200.

1. Go off-hook on the associated voice terminal. (Dial tone)
2. Dial the access code or extension number for the desired data end point. (Call-progress tones such as ringing tone)
3. When ready tone is returned, press the **[DATA]** button on the voice terminal to transfer control of the call to the data terminal (login prompt appears on the data terminal CRT).

DCP Voice Terminals

To place a call using the DATA button:

1. Go off-hook on the associated voice terminal. (Dial tone)
2. Dial the access code or extension number for the desired data end point. (Call-progress tones such as ringing tone)
3. Press the **[DATA]** button for the data terminal to be used. (Call progress messages, such as RINGING, ANSWERED, etc., appear on the data terminal CRT unless ready tone is received before the DATA button is pressed. When ready tone is received, a login prompt appears on the CRT.)

To place a call using the Transfer feature:

1. Go off-hook on the voice terminal (Dial tone).
2. Dial the access code or extension number for the desired data end point. (Call-progess tones such as ringing tone).
3. When ready tone is received, press the **[TRANSFER]** button. (Appearance lamp flashes.)
4. Dial the other data end point to be connected. (Call-progress tones such as ringing tone.)
5. Press the **[TRANSFER]** button again. (Connection with ringing tone is transferred to the originally dialed data terminal. Call progress messages appear on the data terminals CRT unless ready tone is received before the **[TRANSFER]** button is pressed. When ready tone is received, a login prompt appears on the CRT.)

For an incoming (off-premises) call, the data user and the assisting user should be connected via a normal voice call. To transfer the data user to a data endpoint, the assisting user puts the data user on soft hold and dials the data access code. When the assisting user receives ringback tone, they transfer the call with ringing tone to the data user by going on-hook. The data user then waits for a response from the data endpoint and performs the appropriate actions that will establish a connection.

To terminate a call setup using the DATA button:

1. Go off-hook on the associated voice terminal. (Dial tone.)
2. Press the **[DATA]** button associated with the data call to be terminated. (Control of the connection is transferred to the voice terminal appearance. Ready tone is heard on the voice terminal, and the message DISCONNECTED-TRANSFER appears on the data terminal CRT.)

ISDN—BRI Voice Terminals

To place a data call from a BRI voice terminal:

(Applies only to ISDN—BRI voice terminal equipped with the optional ADM—T terminal adapter.) The voice terminal (hand set) may be either off-hook or on-hook.

1. Make sure that data terminal is turned on and options are set correctly.
2. Press the **[DATA/SEND/OFF]** button. [The red status lamp lights, and dial tone is heard. On display capable terminals, `[DIAL:]` appears on the display.]
3. Dial the extension number or dial access code desired (a mnemonic cannot be entered on a voice terminal).
4. Press the **[DATA/SEND/OFF]** button again. [The red status lamp stays on and the green status lamp blinks. When the far end answers, the green status lamp goes to a steady on state and the red status lamp remains in the steady "on" state. The data terminal screen displays:

   `[CONNECTED - MODE 2]` and `[FAR END SPEED - 19200].]`

   NOTE: The figures for data mode (mode 2) and data rate (19200) shown in these examples are used for example purposes only. The mode and rate that will appear on your display will reflect the actual state of the far end of your connection and may differ from the examples shown.

5. Press **[RETURN]** on the data terminal keyboard and proceed with appropriate log on procedures.

To end a data call from a BRI voice terminal:

Press the **[DATA/SEND/OFF]** button. [Both the red and green status lamps go dark.]

Keyboard Dialing

Basic Terminal Dialing From a DCP Data Terminal

— To place a call using basic keyboard dialing.

1. Press the **[ORIGINATE/DISCONNECT]** button on the assigned data module,

   or

   Press the **[DISCONNECT BREAK]** key on the data terminal keyboard. (The message **DIAL:** appears on the CRT.)

2. Enter the dialing string for the desired data end point on the keyboard. (Dialing string can be a dial access code, extension number, or abbreviated dialing entry.)

3. Press the **[RETURN]** key. (Call progress messages such as **RINGING, ANSWERED,** and a login prompt appear on the CRT.)

— To answer an incoming call:

NOTE: The data terminal must be on and in an idle (disconnected) state. Otherwise, the call must be answered manually. This is done by pressing the keyboard BREAK key or the data module ORIGINATE/DISCONNECT button. The terminal bell then rings to alert the user of the incoming call.

1. Press the **[BREAK]** key (BREAK key must be held down for 2 or more seconds) if long-space disconnect is active,

   or

   Press the **[ORIGINATE DISCONNECT]** button on the data module.

— To place a call using Default Terminal Dialing

1. Press the **[ORIGINATE DISCONNECT]** button on the data module,

   or

   Press the **[BREAK]** key on the terminal keyboard. (System returns DIAL: prompt.)

2. Press the **[RETURN]** key or space bar followed by the **[RETURN]** key (call progress messages and login prompt as appropriate).

— To place a call using Mnemonic Dialing

1. Press the **[ORIGINATE DISCONNECT]** button on the data module,

   or

   Press the **[BREAK]** key on the terminal keyboard. (System returns **DIAL:** prompt.)

2. Enter the mnemonic string for the destination desired.
3. Press **[RETURN]** key (call progress messages and login prompt as appropriate).

Basic Terminal Dialing From a BRI Data Terminal

— To place a data call from the BRI data terminal:

1. Make sum the data terminal is turned on and options are set correctly.

   If the `[CMD:]` prompt does not appear on the screen, press the **[BREAK]** or **[RETURN]** key and then enter "AT" to make certain that speed and parity are correct.

2. Enter "dial" (or "d"), followed by a space and the number to be called (extension number, mnemonic, access code, etc.)
3. Press the **[RETURN]** key. [Call progress messages, `[CALLING aaa]`, `[Type E to end call:]`, `[RINGING]`, `[NNN NNN]`, `[CONNECTED - MODE 2]`, `[FAR END SPEED - 19200]` are displayed.]
4. Press the **[RETURN]** key. [Far end login prompt or other response is retuned.]

— To end a data call from the data terminal:

1. Enter the attention sequence "+++" [Terminal returns to command mode and
   `[Call Status: Data Call Active]`, `[Type H for help.]` and `[CMD:]`

   appear on the screen.]
2. Enter "end" or "e." [The call progress messages `[CLEARING]`, `[ENDED]`, `[Call Status: Idle]`, `[Type H for help.]` appear on the screen.]

or

You can log off from the far end (host etc.). This will cause the distant end to terminate the call. [The call progress messages [CLEARING] , [ENDED] , [FAR END REQUESTED] , [Call Status: Idle] , [Type H for help.] appear on the screen.]

NOTE: To end a call during call setup (before [CONNECT] is displayed), enter "end."

Host Computer Dialing

Host computer dialing can use any of the keyboard dialing processes. This method of data call setup is loaded into the host computer software and is subsequently not apparent to the user.

Specific software steps are dependent on the host computer being used. Detailed information needed to set up a Host Computer Dialing Program is contained in Appendix D: Data Communications.

Considerations

Hard and Soft Processor Swaps

The contents of the mnemonic dialing list are stored in a translation portion of switch memory. Therefore, this list will endure a hard processor swap.

Each data terminal's default dialing number is stored in a translation portion of switch memory. Therefore, these numbers will endure a hard processor swap.

Stable calls placed using the Data Call Setup feature will endure a hard swap. However, Data Call Setup cannot be used to place a call during a hard swap.

The Data Call Setup feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

Using keyboard dialing, Abbreviated Dialing can be used to originate data calls. In switches prior to Release 2, Version 4, Abbreviated Dialing lists were assigned based on extension numbers For these switches, data terminals could have their own Abbreviated Dialing lists.

In Release 2, Version 4, and in DEFINITY Generic 2, Abbreviated Dialing lists are assigned bawd on equipment locations. For these switches, data terminals share Abbreviated Dialing lists with their associated voice terminals.

ACCUNET Service Interface

The Voice Terminal Dialing portion of the Data Call Setup feature procedures is used for the ACCUNET Service Interface feature. Keyboard dialing as described in the Data Call Setup feature, does not work for ACCUNET Service calls. Also, ISDN—BRI terminals cannot access the ACCUNET Service Interface feature.

APLT (Advanced Private Line Termination)

Access to a remote data end point over private facilities is the same as that described for the appropriate network interface feature (such as APLT, AAR, or WCR features).

Authorization Code

The Authorization Code feature can be used for data calls as well as for voice calls. The feature works in the same way regardless of which type of call is being placed or whether keyboard dialing or voice terminal data call setup is being used.

There are some differences in the system response for keyboard dialing between Version 4 and earlier switches and DEFINITY Generic 2 switches. These differences involve the use of a universal module, and they appear in the form of the ***Call Progress Monitoring and Control*** messages as follows:

Call Progress Monitoring and Control Messages For Authorization Code Feature		
	Dial Tone	Recall Dial Tone
Version 4 and Earlier	DIAL:	CODE:
DEFINITY Generic 2 (Universal Module)	DIAL:	DIAL:

Also, in DEFINITY Generic 2, the ISDN—BRI data terminals are available for use with the Data Call Setup Feature. With ISDN—BRI, the interface generates ***Call Progress Monitoring and Control*** displays from the ISDN messages received at the terminal interface rather than displaying switch generated messages. These are a different set of messages than those seen at a DCP terminal. See the ISDN—BRI feature description for more details.

AAR (Automatic Alternate Routing)

For System 85 and DEFINITY Generic 2.1 switches, the AAR feature is used to improve the routing of data calls over private network facilities, just as it is for voice calls. However, without ISDN, routing pattern searches are different and routing pattern searches work differently for Release 2, Version 3 and earlier, Version 4, and DEFINITY Generic 2.1 switches.

System 85, Version 3 and Earlier

For Version 3 and earlier, data call routing patterns work in the same way as they do for voice calls.

System 85, Version 4

In Version 4, ISDN—PRI is introduced, and with that new feature the concept of ***Bearer Capability*** becomes a factor in selecting AAR routing patterns. Also, the class of service function is altered to indicate ISDN requirements as follows:

- ***ISDN Facilities Required:***

  These calls must be routed over ISDN facilities (PRI trunks). If ISDN facilities are not available, and queuing, authorization code, FRL raising and other alternatives having been exhausted, the call is denied.

- ***ISDN Facilites Preferred:***

  An attempt is made first to route this type of call over ISDN facilities. If no ISDN facilities can be found for the call the call is routed using any type of facility that will otherwise support the call.

- ***Any Available Facilities:***

  These calls do not require ISDN facilities and they are routed over the first available and accessible trunk group.

Bearer Capability Class

Bearer capability, the type of call a facility can support, is also a factor in call routing in Version 4. Five bearer capability classes are recognized:

BCC 0 Voice and Voice Grade Data (Modem Pooling required)

BCC 1 Mode 1 Data, 56 Kbps allowed

BCC 2 Mode 2 Data, 64 Kbps allowed

BCC 3 Mode 3 Data

BCC 4 Mode 0 Data.

Routing of data calls is based on a match. That is, if a data call requires mode 2, only preferences that support mode 2 are searched. If none are available, the call is denied.

DEFINITY Generic 2.1

With DEFINITY Generic 2.1, the routing pattern and preference search becomes more sophisticated. The same class of service requirements apply (ISDN Required, ISDN Preferred, or Any Facility), however, the bearer capability class is expanded (from five possibilities to 256 possibilities) into a ***BCCOS (Bearer Capability Class of Service)*** and is extended to all calling and call support facilities. This includes not only ISDN facilities, but DCP terminals, Modem Pooling conversion resources, Host Access ports, and AAR/ARS patterns.

The AAR pattern search checks the preferences in order to find a preference that will support the data call ***without requiring a Modem Pooling conversion resource.*** If such a preference is found, it is then checked for an available trunk. If an available trunk is found, the data call is routed using that trunk. If no available trunk is found, the next preference is checked. This is continued until all preferences have been checked.

If no route is found that does not require a Modem Pooling conversion resource, software checks for a route that will support the call ***with a Modem Pooling conversion resource.*** This continues until all preferences in the pattern have been checked for routing possibilities.

Queuing on AAR/ARS Routed Data Calls

On System 85 and DEFINITY Generic 2.1 switches, Queuing is allowed (including pattern queuing) for data calls being routed via the AAR or ARS features. Based on assigned criteria and the switch version used (FRL, BCCOS, and ***N*** for pattern queuing) a data call can queue on the best choice trunk group (or ***N*** trunk groups starting with the first accessible) that will support the call.

Automatic Callback

Automatic Callback can be used by both voice terminal and data terminal users. However, an Automatic Callback call to a data terminal must be answered using the manual answering procedure.

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, the ARS feature can be used to improve the routing of data calls as well as voice calls. However, like the AAR feature, if ISDN is not used, separate routing patterns must be used to prevent voice calls from terminating on data only extensions and vice versa. Differences between switches are the same as for the AAR feature discussed earlier.

Bearer Capability

The Bearer Capability feature is first available with DEFINITY Generic 2, although the principle of bearer capability was applied in System 85, Release 2, Version 4, with the ISDN—PRI feature. Bearer Capability identifies specific facilities that are compatible (and incompatible) with specific calls. Bearer Capability also provides instructions on call support requirements such as Modem Pooling for specific calls.

For DEFINITY Generic 2, the Bearer Capability feature supports, and is essential to the successful operation of the Data Call Setup feature. The Bearer Capability feature does not apply to switches prior to DEFINITY Generic 2.

Call Coverage

Call Coverage is blocked for a data module. That is, if a data module is placed in a coverage path, it will be skipped in the search for an available coverage point. If a data module calls an extension that has Call Coverage active, the call does not cover. Rather, it stays at the terminal called.

Call Forwarding—Busy and Don't Answer

This feature can be used by voice terminal and data terminal users. It is recommended in lieu of Call Coverage.

Call Forwarding—Follow Me

This feature can be used by voice terminal and data terminal users. When Call Forwarding—Follow Me is in effect for a data terminal, the display message "FORWARDED" and the ASCII bell ring character are sent to the forwarding terminal each time a call is forwarded.

Call Pickup

While a data terminal (or more precisely, its extension number) can be made a member of a call pickup group, Call Pickup is blocked for data terminals. They cannot be used to pickup a call, nor can a call directed to a data terminal be picked up by another member of the pickup group.

Data Protection

Data Protection should be used for all data calls to prevent disruption by bridging and override features. Data endpoints that are used for data communications exclusively can be assigned Data Protection—Permanent.

DID (Direct Inward Dialing)

On-premises host computers can be accessed from off-premises terminals using DID facilities in the same way a DID voice call is placed. By dialing the extension number assigned for accessing the host computer (this may be via extension number steering). The switch decides if a conversion resource is required and inserts the resource into the path of the data call (*see* Modem Pooling).

DCS (Distributed Communications System)

In a DCS environment, the Transfer feature should not be used to set up data calls unless all tie trunks involved are DS1 AVD (Alternate Voice Data). Data calls between data endpoints on separate nodes of a DCS arrangement require a modem pooling conversion resource unless DS1 AVD tie trunks are used. A call transferred between nodes using the Transfer feature does not receive a Modem Pooling conversion resource.

In the DCS environment, it is possible (perhaps even likely) that a caller will not know what switching node supports a particular data endpoint. The only way to assure that the call will be allowed is to use preindication and one button transfer or data terminal dialing where preindication is automatic.

For local (on-premises) calls, data rates up to 64 Kbps are supported. On internode calls not using DS1 AVD tie trunks, a Modem Pooling conversion resource is needed. In this case, the data rates are limited by the modems available at both switches. Generally, modems are limited to a maximum data rate of 4800 bps with 300 to 1200 bps being the most common (and economical).

Hot Line

Default Dialing

The default dialing capability of the Data Call Setup feature is used to perform the dialing function for digital data Hot Line stations. Data modules used for Hot Line stations must be assigned default dialing. The default dialing number assigned to the data module becomes the Hot Line destination number when the data module is administered as a Hot Line station.

If a DATA button is assigned for a digital data Hot Line station, the DATA button can be used to bypass the Hot Line designation. This is done by setting up a data call with the voice terminal and then using one button transfer to connect the data terminal to the call.

One Button Transfer

The one button transfer function of the Data Call Setup feature can be used to override the Hot Line assignment for a Hot Line data station and complete a call to a destination other than the Hot Line destination for that station.

ISDN—BRI (Basic Rate Interface)

The Data Call Setup feature works in basically the same way for BRI terminals as it does for DCP terminals. The following exceptions, however, are worth noting:

- Voice Terminal Data Button

  BRI voice terminals do not have a DATA button in the same sense as DCP terminals. As a result, the data button functions: One Button Transfer, Return to Voice, and Data Preindication, do not apply to BRI terminals.

  Data calls can be originated by the voice terminal for an ISDN—BRI voice / data station with an associated data terminal (ADM-T interface). The basic differences are in software and call processing. Except for the ***data button functions*** mentioned above these differences are not readily apparent to the user.

- Data Terminal operations

  ISDN—BRI data terminals begin operations in a ***local command mode*** which is noticeably different from DCP data terminal operations. For specific information on this data terminal command mode, see the ISDN—BRI feature chapter in this reference manual.

ISDN—PRI (Primary Rate Interface)

The data call setup feature is used to establish data call connections to and from ISDN—PRI trunking facilities. Access for a given terminal is based on COS. The ***interworking*** function for DCP terminal users makes the use of ISDN—PRI facilities essentially transparent to the user. The only noticeable difference may occur if the Display — Voice Terminal feature is being used in conjunction with Data Call Setup, and the far end of the ISDN connection provides called terminal name or number display information.

ISN (Information System Network) Interface

The Data Call Setup feature is used with the ISN Interface feature from the System 85 or DEFINITY Generic 2 side of the interface. This allows switch side data terminal users to place data calls to data end points on a local ISN. ISN station users can also place calls to data end points on the local switch and access other network connections through the System 85 or DEFINITY Generic 2 switch. Calls from the switch side require two stage dialing. That is, the user must first dial an access code or extension number to reach the ISN Interface port, receive a second dial prompt from the ISN Network Controller, and then dial the desired ISN data end point.

Last Number Dialed

The Last Number Dialed feature does not redial keyboard dialing sequences because these sequences use Abbreviated Dialing to outpulse the digits.

Modem Pooling

The Modem Pooling feature works with the Data Call Setup feature to support off-premises data connections. Either incoming or outgoing calls are supported. Modem Pooling provides analog-to-digital and digital-to-analog signal conversions when required.

Multiple Listed Directory Numbers

Attendant seeking incoming calls cannot connect directly to a DCP (Digital Communications Protocol) interfaced data end point. To complete such a call requiring a conversion resource, the attendant must extend the call to a multiappearance voice terminal that has a Data button assigned for the desired data end point. The multiappearance voice terminal can then transfer the call to the data end point using one button transfer procedures.

Precedence Calling

Incoming Precedence Calling calls cannot preempt calls connected using the Data Call Setup feature. Precedence calls directed to data call setup connections are routed to the attendant.

Queuing

Either ringback or off-hook queuing is available for voice terminal data call setup using one button transfer. However, when keyboard dialing is used, only off-hook queuing is available. Going on-hook after queuing on a data call originated using keyboard dialing results in disconnect. Also, attempting to transfer an on-hook queued call to a data terminal before the call is completed will result in a disconnect.

Remote Access

Remote access trunks can be used to access data extensions on the System 85 or DEFINITY Generic 2 switch. Touch-tone dialing must be used to access the data extensions directly. Rotary trunks can access the attendant, who can then transfer the call to the data extension. Modem Pooling cannot be used in this last case, since the voice terminal Transfer feature is used.

Restriction—Attendant Control of Voice Terminals

Restrictions, such as Outward Restriction, which would prevent a data extension from dialing an outside data endpoint can be applied by the attendant. If the data extension has an associated DATA button appearance on a voice terminal, the voice terminal could originate the data call and use the one button data call transfer procedure to establish the call, unless the same restrictions are applied to the activating voice terminal.

Tenant Services

Data Call Setup is compatible with the Tenant Services feature. In general, Data Call Setup works within a partitioned switch in much the same way as voice terminal dialing.

Keyboard Dialing

Keyboard dialing is a partitioned function of the Data Call Setup feature. A data terminal user (in a partition other than Extension Partition 0) can use keyboard dialing to access data trunk groups that are dedicated to or shared with the partition. If the data terminal user tries to access any other data trunk group, the switch will return Intercept Treatment to the calling party.

A data terminal user in Extension Partition 0 can use keyboard dialing to access any data trunk group in the switch.

Mnemonic Dialing

The mnemonic dialing list, assigned in Procedure 013 Words 1 and 2, is a system-wide resource. These 1000 names are shared by every partition in a partitioned switch.

Mnemonic dialing is a partitioned function of the Data Call Setup feature. A data terminal user (in a partition other than Extension Partition 0) can use mnemonic dialing to access data trunk groups that are dedicated to or shared with the partition. If the data terminal user tries to access any other data trunk group, the switch will return Intercept Treatment to the calling party. (This Intercept Treatment is returned after a disallowed name has been converted to digits that are outpulsed by the switch.)

A data terminal user in Extension Partition 0 can use mnemonic dialing to access any data trunk group in the switch.

Default Dialing

Default dialing numbers are not checked for legality at the time they are assigned, but calls to these disallowed numbers are denied when the digits are outpulsed by the switch. When a data terminal user attempts to use default dialing to place a call over another partition's trunk group, the switch returns intercept treatment to the calling party.

A data terminal user in Extension Partition 0 can use default dialing to access any data trunk group in the switch.

Transfer

The Transfer feature can be used to set up data calls in lieu of a DATA button. This method does not include a Modem Pooling conversion resource, however. If the call requres the resource, the call attempt fails. See also the DCS interaction. Transfer disallows the use of Data Preindication.

Trunk Verification—Voice Terminal

The keyboard dialing form of Data Call Setup is used with the Trunk Verification—Voice Terminal feature to identify Modem Pooling trunk groups and trunks that need to be checked.

The call progress message "RINGING XXX XXX" identifies the trunk group and trunk of the digital member of the Modem Pooling conversion resource assigned. This information can be used to obtain the dial access code used with the Trunk Verification—Voice Terminal feature. Note that the Trunk Verification—Attendant feature cannot be used to check Modem Pooling trunks.

WCR (World Class Routing)

The Data Call Setup feature works with the World Class Routing feature in the same way that it did with the earlier networking features, AAR and ARS on DEFINITY Generic 2.1 switches. That is, data calls route over WCR networking facilities in much the same way as voice calls.

The availability of digital trunk facilities (DS1 and ISDN—PRI) provides call support capabilities that are of special advantage to data calling. This fact can be taken into consideration with the World Class Routing feature to a greater degree than was possible with the earlier networking features on Generic 2.1 switches. For example, with ARS, preference arrangement within patterns was typically on a least cost basis, and all calls to the same destination (both voice and data) were routed over the same routing pattern. Discrimination between preferences (for data calls as opposed to voice calls) could be made only on the basis of BCCOS (Bearer Capability Class of Service).

With World Class Routing, up to seven different routing networks can be available. One or more of these networks could be designed arround data calling requirements with pattern preferences arranged in an order based on data calling requirements specifically rather than general purpose considerations like least cost routing.

Restricting Feature Use

Restrictions that can be administered to voice terminal extensions can also be administered to data extensions. Restrictions can either be activated toward an extension by the attendant for temporary use or assigned in switch translations for more permanent use.

Attendant Control of Voice Terminals Restrictions

The attendant can activate the following restrictions using the Attendant Control of Voice Terminals feature:

- Terminal-to-Terminal Restriction
- Termination Restriction

- Total Restriction
- Outward Restriction.

Voice Terminal Restrictions

Fixed Voice Terminal Restrictions that can restrict the Data Call Setup feature include the following:

- Origination Restriction
- Termination Restriction
- Terminal-to-Terminal Only Restriction
- Inward
- Outward.

Hardware Requirements

The following special hardware items are required for the Data Call Setup feature:

For Traditional Modules:

- SN253C Auxiliary Tones Circuit Pack

  The auxiliary tones circuit pack is used with the Data Call Setup feature only when a Voice Terminal is used with the Transfer feature to set up a data call. When the called terminal is a data module, the SN253 produces the audible "ready tone" heard in the voice terminal receiver when the data module answers.

- SN255B Tone Detector Circuit

  The tone detector circuit identifies call-progress tones (including ready tone) and provides call-supervision signals. These signals support off-premises and keyboard (terminal) dialing.

- Data Modules

  The System 85 and DEFINITY Generic 2 switches use the DCP (Data Communications Protocol) protocol for internal digital transmissions and for the Data Call Setup feature. For DEFINITY Generic 2 switches, the ISDN—BRI protocol is also available. Data modules provide protocol conversion for equipment using an EIA protocol (or bisynchronous protocol) to DCP or ISDN—BRI for interface with the switch. The following data modules are available:

DCP Data Modules

— DTDM (Digital Terminal Data Module)

— MDMs (Modular Data Modules)*

— PDM (Processor Data Module)

— TDM (Trunk Data Module)

— 3270 (A, C, or T) Data Module

— 7400A Data Module

— 7400B Data Module.

For Universal Modules:

- TN748C Tone Detector circuit

  The tone detector circuit identifies call-progress tones (including ready tone) and provides call-supervision signals. These signals support off-premises and keyboard (terminal) dialing.

- TN768 Tone/Clock

  The tone/clock is used with the Data Call Setup feature only when a Voice Terminal is used with the Transfer feature to set up a data call. When the called terminal is a data module, the TN768 produces the audible "ready tone" heard in the voice terminal receiver when the data module answers.

- Data Modules

DCP Data Modules

For DCP terminals, the same data modules are used with the Universal Module as with the Traditional Module.

ISDN—BRI Data Modules

In addition to the DCP Data Modules, BRI Data Modules can be used with the Universal Module on DEFINITY Generic 2 switches. ISDN—BRI is available only on the Universal Module, therefor BRI Data Modules do not apply to traditional modules.

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* Data modules on earlier System 85s (Release 1 and Release 2, Version 1) were fixed units, either PDM or TDM. On System 85 Release 2, Versions 2, 3, and 4, and on DEFINITY Generic 2, data modules (other than the DTDM and 3270s) are modular and are configured as either MPDMs or MTDMs.

— ADM-T (Asynchronous Data Module - T Interface)

— 7500 Data Module.

Data Module Characteristics

While most data modules will work together in the same data mode, there are instances where specific combinations do not work or do not work well because of divergent characteristics such as bit inversion or incompatible handshaking requirements. Table 42–B lists some of the current data modules and their operating characteristics. This information can be used to identify data module combinations that will or will not work properly.

TABLE 42-B. Data Module Operating Characteristics

Data Module	DMI Mode	User Data Rate	Sync	Async	Bit Invert	Protocol Packaging	Handshake	Notes
7400 Series	2	to 19.2K	no	yes	yes	HDLC	mode 2	
DTDM	2	to 19.2K	yes	yes	yes	HDLC	mode2	
MPDM	0	64K	yes	no	yes	no	mode 2	
	1	56K	yes	no	no	DDS	mode 2	1
	2	to 19.2K	yes	yes	yes	HDLC	mode 2	
MPDM/M1*	0	64K	yes	no	yes	no	none	
	1	56K	yes	no	no	DDS	none	2
	2	to 19.2K	yes	yes	yes	HDLC	none	2
3270 A	2	to 9.6K	yes	yes	yes	HDLC	mode 3/2 adpt	3
	3	64K	yes	no	yes	LAPD	mode 3/2 adpt	4
3270 T	3	64K	yes	no	yes	LAPD	mode 3	4
PC Interface with ASCII	2	to 19.2K	no	yes	yes	HDLC	mode 3/2 adpt or mode 2	5
Terminal Emulation	3	64K	no	yes	yes	LAPD	mode 3/2 adpt	
PC Interface with 3270 Emulation	3	64K	yes	no	yes	LAPD	mode 3	5

(Continued)

TABLE 42-B. Data Module Operating Characteristics ***(Continued)***

Data Module	DMI Mode	User Data Rate	Sync	Async	Bit Invert	Protocol Packaging	Handshake	Notes
BRI Data Modules (7500 UDM-T and the ADM-T)	0	64K	yes	no	no	none	none	
	1	56K	yes	no	no	DDS	none	
	2	to 19.2K	yes	yes	yes	HDLC	mode 3/2 adpt	6
	3	64K	yes	no	yes	LAPD	mode 3/2 adpt	7 & 8
	3/2	64K	yes	no	yes	LAPD	mode 3/2 adpt	7 & 8
PC/ISDN Platform with 3270 Emulation	3	64 or 56K	yes	no	optional	LAPD	mode 3	9
PC/ISDN Platform with ASCII Terminal Emulation	2	to 19.2K	no	yes	yes	HDLC	mode 3/2 adpt or mode 2	10
	3	64 or 56K	yes	no	optional	LAPD	mode 3	10
AT&T ISDN Advantage	3	64K	yes	no	yes	LAPD Data Phase	mode 3	

NOTE 1: The mode 2 handshake will not work over other than 64 Kbps facilities, for example, robbed-bit facilities. Use the MPDM/M1* for mode 1 calls made over robbed-bit facilities.

NOTE 2: Use the MPDM/M1* for mode 0 and mode 1 calls when the far end DCE is not another AT&T data module (does not do a mode 2 handshake).

Although the MPDM/M1* will also suppress the handshake in mode 2, it is recommended this not be done because rate adaption will not be possible.

NOTE 3: Mode 3/2 adaptive means that a mode 3 handshake is attempted first. An algorithm is then followed to determine the far-end's data mode and either switch to mode 2 or continue in mode 3.

NOTE 4: Mode 3 data can only be circuit switched in Generic 2.1 and Generic 1.1. Also, mode 3 on the 3270 A and T requires a 3270 C on the far end.

NOTE 5: Mode 2 on the PC Interface is supported under the ASCII Terminal Emulation Package. The PC Interface in mode 2 uses a mode 3/2 adaptive handshake if the bit rate is set at 64 K If the rate is set at 19.2 K or lower, a mode 2 handshake is used. 3270 emulation on the PC Interface requires a 3270 C data module on the far end. Mode 3 operation is defined as synchronous when in 3270 emulation, otherwise mode 3 operation on the PC Interface is defined as asynchronous.

(Continued)

NOTE 6: Mode 2 on the BRI data modules is implemented in the incoming direction only. Outgoing calls requiring mode 2 speeds use mode 3/2 adaptive bearer capability.

NOTE 7: On outgoing mode 3 and 3/2 adaptive calls, BRI data modules always invert bits. On incoming mode 3 and mode 3/2 adaptive calls, the BRI data modules check the restriction bit in the low-layer compatibility IE and either invert or do not invert depending on the contents of the IE. This is not done for incoming mode 0 calls, however.

NOTE 8: The algorithm for the mode 3/2 handshake is different for DCP and BRI data modules. When called, BRI data modules start a mode 3 handshake. If a mode 3 or mode 2 handshake is received from the calling end within a specified number of seconds, the BRI data modules switch to that mode. If a mode 3 or mode 2 handshake is not received within that time, the BRI data modules switch to mode 2. If a mode 2 handshake is not received within 15 more seconds, the BRI data modules time out and the call is dropped.

NOTE 9: Options exist on the PC/ISDN Platform with 3270 Emulation to allow the user to choose either 56 kbps or 64 kbps and to choose to invert or not invert bits. The PC/ISDN Platform with 3270 Emulation requires a 3270 C data module on the far end.

NOTE 10: Options exist on the PC/ISDN Platform with ASCII Terminal Emulation to allow the user to choose either 56 kbps or 64 kbps and to choose to invert or not invert bits. Either mode 3/2 adaptive or mode 2 handshakes are used depending on the baud rate option setting. If the setting is 19.2 kbps or slower, a mode 2 handshake is used. Mode 3 operation is defined as synchronous when in 3270 emulation, otherwise mode 3 operation on the PC/ISDN Platform is defined as asynchronous.

For complete definitions of what constitutes the DMI modes 0, 1, 2, and 3, refer to *Digital Multiplexed Interface (DMI) Technical Specification,* Issue 3.1, August 1986, select code 500-029. Ask for the most recent version.

Feature Administration

The Data Call Setup feature is assigned on an extension COS (class of service) basis. The data preindication, one button data call transfer, and return-to-voice functions of the DATA button are assigned individually on a per button, per voice terminal basis. These DATA button functions are not available to ISDN—BRI terminals. Keyboard Dialing also known as terminal dialing, is assigned on a per data terminal or data module basis.

On System 85 switches, the Data Call Setup feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — DATA CALL SETUP			
PROCEDURE	WORD	PURPOSE	SMT
013	1	Assigns alphanumeric strings (names) for the mnemonic dialing function.	Yes
013	2	Assigns telephone numbers to the alphanumeric strings entered in Procedure 013, Word 1 for mnemonic dialing.	Yes
013	3	Displays the number of mnemonic dialing names that can still be added to the system list.	Yes
051	1	Assign a 72/74-Series terminal with data module and keyboard dialing.	Yes
052	1	Assign appearances for voice and data.	Yes
055	2	Assigns the DATA button to 72-Series or 74-Series voice terminals.	Yes
059	4	Assigns default dialing telephone number to a specific data module.	Yes
350	2	Assigns dial access code for terminal entry of default dialing (Abbreviated Dialing) list item.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — DATA CALL SETUP	
PATH NAME	PURPOSE
terminal-change system mnemonic-dial	Displays the number of mnemonic dialing names available. Also assigns new mnemonic names and associated telephone numbers or changes telephone numbers associated with existing names.
terminal-change terminal equipment	Assigns a 72/74 Series voice terminal with data module and terminal dialing. Also assigns a data module with terminal dialing.
terminal-change terminal buttons	Assigns appearance and feature buttons to a multi-function voice terminal.

The following are the applicable FM path names used with the AP 16.

FM SCREENS — DATA CALL SETUP	
PATH NAME	**PURPOSE**
facilities-mgmt system-parameters mnemonic-dial	Displays the number of mnemonic dialing names available. Also assigns new mnemonic names and associated telephone numbers or changes telephone numbers associated with existing names.

Notes:

Data Communications Access

Description

The DCA (Data Communications Access) feature provides an analog interface to local (on-premises) computer facilities. This feature is useful for host computers already setup with data sets (modems) for analog conversion. DCA can also be useful when a large percentage of calls to the supported host will be from analog facilities (analog trunk calls or analog interfaced local terminals). DCA ports are connected directly to analog tie trunk circuits on the switch. Figures 43-1 and 43-2 show several types of data endpoints that can access the host through these tie trunk circuits.

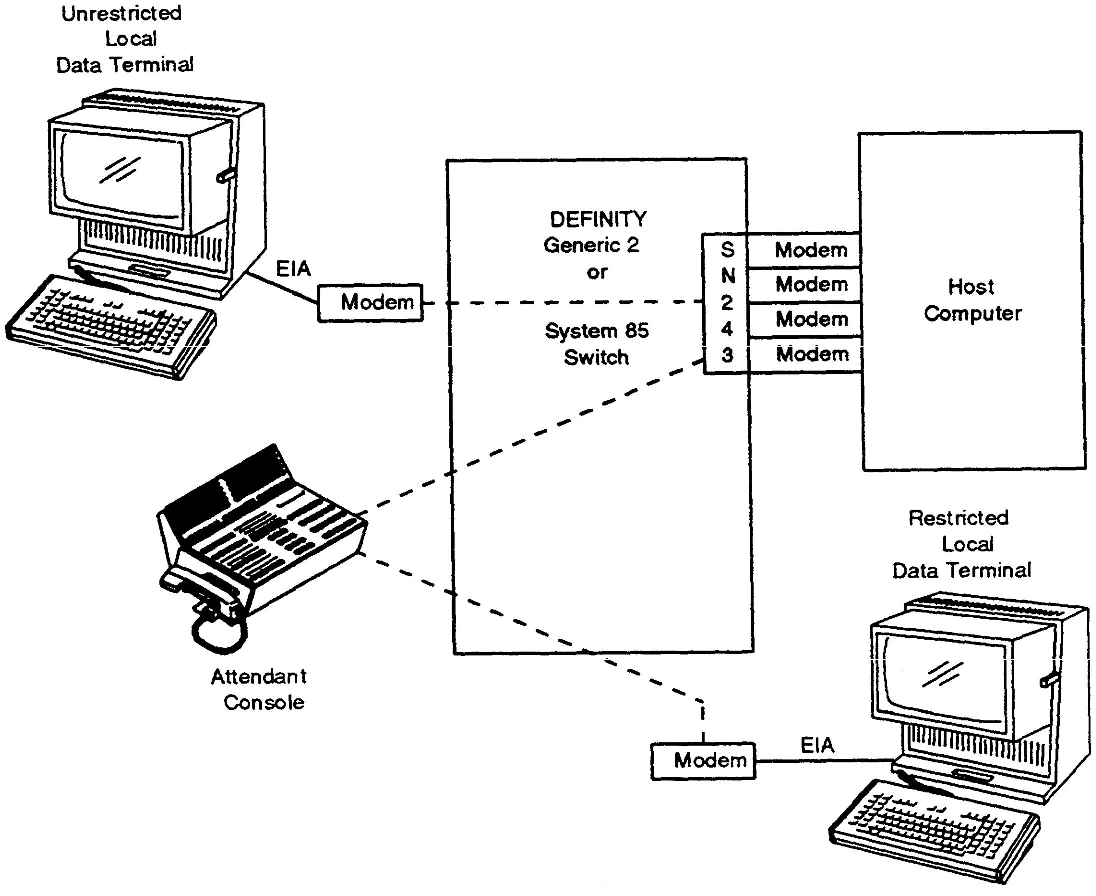

Figure 43-1. Data Communications Access, On-Premises Connections

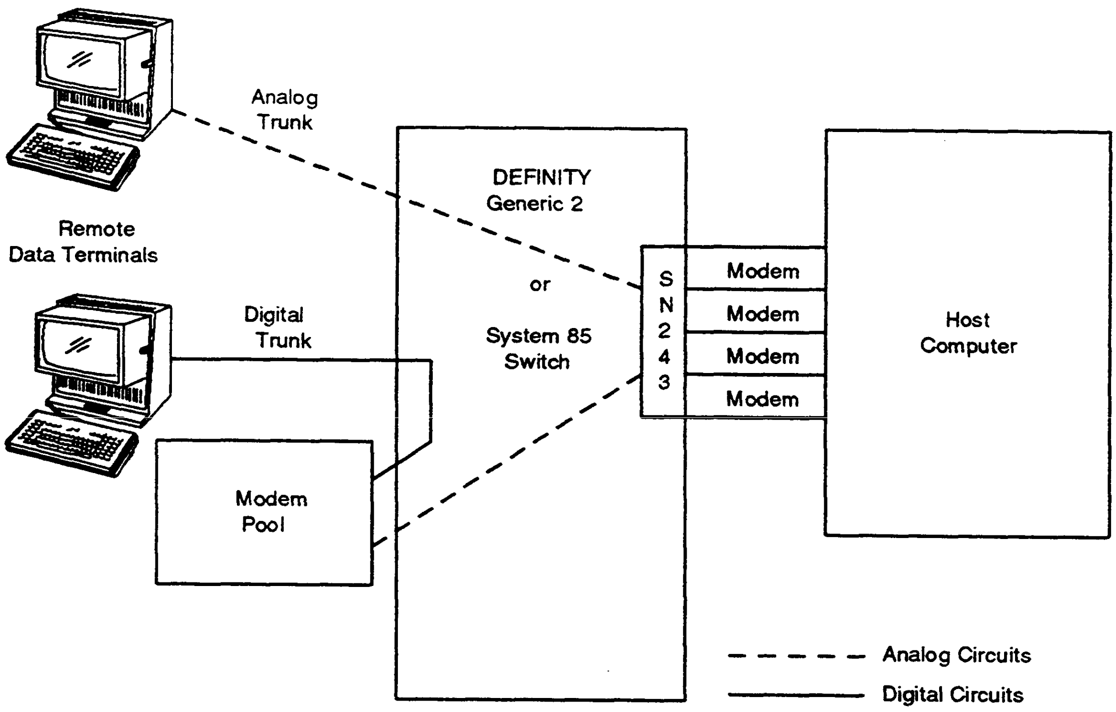

Figure 43-2. Data Communications Access, Off-Premises Connections

Figures 43-1 and 43-2, show some typical DCA connections. Most DCA users will access the host as shown in Figure 43-1, using an analog voice terminal paired with a data terminal and modem ***(Analog Data Call Setup).*** Access to the host through the attendant console is also shown to illustrate the attendant's ability to extend data calls for restricted users (see Restricting Feature Use). It should be noted that this attendant ability applies only to the DCA feature (not the Host Access feature) and that attendant-extended calls are not provided with Data Protection—Temporary.

Figure 43-2 shows that external (off-premises) data calls can also access DCA ports. External data calls that use analog trunks can access DCA ports directly. External data calls that come in on digital trunks can access DCA ports if the Modem Pooling feature is available. However, digital interfaced (DCP and BRI) terminals on the local switch are not supported by local Modem Pooling. Features that support digital data endpoint access to a local host computer include: the Host Computer Access feature and the DMI (Digital Multiplexed Interface) feature.

DCA ports usually appear as trunk circuits arranged in trunk groups on the switch. In this way, the DCA feature can take advantage of trunk features, such as Queuing and Route Advance, to improve user access to the host computer (see Interactions With Other Features).

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for the Data Communications Access feature.

Analog Voice Terminal Users:

Data users who are interfaced to the switch via an analog voice terminal and modem use Analog Voice Terminal, Data Call Setup procedures to access the DCA feature.

Specific operations may vary depending on the type of modem used. The following procedures are based on use of the DATAPHONE Modem 300/1200:

1. Go off hook on the associated voice terminal (Dial tone).
2. Dial the DCA access code or extension number (Call-progress tone, such as ringing tone).
3. When the host computer responds (Data tone), press the **[DATA]** button to transfer control of the call to the modem.

   (When the connection has been successfully setup, an appropriate login prompt appears on the CRT).

Off-Premises Analog Facility Users

DID (Direct Inward Dialing):

When properly administered (DCA access available via extension number steering) an off premises data user can access the DCA ports over the public network as follows:

1. Dial the 7- or 10-digit number assigned for DCA access (Call-progress tone).
2. When the host computer responds (data tone), press the **[DATA]** button to transfer control of the call to the modem.
3. (When the connection has been successfully setup, an appropriate login prompt appears on the CRT).

Remote Access:

An off-premises data user with access via the Remote Access feature, can access the DCA feature as follows:

1. Using a touch-tone dialing terminal, dial the 7- or 10-digit Remote Access number for the switch (call-progress tones followed by local switch dial tone).
2. Dial the barrier code or authorization code if required.
3. Dial the DCA access code or extension number (Call-progress tone).
4. When the host computer responds (Data tone), press the **[DATA]** button to transfer control of the call to the modem.

(When the connection has been successfully setup, an appropriate login prompt appears on the CRT).

Access Via an Attendant:

In certain cases, DCA call attempts will be routed to an attendant. This is due to restrictions applied to the users or because of the type of facilities being used. The attendant can perform the dialing of the DCA access code and transfer the call to the restricted user if desired.

Considerations

ETN (Electronic Tandem Network) ACCESS

Users located on another switch in an ETN arrangement can access the host computer in two ways.

- ***Extension Number Steering***

  If an Extension Number Steering conversion code is provided, it can also be dialed by ETN users after dialing the private network location code.

- ***Routing Via Location Code***

  The DCA ports can also be reached using the AAR (Automatic Alternate Routing) feature on System 85 or DEFINITY Generic 2.1 switches or the WCR (World Class Routing) feature on DEFINITY Generic 2.2 switches. The DCA access code can be assigned a unique private network location code. The local switch will have two private network location codes in this case. One for routing local communications to voice terminals, attendants, etc. The other code routes data calls to the host computer. The extension number digits that follow the location code may be ignored in this case, but still must be dialed to conform to the numbering plan.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Direct Extension Selection With Busy Lamp Field

When an extension number is assigned to a Data Communications Access port, a Data Communications Access trunk group can be accessed using the appropriate DXS (Direct Extension Selection) button. However, the associated busy lamp does not indicate busy or idle status of the Data Communications Access port.

AAR (Automatic Alternate Routing)

For System 85 and DEFINITY Generic 2.1 switches, the DCA feature is compatible with the AAR feature. Calls to DCA ports that are originated from modem interfaced facilities can use AAR facilities and complete directly to a DCA port. Calls that originate from digital interfaced facilities will require the use of Modem Pooling to complete to DCA ports. However, in either case, AAR facilities and routing patterns can be used to for interswitch DCA calls.

Bearer Capability

The Bearer Capibility feature works with the DCA feature (on DEFINITY Generic 2 switches) in the same way it does with other data interfaces. Bearer Capability defines which types of calls can route directly to DCA ports and which types of calls require Modem Pooling support.

CDR (Call Detail Recording)

The CDR feature can provide statistical information for DCA calls.

Data Protection

Data Protection—Permanent should be assigned to DCA trunk groups. This prevents local system generated tones from interrupting data communications. These tones are issued in response to the activation of certain features, such as Override and Priority Calling. The permanent version of the Data Protection feature should be used because DCA port circuits will be used for data communications only, and this will provide protection for attendant-extended calls which cannot use the temporary form of Data Protection.

DID (Direct Inward Dialing)

Users located off-premises can use public network DID trunks to reach the host computer. For this application, the DCA trunk-group dial access code is assigned to an Extension Number Steering conversion code. Extension Number Steering translates an otherwise unused extension number to the dial access code.

Hunting

The Hunting feature is used to check if an alternate trunk circuit can serve a call when the circuit dialed is busy. Hunting is automatically provided for trunk circuits arranged in trunk groups. When dialing a DCA trunk group, the user is provided with access to the first of as many as 99 idle computer ports. Trunk groups can also be combined using Route Advance to provide additional hunting.

ISDN—BRI (Basic Rate Interface)

The DCA feature is not directly compatible with local ISDN—BRI terminal users. The ISDN—BRI is a digital interface feature and for BRI terminal users to access DCA ports on a local host, a conversion resource must be provided.

ISDN—PRI (Primary Rate Interface)

The DCA feature can be accessed by callers using ISDN facilities (either public or private), provided that the Modem Pooling feature is available locally.

Last Extension Dialed

The Last Extension Dialed feature cannot be used to redial calls to Data Communications Access trunk groups that use extension number steering.

Last Number Dialed

The LND feature can store and redial DCA calls to a host computer.

Modem Pooling

Modem Pooling provides support to the DCA feature for incoming calls that use digital facilities (DS1 trunks). Without Modem Pooling, these calls cannot be successfully connected to the DCA ports.

Queuing

The Queuing feature is compatible with the DCA feature. Queuing can provide a waiting list for DCA trunk groups when these data resources are busy. Then, as resources become available, the first user in queue is served.

A DCA trunk group can be directly accessed using a data terminal. For data-terminal access, off-hook queuing is the only type of queuing that can be used. With off-hook queuing, the data terminal user receives a RINGING prompt until a circuit becomes available.

A DCA trunk group can also be indirectly accessed by using a voice terminal and then by transferring the call to an associated data terminal. For voice-terminal access followed by transfer to a data terminal, either off-hook or on-hook queuing can be used. For off-hook queuing the switch returns the music, announcement, or silence. But for on-hook queuing the calling party receives confirmation tone and then goes on-hook to wait for the queuing callback.

To allow data terminals to directly access and voice terminals to indirectly access the same DCA trunk group, off-hook queuing must be assigned to the DCA trunk group.

If the calling party hears reorder tone, the queue might be full. The user should go on-hook and try again.

Remote Access

The Remote Access feature provides users with access to switch features and services from off-premises locations over the public network system. Access to DCA ports is included in the available features and services. Remote Access users can access the host using the DCA access code or the Extension Number Steering code just as though they were calling from a local terminal. Remote Access users must, however, first gain access to the switch by dialing a barrier or authorization code.

Route Advance

The Route Advance feature provides access to as many as five trunk groups with a single access code. When every circuit in the first trunk group is busy, the system checks the other trunk groups for an idle circuit. Since each data trunk group can contain as many as 99 trunks, this arrangement can provide access to as many as 495 DCA ports using a single dial access code.

Tenant Services

Data Communications Access (analog access to a host computer) is a partitioned feature on System 85 and DEFINITY Generic 2 switches.

Line-side computer access is partitioned using partitioned extension numbers. Each extension number is assigned to a modem and is usually included in a hunt group. In turn, the modem's extension number is assigned to an extension partition allowing data-terminal access for users in that partition and Extension Partition 0.

Trunk-side computer access is partitioned using partitioned trunk groups. Since the DCA trunk type (37) can be partitioned, access to these trunk groups can be dedicated to or shared by an extension partition. However, if a computer is accessed from outside the switch, ***trunk-side*** partitioning would have no effect. There are no partitioning checks between the incoming trunk group and the outgoing DCA trunk group.

WCR (World Class Routing)

For DEFINITY Generic 2.2 switches, the DCA feature is compatible with the WCR feature. Calls to DCA ports that are originated from modem interfaced facilities can use WCR facilities and complete directly to a DCA port. Calls that originate from digital interfaced facilities require the use of Modem Pooling to complete to DCA ports. However, in either case, WCR facilities and routing patterns can be used to for interswitch DCA calls.

Restricting Feature Use

Attendant Control of Trunk Group Access

Calls to DCA ports can be restricted by applying Attendant Control of Trunk Group Access to the DCA trunk groups. Any attempt to access a DCA port is redirected to an attendant for screening. The Attendant can then transfer the call to a DCA port. Attendant-extended data calls cannot use Data Protection—Temporary (see Interactions). Therefore, Data Protection—Permanent should be assigned to the DCA Trunk Group if this control option is to be used.

Attendant Control of Voice Terminals

An attendant can restrict selected terminals from access to the DCA feature with the Attendant Control of Voice Terminals feature. An attempt to access a DCA port by one of these restricted terminals is redirected to the attendant for screening. The attendant can then decide whether or not to extend the call. Again, attendant-extended calls cannot use

Data Protection—Temporary (see Interactions). Therefore, Data Protection—Permanent should be assigned to the DCA Trunk Group if this control option is to be used.

Facilities Restriction Levels

FRLs (Facilities Restriction Levels) provide a means of restricting terminal access to trunk groups via the network routing features (AAR and WCR). The DCA trunk groups are placed in a separate routing pattern to which an FRL is assigned. When calling the DCA number, the user's FRL is compared to the FRL of the routing pattern before access to the DCA port is granted. The attendant can also use the alternate-FRLs attribute to alter these restrictions when appropriate.

Miscellaneous Trunk Restrictions

Local voice terminal lines may be restricted from accessing a DCA trunk group by the Miscellaneous Trunk Restriction feature.

Trunk-to-Trunk Restrictions

Trunk-to-trunk restrictions applied to DCA trunks restrict trunk calls from direct access to the host computer. Attendant assistance is required to reach DCA ports from the off-premises locations. These attendant-extended calls are denied Data Protection.

Trunk Group Restrictions

Access to DCA ports via a trunk-group dial access code can be restricted by activating "dial access denied" for the trunk group. This makes DCA access possible only via the Extension Number Steering feature or one of the network routing features (AAR or WCR). These features use indirect access to the DCA ports.

Hardware Requirements

The following special hardware is required for the DCA feature.

For Traditional Modules:

- SN243 Computer data port circuit pack

  Each computer port appearance of the DCA feature requires the use of a data port circuit on the SN243 circuit pack A single pack contains four circuits. The SN243 circuit packs can be installed in the same slots used by SN233 tie trunk circuit packs.

For Universal Modules:

- TN7A2 Analog line circuit pack

  Each computer port appearance of the DCA feature requires the use of a circuit on the TN742 circuit pack. A single circuit pack contains eight circuits.

Regardless of the Module Type:

- Modem (Modulator/De-modulator)

  A Data Set (such as the 212A or DATAPHONE Modem 300/1200) is required at each analog data endpoint and each host computer port.

Feature Administration

The DCA feature administration is by assigning DCA ports to equipment locations and assigning the equipment locations to one or more trunk groups. The trunk groups are then administered as DCA trunk groups.

On System 85 switches, the DCA feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — DATA COMMUNICATIONS ACCESS			
PROCEDURE	WORD	PURPOSE	SMT
100	1	Assigns trunk type DCA, Remote Access, Off-Premises Terminal), DAC, and dial access restriction. For System 85, R2 V3 and earlier, also assigns Route Advance. The applicable trunk-type is: 37 2-way tie trunk; dial repeating in/automatic out	No
100	4	For System 85, R2 V4 and DEFINTIY Generic 2, assigns Trunk Groups to Route Advance patterns.	No
101	1	Assigns trunk-group characteristics.	No
110	1	Assigns DAC for Trunk-to-Trunk Restrictions.	No
111	1	Assigns Trunk Groups to Trunk-to-Trunk Restrictions.	No
150	1	Assigns trunk circuits to trunk groups.	No
350	1	Assigns first digit dialing plan.	No
354	2	Extension Number Steering codes.	No

Notes:

Data Protection

Description

The Data Protection feature blocks intrusions by bridge-on features (such as, Call Waiting, Override, Busy Verification) that would disrupt data transmissions. This protection applies to intrusions from the local switch only. Unwanted tones generated by a distant switch must be blocked at that end. Any signals present on an incoming data transmission are seen as data.

The Data Protection feature is available in two forms: Temporary and Permanent.

Data Protection—Temporary

The temporary form is activated by users on a per call basis. This is done using a dial access code (or an administrable feature button if available). This option is useful for extensions that are used for both voice and data calling (for example, when a voice terminal is used with an acoustic coupled modem).

Data Protection—Permanent

The permanent form is assigned via switch translation to an extension or trunk group. With this form, data protection is provided automatically for all calls (including voice calls) using these extensions and trunks. The permanent form is useful for extensions and trunk groups that are used exclusively for data calls.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

Data Protection—Temporary:

1. Go off-hook and dial the Data Protection—Temporary access code. [Confirmation tone]
2. Dial the data extension number or access code. [Call-progress tone]

Data Protection—Permanent:

Data Protection—Permanent does not require any user action. It is provided on every call involving the associated extension number or trunk group.

Considerations

Hard and Soft Processor Swaps

Data Protection—Temporary activations are stored in a status portion of switch memory. Therefore, if a data terminal user places a protected data call using the access code and then a hard swap occurs, the call does not remain protected after the hard swap.

Data Protection—Permanent assignments are stored in a translation portion of switch memory. Therefore, if a data terminal user places a protected data call and then a hard processor swap occurs, the data call remains protected after the hard swap.

Interactions With Other Features

The following features affect or are affected by the operation of this feature or are affected by this feature.

Features Using Bridged-On Appearance or Injected Tones

Any feature that uses a bridge-on appearance or injects a tone into an established call would disrupt data flow and possibly cause erroneous data transmission. Data Protection blocks such features, including the following:

- Attendant Call Waiting
- Bridged Call
- Busy Verification of Lines
- Call Waiting
- Override
- Precedence Calling
- Priority Calling
- Timed Recall on Outgoing Calls
- Trunk Verification by Attendant or Voice Terminal.

Attendant Control of Trunk Group Access

Trunks under control of the attendant group cannot be directly accessed. An attendant must access these trunks for voice terminal users. Data Protection—Temporary is not available for attendant-extended calls. These trunks, if used for data transmission, should be assigned Data Protection—Permanent.

Bridged Call

Attempting to bridge onto a call that has Data Protection activated is denied. The switch returns reorder tone.

Data Protection—Permanent is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When Data Protection—Permanent is assigned to a ***shared extension,*** this protection applies to every image of that extension.

Dedicated Switch Connections

To prevent interruptions from bridge-on features, Data Protection is automatically assigned to a Dedicated Switch Connection.

Modem Pooling

Data Protection—Permanent should be assigned to all trunk groups associated with Modem Pooling to prevent any intrusions attempted by bridge-on feature users.

Precedence Calling

The Data Protection feature takes precedence over the Precedence Calling feature. That is, a Precedence Calling call does not preempt a call with Data Protection active.

Precedence calls directed to extensions with data protection active are diverted to attendant assistance. The attendant cannot override data protection by using attendant preemption. Data protection permanent should not be administered to AUTOVON access trunks as it invalidates the Precedence Calling feature.

Restriction—Attendant Control of Voice Terminals

Data Protection—Temporary is not available on attendant-extended calls. If the voice terminal extension is denied direct access to the trunk by the activation of this feature, Data Protection—Temporary cannot be used. Data Protection—Permanent can be provided, however.

Ringing—Distinctive Ringing

When Data Protection—Permanent is assigned to a class of service, all ringing for that class of service is 1-burst ringing.

Tenant Services

The Data Protection feature is a switch-wide resource that can be shared by data-terminal users in every partition. Data Protection—Permanent is assigned to a class of service, and the 63 classes of service are shared by the various extension partitions. Meanwhile, Data Protection—Temporary is activated with a dial access code, and dial access codes are common to the various extension partitions.

Hardware Requirements

None.

Feature Administration

Data Protection—Temporary is assigned on a per-system basis. Dial access is then provided to all users provided with the access code. Data Protection—Permanent is assigned on an extension class of service basis or on a per-trunk group basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — DATA PROTECTION			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
010	3	Assigns Data Protection—Permanent to an extension class of service.	Yes
103	1	Assigns Data Protection—Permanent to a trunk group.	Yes
350	1	Assigns the first digit of the dial access code for Data Protection—Temporary (if required).	No
350	2	Assigns the Data Protection—Temporay dial access code. The applicable encode is: 13 Data Protection—Temporary.	No

Dedicated Switch Connections

Description

A DSC (Dedicated Switch Connection) acts like a hard-wired link between two ports on the switch. In effect, this feature provides a full-time open line between the assigned end points.

These connections include intraswitch line connections or trunk connections terminating to a point on a distant switch.

Once established, a dedicated switch connection remains in effect until specific action is taken to remove it. If a power failure occurs or if the system is reinitialized, the switch automatically reestablishes the connection when power is restored or the initialization is complete.

A dedicated switch connection is administered as either active or inactive. In the active state, a communications path is maintained between the two ends of the connection. In the idle state, the two ends of the connection are held in a busied out state, but are not interconnected. The status of the connection can be changed from one state to the other through switch administration using Procedure 360, Word 1, Field 1.

Prior to R2 V4, Issue 1.1, this feature could be used for either voice or data connections, provided that switch-originated signaling was not needed.

Signaling originated by the endpoints in a DSC connection was only permitted when both endpoints were connected to DS1 ports in the same System 85 module. For this arrangement, the DSC feature provided transparent signal passing.

Feature History and Development

The Dedicated Switch Connections feature was first available on System 85 with Release 2, Version 3.

In its initial form, the DSC feature had limited capabilities as a data feature. The DSC feature was effectively limited to analog circuits. This limited dedicated switch connection service to voice connections. While analog-interfaced data connections could be made, the switch did not provide signaling, and leased line modems or their equivalent were needed to set up an effective data DSC.

Beginning with System 85, Release 2, Version 4, Issue 1.1, connections can be made between DCP data modules. With DEFINITY Generic 2, dedicated switch connections can also be made to ISDN—BRI data modules.

Dedicated switch connections between data modules can be established as long as:

- A communication path and the endpoints for the path are available when the DSC is activated.
- Data modules use compatible options (i.e., modes, data rates, synchronous or asynchronous transmission).

DSCs to Off-Premises Data End Points

When a DSC is established between data modules, the switch reserves and initializes the connection path and the terminating data module(s). Each endpoint is set up separately, so connections can be established between two data modules or between a data module and an AVD (Alternate Voice / Data) DS1 trunk. This allows the system to establish data DSCs to off-premises end points over clear channel DS1 carriers.

Switch Operations

Beginning with Release 2, Version 4, Issue 1.1, the System 85 and DEFINITY Generic 2 switches provide the signaling needed to initialize the data modules. That is, the switch sends signals to each DSC data module that cause it to go off-hook and start handshaking. After handshaking is successfully completed, the data module goes into the Transmit Data mode.

DSC Configurations

The following configurations will be possible with Data Dedicated Switch Connections:

- Connections between a host computer and either a data terminal, a printer, or another host computer
- Connections between two host computers with a high-speed link
- Connections between a host computer with a remote job entry terminal or printer station
- Connections between packet switches across a network of System 85 and DEFINITY Generic 2 switches
- Using a DS1 trunk as the DCIU link between two switches without the need for Channel Division Multiplexors or DDS data sets
- Establishing a mode 1 data module connection to 56 Kbps DSUs (Data Service Units) on a remote D4 Channel Bank by using a suppressed signaling DS1 trunk.

These configurations are illustrated in Figures 45-1 through 45-6.

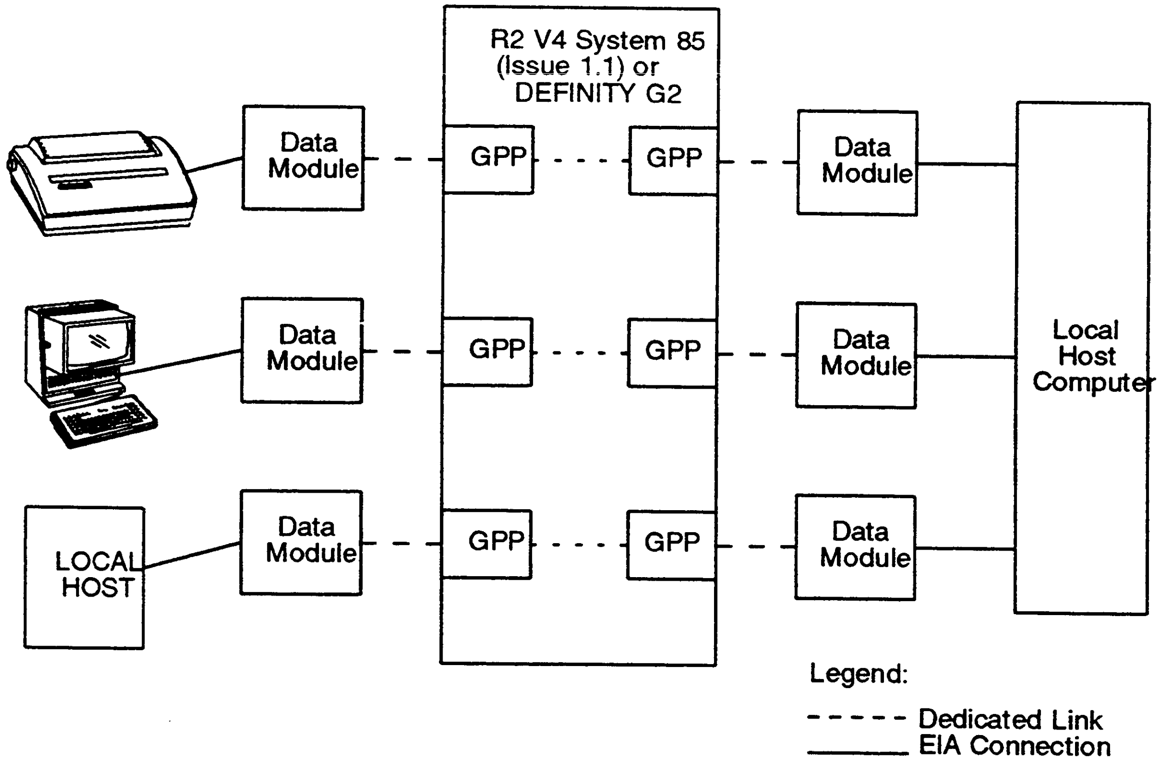

Figure 45-1. Local DSC Configurations

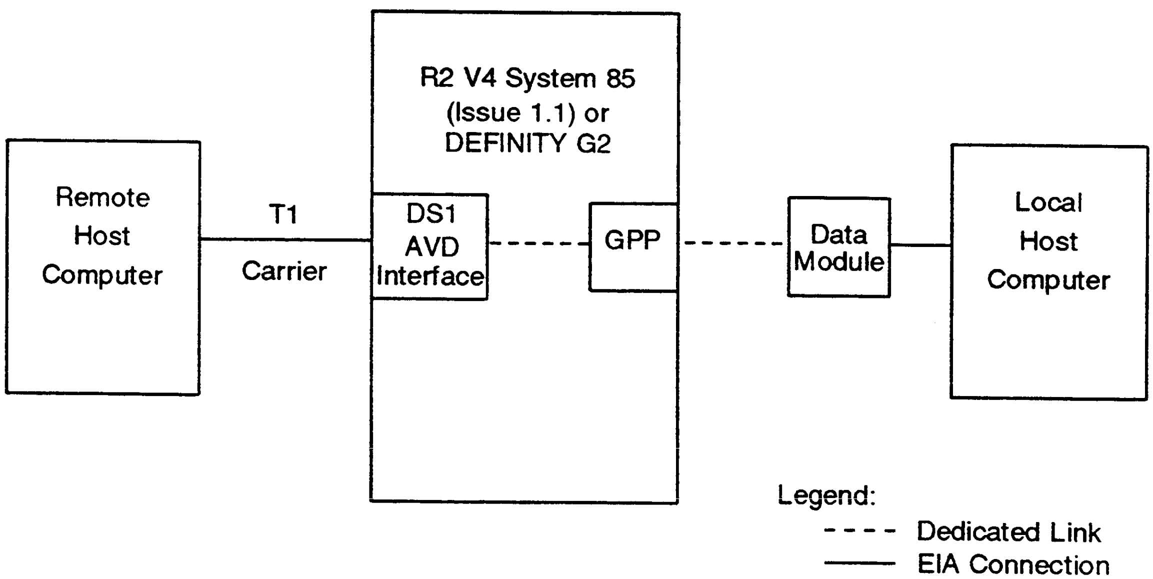

Figure 45-2. DSC Between Two Host Computers

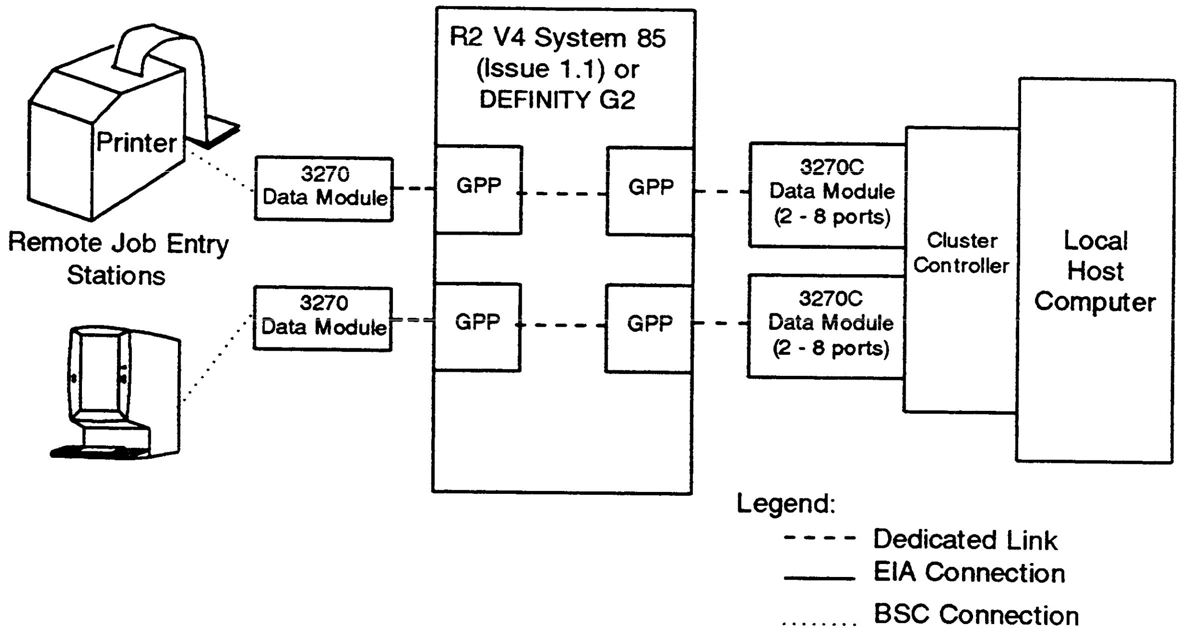

Figure 45-3. DSC Between Host Computer and Remote Job Entry Stations

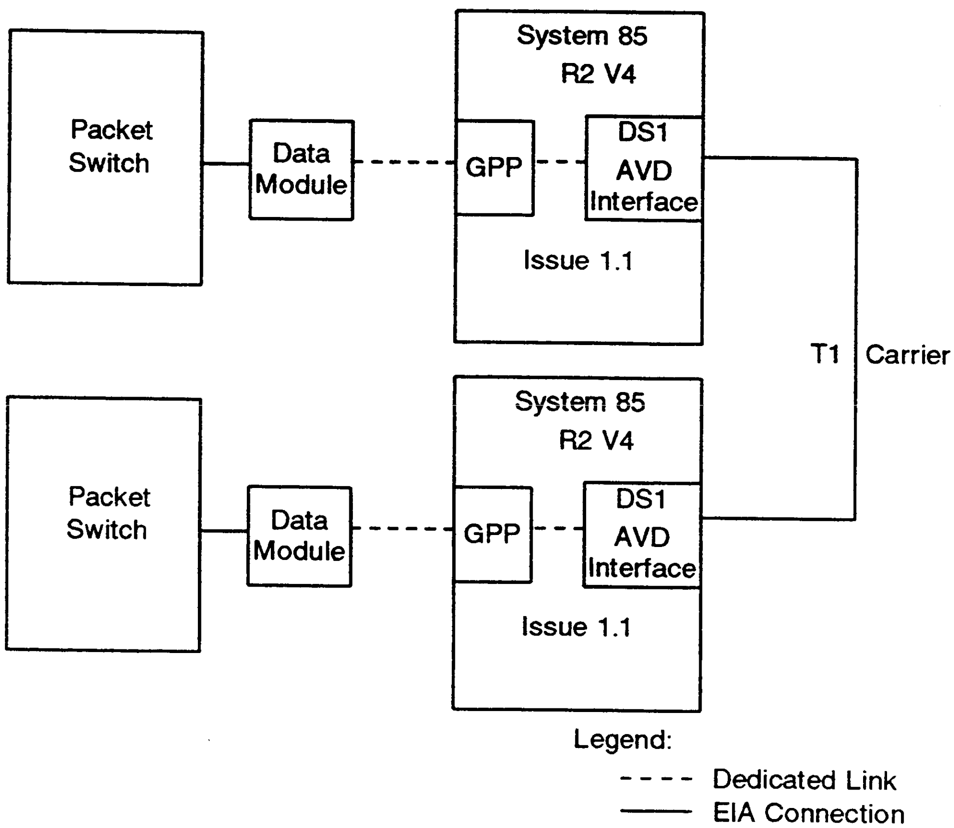

Figure 45-4. DSC Between Packet Switches

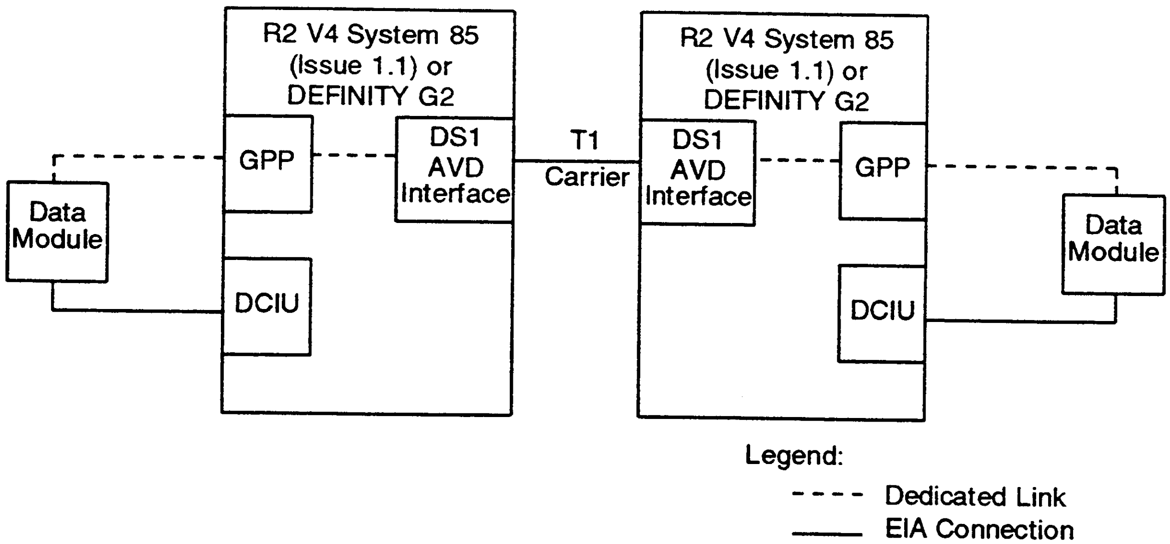

Figure 45-5. DSC as DCIU Link

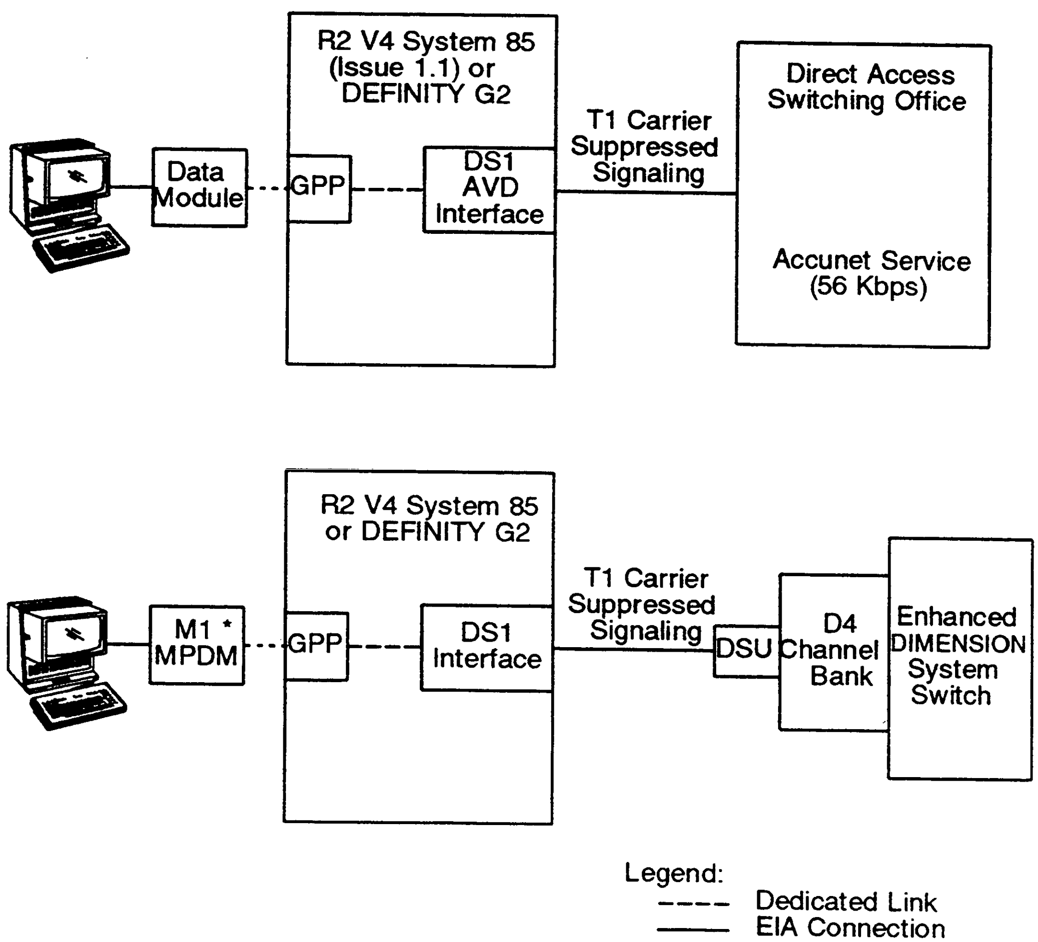

Figure 45-6. DSC to DS1 Trunk Using Suppressed Signaling Channel

Call Processing Changes to DSCs

The switch treats each endpoint of a DSC independently. The communication path is created during administration or initialization. Once initialized, endpoints are free to communicate as necessary. This means that DSCs on separate switches, connected by AVD trunks, do not need trunk signaling between switches. Call-processing software independently maintains each endpoint in the dedicated mode. This minimizes the needed call-processing intervention. Call-processing support is limited to audit routines ensuring that the connection is still up, and that the data modules are in the Transmit Data mode.

User Operations

Dedicated Switch connections are fixed and automatic. Once a DSC is established, there are no user operations required for the feature.

Considerations

Capacity

Up to 1,023 Dedicated Switch Connections can be provided on a single switch. The actual number depends on the availability of time slots in the switch.

Intermodule DSCs (on a multimodule switch) are supported. However, transparent signal passing is not supported using this arrangement.

Processor Data Modules and Trunk Data Modules

With R2 V4, Issue 1.1, PDMs (Processor Data Modules) and TDMs (Trunk Data Modules) can be used as interface devices on DSCs. Prior to R2 V4, Issue 1.1, PDMs and TDMs could not be used as DSC interface devices. PDMs and TDMs require S-channel signaling on the switch side. This was not provided on a dedicated switch connection. Furthermore, PDMs and TDMs rely on the switch for necessary call-control signaling which was not provided for a dedicated switch connection.

EIA Ports

With R2 V4, Issue 1.1, EIA ports can be used as interface devices on DSCs. Prior to R2 V4, Issue 1.1, EIA ports could not be used with dedicated switch connections. Like data modules, EIA ports require S-channel signaling and rely on the switch for necessary call control signals that were not provided for a dedicated switch connection.

3270 Series Protocol Converters

The 3270 Series is a group of protocol conversion units that provide DCP interface for IBM 3270 type BSC (Binary Synchronous Communications) cluster controllers and terminals. Beginning with R2 V4, Issue 1.1, the 3270 Series protocol converters operate in the dedicated mode.

Multiple DSC Assignments

The same endpoint (equipment line location) can be assigned to more than one DSC. However, when this is done, only one of these multiple DSCs can be active at any given time.

Switch Services

Many switch services including call-origination signaling, call-termination signaling, and access to features are disabled.

Prior to R2 V4, Issue 1.1, the terminals or interfaces that connect to each end of the DSC must provide all required supervision, signaling, and maintenance. Beginning with R2 V4, Issue 1.1, the switch provides supervision and signaling for DCP data modules, but these modules must still provide the required maintenance.

The switch ***does*** perform audit checks to ensure that the connection remains intact and that the data modules are in the Transmit Data mode.

Rotary Dial Signals

When transparent signaling is used between DS1 ports on the same System 85 or DEFINITY Generic 2 module, rotary dialing through a DSC typically works. However, this is not fully reliable, and is not guaranteed.

Touch-Tone Signals

If digit sending and receiving is required, the digits must be sent on a touch-tone cut-through basis. For ports involved in a DSC, System 85 or DEFINITY Generic 2 does not collect dial pulses, detect touch-tone signals, or provide digit outpulsing or sending.

Endpoint Compatibility

The switch administrator should ensure that the pair of endpoints in a DSC are compatible (capable of mutual communication) without intervention by the System 85 or DEFINITY Generic 2.

Incompatible Circuits

Certain System 85 or DEFINITY Generic 2 circuitry ***cannot*** be used with the DSC feature:

- Attendant console and attendant conference circuits
- ANI (Automatic Number Identification) circuits
- Contact interface circuits
- DMI (Digital Multiplexed Interface) ports
- Multiappearance voice terminals
- Tone plant circuits.

DS1 DSC Trunks

When a DS1 trunk circuit is used, it must also be set up as a dedicated connection. This trunk circuit will normally be in the form of a tie trunk to another private switch, although it can be a leased line circuit through a CO (Central Offce).

Connections to DS1 facilities are made on the assumption that the far end of the circuit connects to a DSC compatible device somewhere in the network. The local switch has no way to verify this connection.

Whenever possible, AVD DS1 trunk circuits should be used for dedicated switch connections that go off-premises. Bit-robbed DS1 trunks **with suppressed signaling** can be used if required; however, the quality of service cannot be assured. This type of trunk would be needed when the far end of the trunk terminates on a D4 channel bank rather than another System 85 or DEFINTIY Generic 2 switch.

Equipment Combinations

The following list shows permissible combinations for a data DSC. Within this list, a data DS1 trunk is an AVD trunk or a voice grade DS1 trunk with suppressed signaling.

- DCP (Data Communications Protocol) data line to a DCP data line
- DPC data line to a data DS1 trunk
- Data DS1 trunk to a data DS1 trunk

Compatibility Requirements

The following restrictions apply when configuring a DSC:

- Both endpoints must be operating in the same mode.
- Mode 3 endpoints must be administered in complimentary status — one as originator and the other as terminator.
- Data modules must be administered as station, not trunks.

Mixed Analog/Digital DSCs

The Dedicated Switch Connections feature does not support a mixed analog/digital configuration. A user may, however, be able to configure back-to-back digital and analog DSCs by hard-wiring (or cross-wiring) the DCP data module to a leased-line modem on an analog DSC. Such an arrangement would consume two DSCs and two time slots.

Resource Consumption

A dedicated switch connection consumes little processor time. However, a DSC uses two terminal ports and one switch time slot on a full-time basis. In systems with high usage or where assured Zero Blockage is needed, dedicated switch connections should be limited to essential applications or situations where the circuit will be in continuous or nearly continuous use.

For other applications, it may be desirable to consider features that consume switch resources only while in active use, such as the Hot Line feature or default dialing with the Data Call Set Up feature.

Hard and Soft Processor Swaps

Dedicated Switch Connections will endure a hard processor swap. However, Dedicated Switch Connections cannot be established during a hard swap.

The DSC relationships assigned in Procedure 360, Word 1 are stored in a translation portion of switch memory. Therefore, these relationships will endure a hard processor swap.

The Dedicated Switch Connections feature operates normally during a soft processor swap.

Commercial Power Failures

Once a DSC is established and a run-tape operation has been performed, a DSC remains in effect until specific actions are taken to remove it. If a power failure occurs or if the system is reinitialized, the switch automatically reestablishes the connection when power is restored or the initialization is complete.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features are affected by the operation of this feature.

Automatic Circuit Assurance

The ACA feature ignores a long-holding time threshold exception when a DSC is involved in the call.

Bearer Capability

The Bearer Capability feature has no impact on Dedicated Switch Connections. Bearer capability is not checked when a dedicated switch connection is set up.

Data Protection

The Data Protection feature is automatically assigned to a DSC to prevent interruptions from bridge-on features.

DMI (Digital Multiplexed Interface)

The DMI feature is not, at the present time, compatible with Data Dedicated Switch connections.

DS1 Interface

If a pair of DSC ports reside in the ***same*** module and are DS1 ports, the following is available:

- If the ports are located on DS1 boards administered as Robbed Bit Signaling, the DS1 disabled signaling option (Procedure 000 for lines, and Procedure 116 for trunks) can be used to disable Robbed Bit Signaling for the ports. This provides a 64 Kbps clear channel through the DSC. If not, then DSC transparent signal passing is the default.
- If both ports are located on DS1 boards translated as AVD, then a 64 Kbps clear channel is provided.
- If a DS1 port on a robbed-bit DS1 board is connected to a DS1 port on an AVD board, the DSC feature provides transparent signal passing.
- A 64 Kbps clear channel is also available when two DS1 circuits are involved in the same DSC. System 85 and DEFINITY Generic 2 assure frame synchronization between the two ports. However, a pair of DS1 ports cannot use super-frame synchronization.

The DS1 Interface feature is used to provide off-premises data dedicated connections service. The AVD DS1 service is the option of choice. However, voice grade DS1 service can be used as long as the circuit is administered with suppressed signaling.

Host Computer Access

The Host Computer Access feature is compatible with the Dedicated Switch Connections feature. For host computer ports to be assigned to a DSC, they must be administered as separate line appearances and interfaced with a DCP data module. These ports are not, however, provided switched services as are other host computer access ports.

ISDN—BRI (Basic Rate Interface)

ISDN—BRI data terminals can be assigned to a Dedicated Switch Connection, however, BRI voice terminals cannot. If a BRI voice / data station (both voice and data terminal on the same interface with the same extension number) is assigned to a Dedicated Switch Connection, the assignment will effect the data terminal only, the voice terminal will not be effected.

ISDN—PRI (Integrated Services Digital Network/Primary Rate Interface)

The ISDN—PRI feature supports Dedicated Switch Connections. That is, ISDN—PRI B channels can be used for DSC trunking facilities.

Modem Pooling

The Modem Pooling feature does not support data dedicated switch connections.

Tenant Services

There are no tests in Procedure 360, Word 1 to ensure that the endpoints of a dedicated switch connection belong to compatible extension partitions. It is the responsibility of the system manager to ensure that these connections are allowed.

Hardware Requirements

There are no special hardware requirements to implement this feature other than customer-provided equipment interfaced to the DSC ports.

Any existing DCP or BRI data module can be used. These data modules include:

- ADM-T
- PDMs and TDMs
- DTDMs and Data Stands
- EIA ports
- 3270 Data Modules
- 7400A and 7400B Data Modules
- 7500B Data Modules.

Feature Administration

The Dedicated Switch Connections feature is assigned on a per-circuit basis.

On System 85 switches, the DSC feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal), the TCM (Terminal Change Management) feature, or the FM (Facilities Management) feature

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered with the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE — DEDICATED SWITCH CONNECTIONS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	3	Assigns DSC message capability (G 2) in Field 3. Use "0" if using a DTDM, PDM, TDM or MDM. Use "1" if using a 7400 or 3270 data module.	N/A
360	1	Assigns a dedicated switch connection.	Yes

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — DEDICATED SWITCH CONNECTIONS	
PATH NAME	**PURPOSE**
terminal-change dedicated-conn	Assigns DSC to an equipment location.

The following is the applicable FM path name used with the AP 16.

FM SCREEN — DEDICATED SWITCH CONNECTIONS	
PATH NAME	**PURPOSE**
facilities-mgmt dedicated-conn	Assigns a DSC to an equipment location.

Dial Access to Attendant

Description

A voice terminal user can access an attendant by dialing an access code, usually zero (0), by dialing an LDN (listed directory number), or by dialing an RNX followed by "0111" (the Universal Attendant Code). In a switch with multiple attendants, a specific attendant can be accessed when an individual access code is assigned.

This feature permits a terminal user to check with an attendant for information or to request assistance. The attendant can then extend the call to another station or trunk, or activate features not directly available to the calling terminal.

Feature History and Development

This feature was first available for System 85 in Release 1. The ability to call a local attendant using an LDN was first provided in Release 2, Version 3.

User Operations

The following are the user operating procedures for this feature.

To Access an Attendant:

1. Go off-hook. [Dial tone]
2. Dial the attendant access code. [Ringback tone]

To Access an Attendant Using an LDN:

1. Go off-hook. [Dial tone]
2. Dial an LDN. [Ringback tone]

To Access an Attendant Serving Another ETN (Electronic Tandem Network) Switch:

1. Go off-hook. [Dial tone]
2. Dial the AAR (Automatic Alternate Routing) dial access code. [Second dial tone]
3. Dial the RNX (location code) of the desired switch in the ETN network.
4. Dial **[0] [1] [1] [1]** . [The call routes to the attendant queue at the dialed ETN switch, and the calling party hears ringback tone.]

To Access a Selected Attendant:

1. Go off-hook. [Dial tone]

2. Dial the selected attendant access code. [Second Dial tone]
3. Dial the selected attendant's assigned 2-digit position number. [Ringback tone]

Considerations

Attendant Conference Entry

A voice terminal user, including a Remote Access user, calling the attendant by Dial Access can request attendant conference service and be included as the first member of an attendant conference. However, once an attendant conference has been started, a user cannot use Dial Access to the Attendant and be added to an existing conference.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect the operation of this feature.

Conference—Attendant Five Party

The requesting party (first member of the conference) can use Dial Access to Attendant and be connected to the conference. However, once an attendant conference has been started, a voice terminal user cannot use Dial Access to Attendant and be added to an active conference. The attendant must originate the call to add a party to an active conference.

Conference—Attendant Six Party

The requesting party (first member of the conference) can use Dial Access to Attendant and be connected to the conference circuit. However, once an attendant conference has been started, a station user cannot use Dial Access to Attendant and be added to an active conference circuit. The attendant must initiate the contact to add a party to an active conference circuit.

Intercept Treatment

The Attendant Diversion to Recorded Announcement function overrides the Dial Access to Attendant feature. When an attendant activates Attendant Diversion to Recorded Announcement, a local voice terminal user cannot math the attendant queue or a selected attendant. Instead, these calls are diverted to the recorded announcement.

Look-Ahead Interflow

If the Dial Access to Attendant feature is enabled at a receiving switch, the receiving Look-Ahead Interflow vector can contain a "route to" step with the Attendant Dial Access code (encode 8) as the destination. This "route to" step (usually used as an ***alternate*** destination) will unconditionally deliver calls to the local attendant queue.

Malicious Call Trace

If a voice terminal user selectively calls the activating or the controlling attendant during an active Malicious Call Trace, the call is denied. The switch returns busy tone to the calling party. A controlling attendant unbusying his or her console will receive the call without disturbing the ICI display that contains the trace information.

Remote Access

A Remote Access user can call the attendant group by dialing the attendant access code.

Beginning with R2 V3 System 85, a Remote Access user can call the attendant group by dialing a DID LDN.

Tenant Services

In a partitioned switch, the Dial Access to Attendant feature is limited. Each extension partition can be assigned to one attendant partition. (More than one extension partition can be assigned to the same attendant partition.) These associations between extension partitions and attendant partitions are used to determine whether the Dial Access to Attendant feature is allowed, and the system manager is required to establish these associations.

When a voice terminal user dials the "general" attendant access code (usually "0"), the call can only complete to an attendant in the associated attendant partition. If there is no attendant partition associated with the user's extension partition, the switch denies the call and returns intercept treatment to the calling party.

Local voice terminal users can dial an LDN to reach the attendant queue. When this is done, partitioning checks are made. When a voice terminal user (in a partition other than Extension Partition 0) dials an LDN, the call is allowed if the user's extension partition is assigned to the called attendant partition or if the user is calling Attendant Partition 0. If not, the switch denies the call and returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can use an LDN to call any attendant partition in the switch.

When a voice terminal user (in a partition other than Extension Partition 0) dials a selected attendant access code, the call is allowed if the dialed attendant resides in the associated attendant partition or in Attendant Partition 0. If not, the switch denies the call and returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 is allowed to call a selected attendant in any partition.

Unattended Console Service—Alternate Console Position

The Alternate Console Position feature provides an alternate attendant position that can be used in lieu of a regular (primary) attendant position. Calls normally directed to the primary position route to the alternate position.

Restricting Feature Use

The Dial Access to Attendant feature can only be resticted by denying a voice terminal the ability to originate calls. This is done with the MAAP, SMT, or DEFINITY Manager II by applying Origination Restriction to a voice terminal class of service, or the attendant can use the Attendant Control of Voice Terminals feature to apply Controlled Total Restriction to a voice terminal.

Hardware Requirements

None.

Feature Administration

Assignment of the Dial Access to Attendant feature is on a per-system basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — DIAL ACCESS TO ATTENDANT		
PROCEDURE	**WORD**	**PURPOSE**
200	1	Enables access to a selected attendant (enter 2 in Field 5).
210	1	Assigns the equipment location of an attendant console position.
350	1	Assigns the first digit of the attendant dial access codes (if required).
350	2	Assigns the attendant dial access codes. The applicable encodes are as follows: 8 Attendant dial access code 30 Extension to selected attendant.

Digital Multiplexed Interface

Description

The Digital Multiplexed Interface feature provides digital connectivity between System 85 or DEFINITY Generic 2 switches and a host computer.

In its original form, this feature used a DS1 signaling format similar to the AVD (Alternate Voice Data) service of the DS1 Interface feature. In this arrangement, twenty-four 64 Kbps (DS0) channels are multiplexed onto a 1.544 Mbps DS1 carrier. Of these, 23 channels are used for communications channels and 1 channel is reserved for signaling. This technique is referred to by a variety of names: common channel signaling, 24th channel signaling, or out-of-band signaling. Refer to the DS1 Interface feature for a detailed discussion of the differences between the AVD service (using out-of-band signaling) and the Voice Grade service (with robbed bit or in band signaling) available from DS1.

For System 85, Release 2, Version 4 and DEFINITY Generic 2, the DMI feature is available in two versions: a BOS (Bit-Oriented Signaling) version that is similar to the original form but with an enhanced signaling format, and a MOS (Message-Oriented Signaling) version that is designed to function with the ISDN—PRI (Primary Rate Interface) feature.

The BOS (Bit-Oriented Signaling) Version

The BOS version of the DMI feature uses an enhanced bit-oriented signaling format on the signaling channel for call control signaling. This version, like the original version, provides 23 communications channels and uses common channel signaling.

The enhanced DMI BOS can be easily retrofitted to earlier System 85 Release 2 versions through a simple hardware upgrade. It provides all of the DS1 functionality of the earlier versions. Some of the supported connections include DEFINITY Generic 2 to System 85, to System 75, to host computers, and to toll-network equipment (Direct or Bypass Access arrangements).

DMI BOS is a generally satisfactory arrangement. However, because it uses BOS, it must be connected to ISDNs through the ***interworking*** functions of the local switch. Interworking is described later in this section and in the ISDN—PRI chapter of this manual. As a result of the interworking requirement, the DMI BOS version cannot take full advantage of all ISDN features and services.

The MOS (Message-Oriented Signaling) Version

The DMI feature is also enhanced in System 85, Release 2, Version 4, and in DEFINITY Generic 2 with the addition of the MOS version. The MOS version is fully compatible with the ISDN—PRI Interface feature. It is actually a specialized form of the ISDN—PRI. DMI MOS is similar to the BOS version in that it uses 23 communications channels (called B or bearer channels in ISDN terminology) and 1 signaling channel (call the D or data channel in ISDN terminology) multiplexed on a 1.544 Mbps, DS1 earner link. This version also uses common channel signaling. However, the DMI MOS version uses the same message-oriented signaling format that is used with ISDN for call control and signaling. Also like ISDN—PRI, outgoing DMI MOS calls must use either AAR, ARS, or WCR for routing. Refer to the lSDN—PRI feature for more information on the ISDN standards.

The Interworking Function

In System 85, Release 2, Version 4 and in DEFINITY Generic 2, the ISDN concept of interworking has been implemented. Interworking allows ISDN and non-ISDN features and services to work together to the maximum extent possible.

In effect, the System 85, Release 2, Version 4 and the DEFINITY Generic 2 switches contain two separate sets of call processing software: one for standard (non-ISDN) calls and one for ISDN calls. Interworking functions essentially as follows.

Non-ISDN Originated Calls

Calls originating on the ISDN capable switch (System 85, Release 2, Version 4 or DEFINITY Generic 2) or calls coming into the switch from a non-ISDN trunk facility are handled by the standard call processing software. In this way, they are provided with all of the features and services allowed to their COS (Class of Service).

When the switch recognizes that the call is to go to an ISDN facility, it notifies the ISDN call processing software. The ISDN call processing routines perform all the functions necessary to support the ISDN side of the call. It also provides ISDN features and services, such as Call-by-Call Service Selection (the ISDN features and services are listed and described in the ISDN—PRI chapter of this manual). When both sets of call processing routines are ready, the call is completed.

ISDN Originated Calls

For calls received on an ISDN facility (trunk appearances, including DMI MOS ports), the calls are processed in the same way except in reverse order. The incoming call is first handled by the ISDN call processing software. The standard call processing software is notified by the ISDN layer that a call is ready for completion, and the standard call processing sets up and services the switch side facilities (stations or non-ISDN trunk connections) in the same way as a conventional incoming call. All of the standard switch features and services that would normally be available (based on COS) are provided.

Figures 47-1 and 47-2 illustrate how interworking operates with either the DMI BOS or the DMI MOS versions.

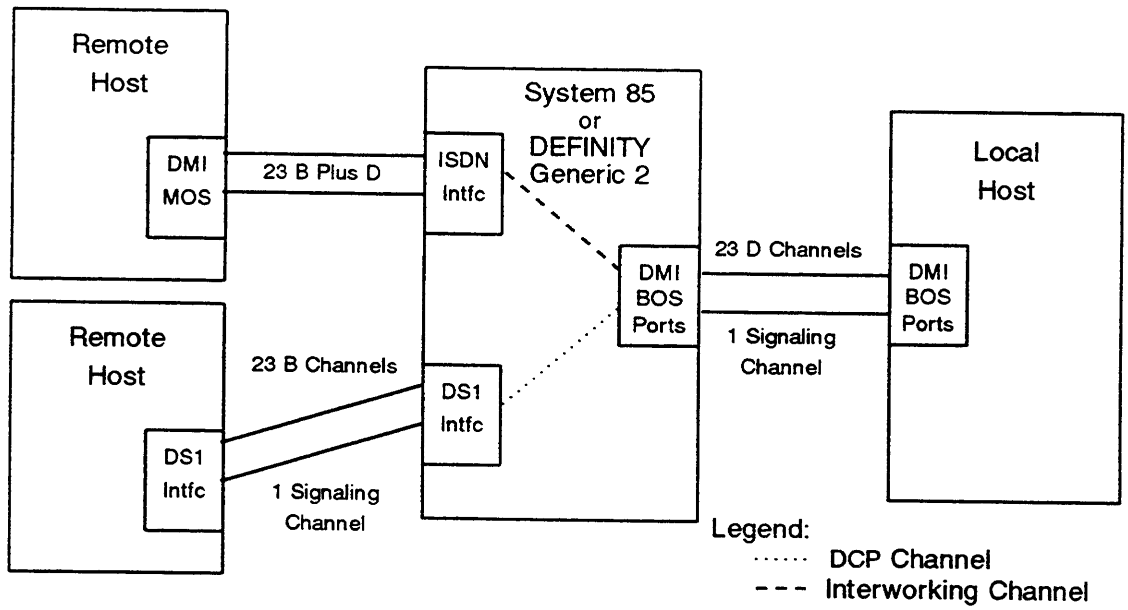

Figure 47-1. Interworking With DMI BOS

It is important to note that the MOS version of DMI is available on ISDN capable switches (System 85, Release 2, Version 4 and DEFINITY Generic 2) ***in addition to*** the BOS version. That is, both versions can be used on the same system at the same time.

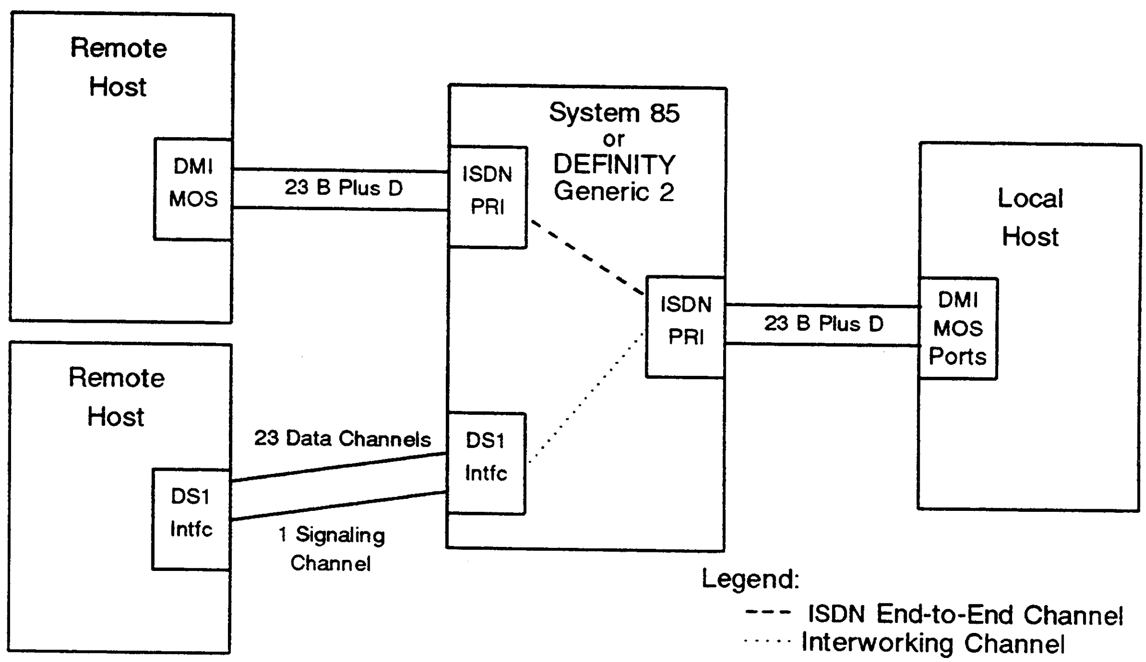

Figure 47-2. Interworking With DMI MOS

While DMI BOS and DMI MOS are not directly compatible, neither are they mutually exclusive. That is, the same host can have both a DMI BOS Interface and a DMI MOS Interface to the same switch. Figure 47-3 illustrates this arrangement.

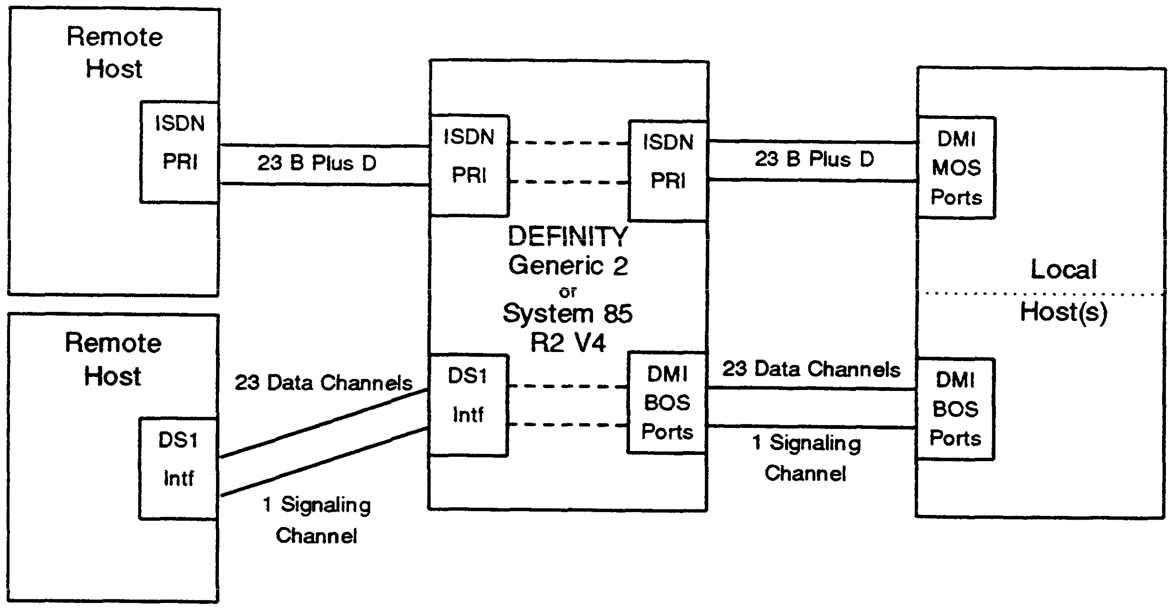

Figure 47-3. Multiple DMI (BOS and MOS) Arrangement

Accessing the DMI

The data channels of a DMI host can be accessed by both analog and digital data endpoints, including data terminals and other hosts. The following are the basic connections possible with the DMI feature:

- Local data terminals:
  - — Digital interfaced data terminal to DMI host, direct connection using DCP (Digital Communications Protocol) or BRI (Basic Rate Interface)
  - — DMI host data endpoint to digital interfaced terminal
  - — DMI host data endpoints to other DMI host data endpoints
- Trunk connections:
  - — Analog trunk facility to DMI BOS host via the Modem Pooling feature
  - — Analog trunk facility to DMI MOS host via the Modem Pooling feature and interworking function
  - — ACCUNET Service Interface trunk facility to DMI BOS host, direct connection using DCP
  - — ACCUNET Service Interface trunk facility to DMI MOS host, via interworking function
  - — Non-ISDN digital (AVD) trunk to DMI BOS host, direct connection using DCP
  - — Non-ISDN digital (AVD) trunk to DMI MOS host, via interworking function

- ISDN—PRI trunk facility to DMI MOS host, direct (ISDN End-to-End) connection
- ISDN—PRI trunk facility to DMI BOS host, via the interworking function

- Host-to-host connections:
  - Analog interfaced host (Data Communications Access feature) to DMI BOS host via the Modem Pooling feature
  - Analog interfaced host (Data Communications Access feature) to DMI MOS host via the Modem Pooling feature and interworking function
  - Non-DMI digital interfaced host (Host Computer Access feature) to DMI BOS host, direct connection via DCP
  - Non-DMI digital interfaced host (Host Computer Access feature) to DMI MOS host, via interworking function
  - DMI BOS host data endpoint to DMI BOS host data endpoint, direct connection via DCP
  - DMI BOS host data endpoint to DMI MOS host data endpoint, via interworking function.

Feature History and Development

- The DMI feature was first available for System 85 in Release 2, Version 2. This was a common channel signaling version that used the DS1 AVD signaling format.
- An improved BOS (Bit-Oriented Signaling) format was introduced in System 85, Release 2, Version 4. This new format can be retrofitted (via a hardware change) to System 85, Release 2, Version 2 and 3 switches.

  Also in System 85, Release 2, Version 4, the DMI signaling format was changed to include MOS (Message-Oriented Signaling) for full ISDN compatibility. DMI MOS is not backward compatible with earlier systems; however, systems using the BOS format can communicate with switches using DMI MOS via the interworking function.

  With the DEFINITY Generic 2 switch, the ISDN—PRI enhancements, specifically NFAS (Non-Facility Associated Signaling) apply also to the DMI MOS feature.

User Operations

For data terminal users, the operation of DMI is transparent, other than the possibility of improved response times and faster cursor movement due to the higher data rate that can be used. The actions required of the data terminal users are described in the Data Call Setup feature.

With the DMI MOS version, the host can be programmed to provide ISDN route selection through the ISDN feature Call-by-Call Service Selection.

Considerations

For Data Terminal Users

Analog Facilities

If analog facilities are to be used, the Modem Pooling feature must be provided.

Digital Data End-Points

If the data endpoints to be connected are all digital (DCP or BRI interfaced), there are no special considerations required to implement DMI.

Hard and Soft Processor Swaps

Stable calls over DMI trunk groups endure a hard processor swap. However, calls cannot be placed over DMI trunk groups during a hard processor swap.

Differences Between Versions

There are two separate and distinct versions: DMI BOS and DMI MOS. The enhanced DMI BOS version can be easily retrofitted to earlier System 85s (with hardware upgrades on the older systems). The DMl MOS version is fully compatible with the ISDN—PRI feature but cannot be retrofitted to switches earlier that System 85, Release 2, Version 4.

Although both versions are generally similar, they are not directly compatible with one another because of differences in the signaling form used. The interworking function allows either version to be used with either ISDN or standard DCP or DS1 Interface facilities. However, when interworking is involved in the call setup, some of the ISDN features and services are not available.

Both versions (DMI BOS and DMI MOS) can be supported on the same host at the same time.

Interactions With Other Features

ACCUNET Service Interface

The ACCUNET Service Interface feature ***is not compatible*** with the DMI feature in the **BOS version.**

The ACCUNET Service Interface feature ***is compatible*** with the DMI feature using the **MOS version,** but only at data rates of 56 Kbps or less. If data rates of 64 Kbps are used, the ACCUNET Service Interface feature cannot be used.

AAR/ARS (Automatic Alternate Routing/Alternate Route Selection)

On System 85 and Generic 2.1 switches, DMI MOS must use either the AAR or ARS feature for outgoing call routing.

Bearer Capability

The DMI feature is compatible with the Bearer Capability feature. The Bearer Capability feature does not apply to switches prior to DEFINITY Generic 2. For DEFINITY Generic 2 switches, the Bearer Capability feature provides the user with the ability to specify detailed information on the types of data calls that can be supported by a specific facility (such as DMI interface circuit).

Data Protection

The Data Protection feature supports DMI. Data Protection—Permanent should be assigned to the DMI ports to prevent the introduction of unwanted signals into a data channel.

ISDN—BRI (Basic Rate Interface)

The ISDN—BRI feature provides an alternate form (to DCP) of digital terminal interface on Generic 2 switches. The BRI connected to data terminals is fully compatible with the DMI feature (either MOS or BOS versions). However, ISDN—BRI is a line side only feature in Generic 2.1 and 2.2. Therefore, the BRI feature cannot be used for the host side connections in a DMI arrangement, even if the host is local to a Generic 2 switch.

ISDN—PRI (Primary Rate Interface)

The ISDN—PRI feature can be accessed by either the MOS or BOS version of DMI. However, when the DMI BOS version is used to access ISDN facilities, the interworking function is involved, and some of the ISDN features and services are not available.

Modem Pooling

The Modem Pooling feature supports connections to the DMI. If analog facilities are used, the Modem Pooling feature must be provided.

Tenant Services

DMI is a partitioned feature on both the System 85 and DEFINITY Generic 2 switches. Since the trunk types (108 and 109) for DMI trunk groups can be partitioned, access to these trunk groups can be dedicated to or shared by an extension partition.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, DMI MOS must use the WCR feature for outgoing call routing.

Hardware Requirements

The origional DMI feature requires a DS1 Interface circuit pack for each T1 Carrier facility to the host. The host must be provided with a matching interface.

For Traditional Modules:

The following hardware is required to support the DMI BOS or DMI MOS on System 85, Release 2, Versions 2, 3, or 4, or on DEFINITY Generic 2 traditional modules:

- ANN11D or ANN11E DS1 Interface Circuit Pack

  These circuit packs are used in place of the earlier ANN-11C to support the DMI BOS signaling format.

- ANN35 ISDN—PRI Port Circuit Pack

  The DMI MOS interface uses the ANN35 circuit pack to provide D (data) channel messaging and support ISDN levels 1 and 2 protocol termination. See the ISDN—PRI feature for more information on the ISDN layered protocol.

For Universal Modules:

The following hardware is needed to provide DMI BOS or DMI MOS on a DEFINITY Generic 2 universal module:

- TN767 DS1 Interface Circuit Pack

  This circuit pack provides T1 carrier service connectivity for the universal module. It is functionally equivalent to the ANN11D on the traditional module.

- TN555 DS1 Packet Adjunct.

  This is used in conjunction with the TN767 when the MOS version is being used.

Feature Administration

On System 85 switches, the DMI feature is administered using the MAAP (Maintenance and Administration Panel) or SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The original (BOS) version of DMI is administered with the same procedures used to administer the DS1 Interface feature. The DMI MOS version is administered like the ISDN—PRI feature.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
DIGITAL MULTIPLEXED INTERFACE

PROCEDURE	WORD	PURPOSE	SMT
100	1	For System 85 switches: Assigns trunk group translations including trunk types. For DEFINITY Generic 2: Assigns trunk group dial access codes, trunk type, and dial access restrictions. The applicable trunk-type encodes are as follows: ***108*** DMI wink in / automatic out for System 85; wink in / immediate out or message-oriented for DEFINITY Generic 2) ***109*** DMI (wink in / wink out for System 85; wink in and wink start / delay dial out or message-oriented for DEFINITY Generic 2) ***120*** ISDN dynamic (message-oriented).	No
100	2	For System 85 switches: Assigns digital trunk group characteristics such as data rates. For DMI MOS trunks, the following characteristics are applicable: ● Data Rate = 64 Kbps ("1" in Field 2) ● Asynchronous ● Full Duplex. For DEFINITY Generic 2 switches: Assigns Bearer Capability Class of Service.	No
100	3	Specifies signaling for DMI MOS trunk groups	No
101	1	Assigns trunk group translations (e.g., use of battery reversal, signaling, CDR, etc.) and assigns DMI MOS trunk groups as AVD.	No

(Continued)

ADMINISTRATION PROCEDURES DIGITAL MULTIPLEXED INTERFACE *(Continued)*			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
103	1	Assigns trunk group translations for network trunks (for example, Data Protection, FRL, etc.). For ISDN (DMI MOS) trunk groups, the following specific encodes are required: Field 14 = "1" for collect all digits before outpulsing Field 15 (For R2 V4 "0" = voice, or voice grade data "1" = mode 1 data "2" = mode 2 data "3" = mode 3 data.	No
108	1	Assigns the DMI MOS Terminating Test Line telephone digits.	No
116	1	Assigns trunks to DS1 Channels. For DMI MOS applications, Field 11, Interface Endpoint blocks the use of DMI MOS trunks with non-ISDN endpoints.	No
260	1	Assigns DS1 circuits to equipment locations and assigns signaling requirements and transmission type. For the DMI MOS application the following specific encodes apply: ● Signaling — Framing: Field 6 = 1 — 23B+D / 24B: Field 7 = 0 (23B plus D) — 24C / RBS: Field 8 = 0 (24th channel signaling). ● Application: Field 14 = 1 (for BOS) or 5 for (MOS).	No
262	1	Administers DMI **MOS** specific options such as: ● Interface type ● ISDN facility test type ● TEI (Terminal Endpoint Identifier) ● ISDN level 2 protocol parameters.	No
262	2	Assigns backup D-channel. (Needed only if the NFAS option is used for DMI-MOS ports. See the ISDN—PRI feature for a description of NFAS.)	N / A

(Continued)

ADMINISTRATION PROCEDURES DIGITAL MULTIPLEXED INTERFACE *(Continued)*			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
275	1	Assigns system class-of-service features, including CDR, tandem and trunk-to-trunk calling.	Yes
275	4	Assigns additionl system class of service features including ISDN active (for DMI MOS).	Yes
290	1	Displays DS1 trunk interface and SCS circuit assignments (equipment locations) and identification information. To search for ISDN (DMI MOS) specific trunk assignments, encode 25 or 26 is entered in the Port Type field.	Yes
309	1	Assigns trunk groups to ARS plans, patterns, and preferences. Identifies specific ARS routings for ISDN (DMI MOS) applications.	Yes*
309	5	Assigns ISDN parameters to ARS routings with ISDN (DMI MOS) applications. Specific parameters included are: ● ISDN Dynamic trunks ● Network specific facilities ● Bearer capacities (For R2 V4 Only): Mode "0" voice or voice grade data Mode "1" data Mode "2" data Mode "3" data. ● Bearer capability class of service (For DEFINITY Generic 2).	Yes*
321	1	Assigns trunk groups to AAR plans, patterns, and preferences. Identifies specific AAR routings for ISDN (DMI MOS) applications.	Yes
* Display only procedure for the SMT.			

(Continued)

ADMINISTRATION PROCEDURES
DIGITAL MULTIPLEXED INTERFACE ***(Continued)***

PROCEDURE	WORD	PURPOSE	SMT
321	5	Assigns ISDN parameters to AAR routings with ISDN (DMI MOS) applications. Specific parameters included are: ● ISDN Dynamic trunks ● Network specific facilties ● Bearer capacities (For R2 V4 Only): Mode "0" voice or voice grade data Mode "1" data Mode "2" data Mode "3" data. ● Bearer capability class of service (For DEFINITY Generic 2).	Yes*
354	3	Assigns the NPA-NXX Designator for outgoing ISDN messages used with ISDN Calling Number Display.	No

* Display only procedure for the SMT.

Digital Service (DS1) Interface

Description

The DS1 Interface multiplexes 24 digitized voice or data signals onto a single carrier. This feature provides System 85 and DEFINITY Generic 2 with digital connections to other switches such as, System 75 switches, DIMENSION System (FP8, Issue 3) switches, other System 85 or DEFINITY Generic 1 or 2 switches, COS (Central Offices), and toll offices (where direct or bypass access arrangements are available).

The DS1 feature provides an economical alternative to analog trunking arrangements. Often, a DS1 link that is not fully utilized (less than 23 or 24 channels are needed) will still be more economical than conventionl analog trunking arrangements.

DS1 service can replace the following types of analog facilities:

- Data Communications Access (computer access feature) facilities via the DMI (Digital Multiplexed Interface) feature
- Private network tie trunks
  - — CCSA (Common Control Switching Arrangement)
  - — DCS (Distributed Communications System)
  - — EPSCS (Enhanced Private Switching Communications Service)
  - — ETN (Electronic Tandem Network)
  - — Main / Satellite / Tributary Networks
- Public network trunks
  - — CO (Central Office) trunks
  - — FX (Foreign Exchange) trunks
  - — WATS (Wide Area Telecommunications Service) and 800 Service trunks
  - — Remote Access trunks
  - — DID (Direct Inward Dialing) and DOD (Direct Outward Dialing) trunks
- Off-Premisis Extension line.

Remote Groups: The DS1 Interface feature is also used for digital access to remote groups. A remote group is a switch cabinet with line and trunk carriers and circuits, located away from the main switch equipment room.
can be located up to 100 miles from the main switch complex without significant degradation in service due to propagation delay. Remote groups support analog and digital line and trunk interfaces and (where still in use) hybrid line interfaces. The Modem Pooling feature and other trunk type features are not supported in remote groups.

Feature History and Development

The DS1 Interface feature was first introduced with System 85 in Release 2, Version 2. At that time DS1 supported tie trunk service and special private network applications only.

In Release 2, Version 3, the following improvements were introduced:

- An interface to the central office that provides connections for CO, FX, WATS, and DID trunks.
- Remote Access trunks and Off-premises Stations.
- Dedicated Switched Connections.
- Remote Groups.

In System 85, Release 2, Version 4, Issue 2.0 and DEFINITY Generic 2.1, Issue 2.0, a local Stratum 3 external clocking capability is added. This is an optional hardware based modification and is also available for System 85, Release 2, Version 3 and later switches.

The T1 Carrier

The T1 carrier is a standard carrier used by the telecommunications industry for interconnecting digital systems. This high-speed, high-volume digital trunking facility is a 4-wire twisted pair metallic cable that uses the 1.544 Mbps DS1 signaling format. The DS1 format provides for synchronization, control, and maintenance. The T1 carrier is used for the DS1 24-channel signaling format to distances of up to 100 miles.

Alternative Carriers

The DS1 Interface feature is not limited to use with the T1 carrier. Alternative carrier vehicles include Fiber Optic Links and Microwave Transmission.

Fiber Optic Links

Fiber optics provide the benefit of a wide band carrier that is immune from local RF (Radio Frequency) interference. A single fiber optic carrier can support several DS1 channels. The RF immunity of this carrier is highly advantageous where a high level of RF signaling is present and high-speed data transmissions are used.

Microwave Transmission

Microwave provides a quick and easy way to support digital service in areas where right-of-way problems exist or where laying continuous physical earners is too expensive or impossible. Microwave transmission requires line of sight between stations but can use relay stations to go around corners or over obstacles such as the "curve of the earth." DS1 compatible microwave equipment is readily available that can support from 24 to 672 channels (the equivalent of from 1 to 28 T1 carriers). The best carrier alternative is a matter of the economics applicable to a specific situation. This can only be determined through case-by-case applications engineering.

The DS (Digital Signal) Numbers

Digital Signal numbers correspond to the digital transmission rates used

- DS0 = 64 Kbps (1 channel)
- DS1 = 1.544 Mbps (24 DS0 channels)
- DS2 = 6.312 Mbps (98 DS0 channels)
- DS3 = 44.736 Mbps (672 DS0 channels), and so on.

DS1 Interface combines 24 DS0 signals (24 x 64 Kbps = 1.536 Mbps) with 8000 spare used for synchronization and control onto one carrier (frequently a T1 earner). When carriers are used that accommodate higher rates than DS1, adapters are used to further multiplex the output of multiple DS1 interface units onto the higher speed carrier. For example, the AT&T NDS DR23 microwave system couples the output of up to four DS1 interfaces into its microwave transmitter before broadcasting the signal in microwave form.

Channels as Trunks or Lines

With the basic DS1 Interface feature, the System 85 or DEFINITY Generic 2 switch treats each DS0 channel (separate channel in the DS1 format) as a trunk or line circuit. Each channel is administered as a trunk or line.

Channels administered as trunks are assigned to trunk groups, assigned trunk features, and restricted like conventional trunks. Different trunk types can occupy different channels on the same DS1 Interface (as long as the interface is set up for voice-grade service). A DS1 Interface setup for AVD (Alternate Voice/Data) or DMI (Digital Multiplexed Interface) applications cannot be shared with other applications. Figure 48-1 shows the DS1 Interface feature used in a tie trunk arrangement.

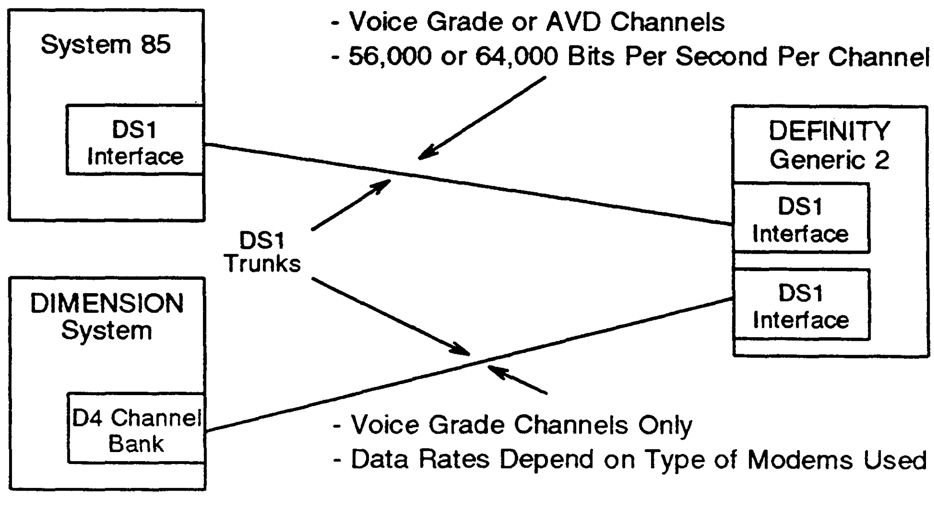

Figure 48-1. DS1 Tandem Tie Trunk Networking Arrangements

Off-Premises Extension Service

DS1 channels assigned as lines are used for Off-Premises Extension Service. This is a special application of the DS1 Interface feature that provides voice grade service to a distant location through a D4 channel bank. For limited scale applications, this service can provide remote service (up to 100 miles) without the expense of additional modules or another switch. Figure 48-2 shows an example of an Off-Premises Extensions application.

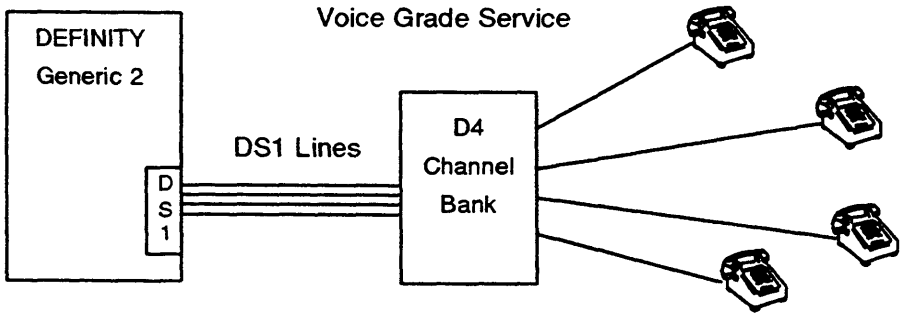

Figure 48-2. DS1 Off-Premises Extensions Application

DS1 Service Arrangements

The basic DS1 interface is set up as either voice-grade service or AVD. Voice-grade service provides 24 channels for transmitting voice and voice-band data signals. The AVD service provides 23 clear channels that can be used for voice or data communications. With AVD service, the 24th channel is used to provide signaling for the other 23 channels.

Voice-Grade Service

The voice-grade service arrangement is used to connect a System 85 or DEFINITY Generic 2 switch to other switches in a private or public networking arrangement. When the DS1 interface is set up for voice-grade service, all 24 channels carry traffic. Data communications is provided on these channels using modems. This can be done with either dedicated modems or through the Modem Pooling feature.

DIMENSION System Connections

Voice-grade DS1 service can be provided between a System 85 or DEFINITY Generic 2 switch and an Enhanced DIMENSION System (Feature Package 8, Issue 3) switch by equipping the DIMENSION System with a D4 Channel Bank. The Enhanced DIMENSION System can support voice-grade data communications with the Modem Pooling feature. Voice-grade is the only DS1 service available to or from a DIMENSION System switch.

Voice-Grade Service Applications

Voice-grade DS1 service can be used for most applications that use conventional tie trunks. These applications include:

- APLT (Advanced Private Line Termination) tie trunks
- CAS (Centralized Attendant Service) arrangements
- CO (Central Office) and FX (Foreign Exchange) trunks
- DCS (Distributed Communications System) tie trunks
- ETN (Electronic Tandem Network) tandem tie trunk switch connections
- Main / Satellite arrangements
- Remote Access trunks
- RLTs (Released Link Trunks)
- WATS Wide Area Telecommunications Service) and 800 Service trunks.

AVD (Alternate Voice/Data) Service

AVD service provides a special capability, end-to-end digital connectivity, not available from conventional analog trunks. This is particularly attractive for data calls between switches because modems are not required. AVD service provides users with access to digital facilities on other switches at higher data rates than are normally available over analog facilities. APs and host computers on other switches can be accessed as easily as those on the local switch. Modems are not needed since the data is not converted to an analog format before transmission.

Alternate use of voice and data is possible during a single call. Data call setup can be used by a voice terminal or a data terminal using the same trunk.

Trunk Routing Features

Both voice-grade and AVD service trunk groups can be used in trunk routing pattern preferences, such as AAR (Automatic Alternate Routing), ARS (Alternate Route Selection), or Route Advance Patterns. However, for switches prior to System 85, Release 2, Version 4, overflow must not be allowed between voice-grade and AVD trunk groups.

For System 85, Release 2, Version 4 and for DEFINITY Generic 2 switches, the addition of a bearer capability code (V4) or bearer capability class of service allows the switch to distinguish between voice grade and data calls. This capability allows voice-grade and AVD trunk groups to overflow because the switch can now discriminate between calls that will and won’t work in an overflow situation.

Other Features Using DS1 Interface Service

Several other features use the DS1 interface with some modification. These features include the following:

- ACCUNET Service Interface Feature
- DMI (Digital Multiplexed Interface) Feature
- ISDN—PRI (Primary Rate Interface) Feature.

Each of these features is discussed in detail in its own chapter of this manual.

56 Kbps Clear Channel Service

This service was developed specifically to support the ACCUNET Service Interface feature but can be used between any group of switches that has an ACCUNET Service Interface capability (System 85, Release 2, Version 3 or later, or System 75, Release 1, Version 2 or later). This service combines the advantages of both the Voice Grade Service and AVD Service. It provides 24 communications channels (like Voice Grade Service) and supports high-speed (56 Kbps) data signals without the use of a conversion resource (Modem (Pooling). See the ACCUNET Service Interface feature in this manual for more information on this service.

Data Preindication

In the System 85 or DEFINITY Generic 2 environment, many off-premises data calls require the use of Data Preindication *(see* the Data Call Setup feature). Exceptions include calls made using the ACCUNET Service Interface feature and data calls that originate on analog facilities. In the DEFINITY Generic 2 environment, any data call originated from a DCP voice terminal requires data preindication.

Data preindication identifies the call as a data call to the switch and reserves a modem pooling conversion resource. The modem pooling conversion resource is released if not needed. An example of an off-premises data call not needing a conversion resource is a DCP data call routed via DS1 AVD service to a DCP end point. Another example is an ISDN—BRI data call routed over ISDN—PRI trunks. A conversion resource is needed for a data call when one or more of the components of the call (end point, switch, interface unit, line, or trunk) is an analog facility. *(See* the Modem Pooling feature.)

Voice-Grade AVD Circuits

The circuits of an AVD tie trunk can be assigned for voice-grade calls, if desired. AVD voice-only circuits are the equivalent of voice-grade tie trunks. When these circuits are used for voice-band data calls, Data Preindication must be used so a modem pooling conversion resource is inserted into the data transmission path.

Supporting Services and Supplemental Arrangements

Clocking and Synchronization System

Timing Source

Timing synchronization is extremely important in digital networks, including DS1 trunking arrangements. If the digital network is not synchronized, switches transmit and receive data bits slightly different rates. This causes receive buffers to either overflow or underflow. When these buffers overflow or underflow by an amount equal to one DS1 frame (one data word for all 24 channels), an entire frame is repeated or deleted to make up for the difference in transmission rates. This is called a "slip." Note that a slip is different from and in no way related to a misframe or framing pattern.

Even though switches at each end of a DS1 span have their own high accuracy clocks, these separate clocks will still produce slightly different frequencies. This will eventually result in slips. To avoid this source of slips, a master-slave arrangement is set up so that one end of the DS1 span controls the timing for both ends.

The Synchronization Hierarchy

The AT&T digital network has an elaborate timing hierarchy in place that provides four different stratums or levels of timing. Each stratum (AT&T node) from 1 to 4 has a progressively less accurate clock.

- Stratum 1

  Stratum 1 is a synchronized network of high accuracy distributed and synchronized clocks that controls timing for the AT&T digital networks.

  A Stratum 1 clock is required to have a minimum accuracy of plus or minus 0.00001 ppm (parts per million) or roughly, three ten thousandths (0.0003) of a second allowable error per year.

- Stratum 4

  Nodes in a private network (such as a System 85 or DEFINITY Generic 2 switch) are traditionally considered to be Stratum 4.

  A Stratum 4 clock may have a minimum accuracy of plus or minus 32.0 ppm or roughly 17 minutes of allowable error per year.

System 85 and DEFINITY Generic 2 switches with a direct DS1 connection to an AT&T Serving Node that has a higher stratum clock (the higher the better), can derive their timing from that higher source. This is done by slaving the private switch timing to the node operating on the higher stratum level. If these slaved switches then have DS1 spans to other switches that do not connect directly to the network or to an equal or higher stratum source, the other switches can in turn use a DS1 span from a directly connected switch to provide their timing source.

Figure 48-3 depicts the AT&T digital network timing synchronization hierarchy.

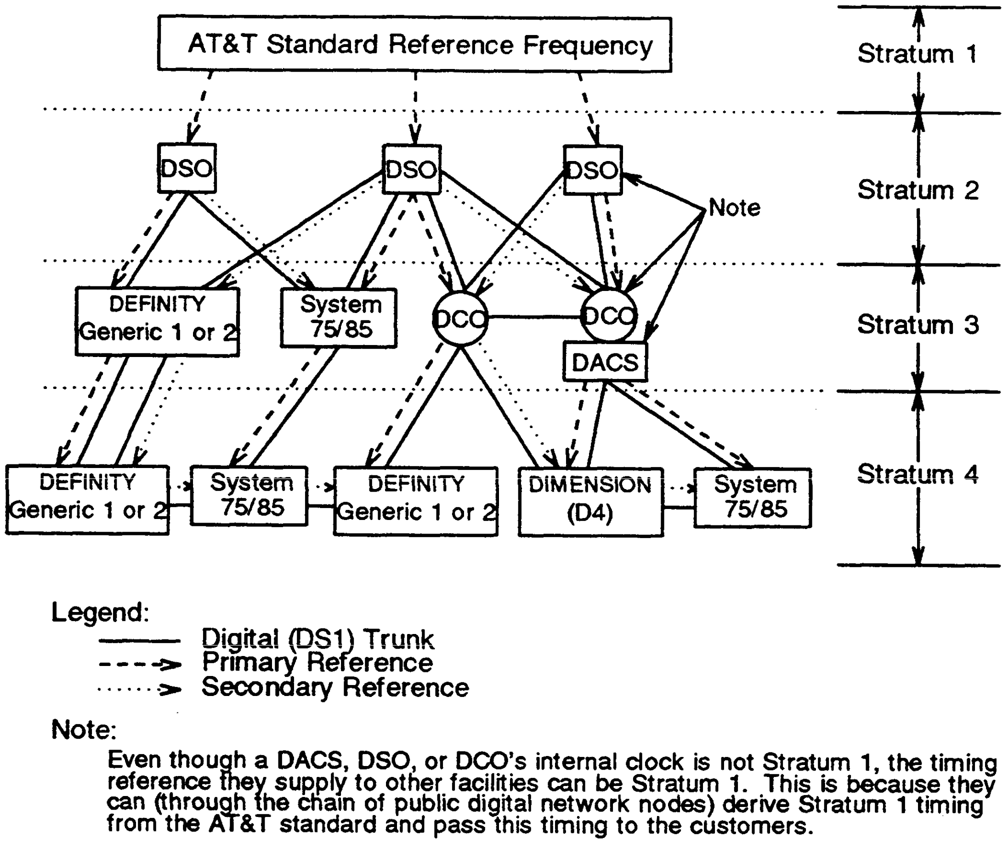

Figure 48-3. Synchronization Hierarchy and Stratum Levels

Using Primary and Secondary Sources

Switches that have more than one DS1 span to their external source (or an equivalent stratum level source), should use one DS1 span as the primary (slave) and another DS1 span as a secondary (slave). This arrangement is shown in Figure 48-4. Such an arrangement provides redundancy in clocking arrangements and reduces the chance of a loss of the timing and synchronization source.

Switching Timing Sources

If the primary timing source is lost or becomes unstable (too many slips), the switch will automatically convert to a secondary source. If both the primary and secondary sources are lost, the switch automatically converts to derive timing from its own high accuracy clock.

The SCS (System Clock Synchronizer): In a standard slaving arrangement, the switch looks for timing from the SCS circuit. The SCS derives its timing from one or two DS1 or ISDN boards (one primary, the other secondary) which in turn derive their timing from the connected DS1 (or ISDN—PRI) span.

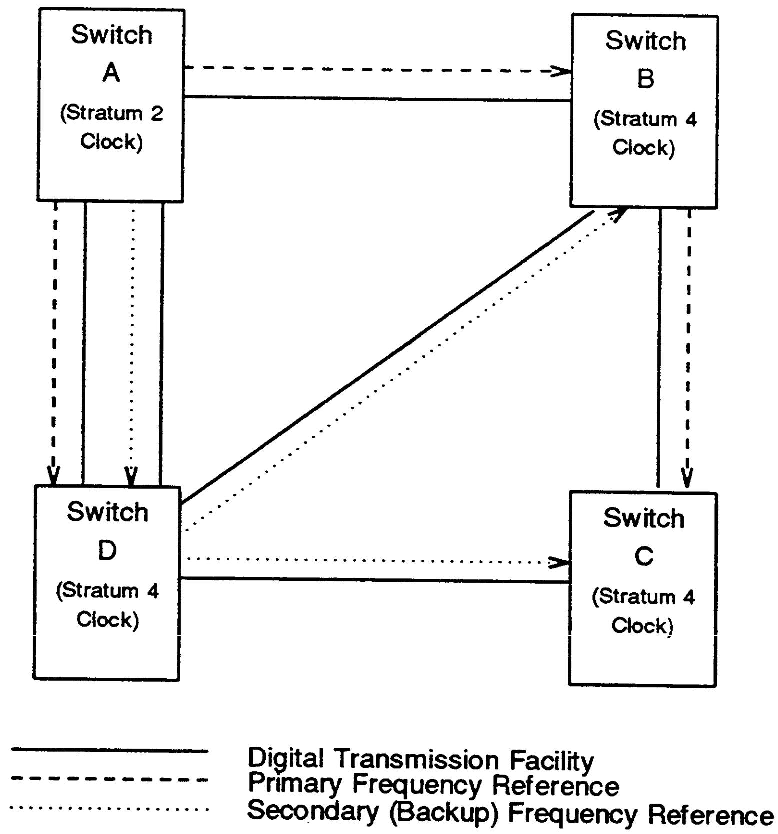

Figure 48-4. Primary and Secondary Timing Sources

In the example shown in Figure 48-4 the following timing arrangements are used:

- Switch A is a high level (stratum 2) source and acts as the master for both switch B and switch D (slaves).
- Switch C does not have a direct DS1 connection to switch A, so it uses switch B for its primary reference and switch D for its secondary reference.
- Switch B has only one DS1 span to switch A so it uses switch D for its secondary timing reference.
- Switch D has two DS1 spans to switch A so it uses one of these spans as a primary reference and the other as a secondary reference source.

If the DS1 span between switch A and switch B goes down, switch B will use switch D for its timing reference. Switch D uses switch A for both primary and secondary reference, but on different DS1 spans, (the same span cannot serve as both primary and secondary). If the DS1 span between switch B and switch C goes down, switch C will use switch D for its timing reference. This redundancy provides a series of alternative timing sources before each switch is forced to rely on internal timing. In this example, all switches synchronize to a stratum 2 source either directly or indirectly.

The Synchronization Clock

With the increasing use of digital facilities on a nationwide basis, it has become necessary to provide a timing synchronization system for switches connected to the network that is more accurate than the traditional stratum 4 clocks used in the past. The standard stratum 4 clock provided with System 85 and DEFINITY Generic 2 switches is actually a stratum 4 type II synchronization system. Table 48-A compares the characteristics of a stratum 3 clock with stratum 4 type I and type II synchronization systems.

TABLE 48-A. Stratum Level System Clock Characteristics

Characteristic	Stratum 3	Stratum 4 Type I	Stratum 4 Type II
Free Run Ability (Accuracy)	4.6 ppm (Parts Per Million)	32 ppm	32 ppm
Pull-in Range	4.6 ppm	32 ppm	32 ppm
Reference Switching	Required	Required	Required
Maximum Time Interval Error	1000 ns (Nano-seconds)	1000 ns	Not Required
Phase Change Slope	61 ppm	61 ppm	Not Required
24 Hour Holdover	.37 ppm	Not Required	Not Required
Hardware Duplication	Required	Not Required	Not Required
External Clock Inputs	Required	Recommended	Not Required
Jitter Filtering	Required	Required	Required

Experience has shown that the standard Stratum 4 (Type II) timing and synchronization system provides timing and synchronization that is satisfactory for digital voice communications and most asynchronous data communications requirements. However, there is an increasing use of high speed bulk data transfer operations. For these uses, the standard synchronization system (the SCS) allows more slippage than is desirable. This results in bit and frame losses that can be extremely expensive for bulk data transfer customers.

For System 85, Release 2, Version 3, and DEFINITY Generic 2 switches, a local, external, Stratum 3 clock is available as an optional hardware adjunct. The Synchronization Clock adjunct connects to the TMS (Time Multiplexed Switch) control on a multi-module switch or the module control on a single module switch. This adjunct provides a sufficiently improved synchronization system to alleviate most, if not all of the synchronization problems currently being experienced and to meet all of the synchronization requirements envisioned for the foreseeable future.

Reliability

Like the standard SCS system, the Synchronization Clock receives incoming timing signals from two selected DS1 (or ISDN—PRI) spans. It is important that the distant end of these selected spans uses a Stratum 3 or higher timing source. Two separate timing sources are used to enhance reliability through redundancy. This same principle of redundancy is used throughout the Synchronization Clock system. (A redundant system, each part of which has a 90% reliability, has a combined reliability of approximately 99%.)

The Synchronization Clock converts these incoming timing signals to two 64Khz composite clock signals which are fed to the External Clock Interface circuit board (TN2131). This circuit board takes the place of the SCS board and is located in the same slots that would normally house the SCS (dependent on switch configuration). The TN2131 converts these 64Khz signals into an 8 Khz format. These 8 Khz clocking signals are then fed to the internal clocking circuitry of the switch where they are used to maintain timing and synchronization. (Again, everything is done in redundant pairs to provide enhanced reliability.)

Retrofit

All communications between the Synchronization Clock system and the switch are accomplished by hardware bridging. This has the advantage of *not requiring* any software changes or administration. It is this advantage that allows the Synchronization Clock adjunct to be easily used with older switches (System 85, Release 2, Version 3).

Administration

No special administration is required for the synchronization clock; however, there are some administration considerations.

- First, the SCS (System Clock Synchronizer) ***must not*** be assigned. This means that Field 11 of Procedure 250 is set to "0".
- Also, all DS1 Interface Circuits (ANN11 series and ANN35s) must be administered for "No SCS." This is done by setting Fields 12 and 13 of Procedure 260, Word 1 to "0". This causes all DS1 and ISDN—PRI circuits to derive their timing from the internal clock, which is itself driven by the Synchronization Clock (stratum 3).

See the Hardware section for additional details on the Synchronization Clock option.

CEMs (Channel Expansion Multiplexer)

In some instances, the 24 channels provided by a DS1 facility (for example, a T1 carrier) may not be sufficient for all the voice and data services required between two switches. In these cases, either another DS1 facility can be added (for 24 more channels), or a CEM could be used to increase the capacity of the existing DS1 facility. The CEM provides a tradeoff between data rates supported and the number of channels provided by increasing the number of channels available. This is done by compressing voice-band channels appearing at the input or multiplexor side. The CEM converts the basic 64 Kbps channel into two 32Kbps channels. Voice and voice-band data signals that operate below 4.8 Kbps can be carried on these compressed channels. High-speed data channels, operating at or above 4.8 Kbps, cannot be compressed. They require the full bandwidth of the basic DS0 channel to transmit data properly. A high-speed channel passes through the CEM untouched, using twice the space of a compressed channel. The CEM can only be used for point-to-point communications because a CEM must be used at each end of the carrier to provide both multiplexing and demultiplexing functions (Figure 48-5).

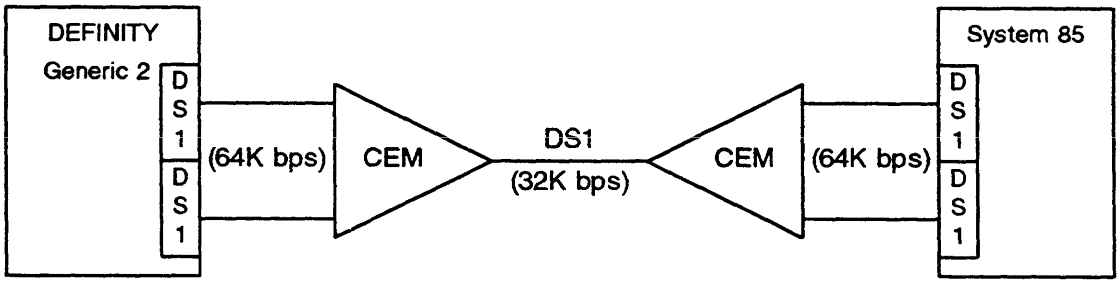

Figure 48-5. CEM (Channel Expansion Multiplexer) Configuration

Two System 85 DS1 Interfaces connect to the input (switch side) of the CEM. The output of the CEM connects to the T1 carrier. The T1 carrier then extends to the carrier side of the CEM at the other switch. With both DS1 Interfaces designated as voice-grade only, 48 compressed channels can be transmitted over one T1 carrier. When a DS1 AVD Interface is used as one input, the AVD signaling channel and each high-speed data channel require the same space as two, adjacent compressed voice-grade channels. The 24 DS1 Interface channels on each CEM input can be compressed in one of two ways: VBR (Variable Bit-Robbed) Signaling or 384 Kbps Bundling. VBR Signaling provides the greatest capacity with the trade off being a lower quality of transmission. This may be acceptable, but it increases the capacity by only two to four channels, depending on types of DS1 Interfaces.

VBR (Variable Bit-Robbed) Signaling Format

VBR signaling provides up to 48 channels when both DS1 Interfaces used are voice-grade only service. Voice-grade channels are encoded using 8000 samples per second, with each sample containing 4 bits. This results in a bit rate of 32 Kbps (8,000 samples per second mu 4 bits per sample = 32,000 bits per second). This is half the rate of the DS0 channels; therefore 48 rather than 24 channels can be placed on a T1 carrier. When AVD interfaces are involved, the full rate (64 Kbps) data channels must be untouched, thus reducing the capacity of the carrier by two for every data channel. To coordinate the transmission of each channel to its destination at the other end, a signaling channel must be provided. This is where we get the term bit-robbed. Every sixth channel sample is robbed of a bit providing a 24 Kbps signaling channel. Every sixth voice sample is encoded using 3 bits instead of the normal 4 bits. This tends to degrade the transmission quality of the voice-band channels. If the extra capacity that VBR signaling provides (two to four channels more than the 384 Kbps Bundling format) is worth the loss in transmission quality, then VBR signaling is a viable method of voice-band channel compression.

384 Kbps Bundling Format

The 384 Kbps Bundling format provides up to 44 compressed voice-band channels with no degradation in transmission quality. Signaling to coordinate the voice-band and data channels is provided by separate channels, so bit-robbing is not necessary. The voice-band channels of the DS1 Interface are formed into bundles of 11. One DS1 Interface channel must be used to provide CEM signaling for the bundle. This signaling channel operates in the voice-band so it too can be compressed. Thus, each bundle requires 12 channels. This means that two 12-channel bundles can be provided for each DS1 voice-grade only interface for a possible 44 usable channels of input to the CEM. If AVD interfaces are used, one full rate, uncompressed DS1 facility channel is required for each AVD signaling channel and each full rate data channel. Thus, for each AVD interface, the number of usable channels is reduced by two (for the AVD signaling channel). Each full rate data channel further reduces the number of usable channels by two. The following illustrates this point:

- CEM Input 1—DS1 voice-grade only interface
  - — Two 32 Kbps signaling channels for 384 Bundling
  - — 22 usable voice-grade channels (also 32 Kbps)
- CEM Input 2—DS1 AVD interface containing one high-speed data channel
  - — One 64 Kbps signaling channel for AVD
  - — One 64 Kbps usable data channel
  - — Two 32 Kbps signaling channels for 384 Bundling
  - — 18 usable voice-grade channels (also 32 Kbps).

The result is 40 compressed voice-band channels and 1 full rate data channel.

Any combination of voice-grade and AVD DS1 Interfaces can be used between two System 85 switches, two DEFINITY Generic 2 switches, or a System 85 and DEFINITY Generic 2 switch. If both interfaces to the CEM are voice-grade, DIMENSION 600/2000 Systems with FP8, Issue 3 can also be used. The CEM interfaces with a D4 Channel Bank on the DIMENSION System. The DIMENSION System is not equipped to handle the AVD interface. If full rate data channels are used they should appear on the same AVD interface, when possible. This might save channels that would be required to provide signaling.

CDMs (Channel Division Multiplexer)

The CDM is used to combine dedicated (nonswitched) and switched channels on one DS1 carrier (Figure 48-5). When a DS1 carrier is only partially filled with switched voice and data channels, the CDM permits the spare channels to be reserved for dedicated point-to-point private line data connections. These channels, once through the CDM, connect directly to a host computer or terminal cluster rather than being switched through the System 85. In a sense, the CDM performs the switching. These channels operate at up to 64 Kbps.

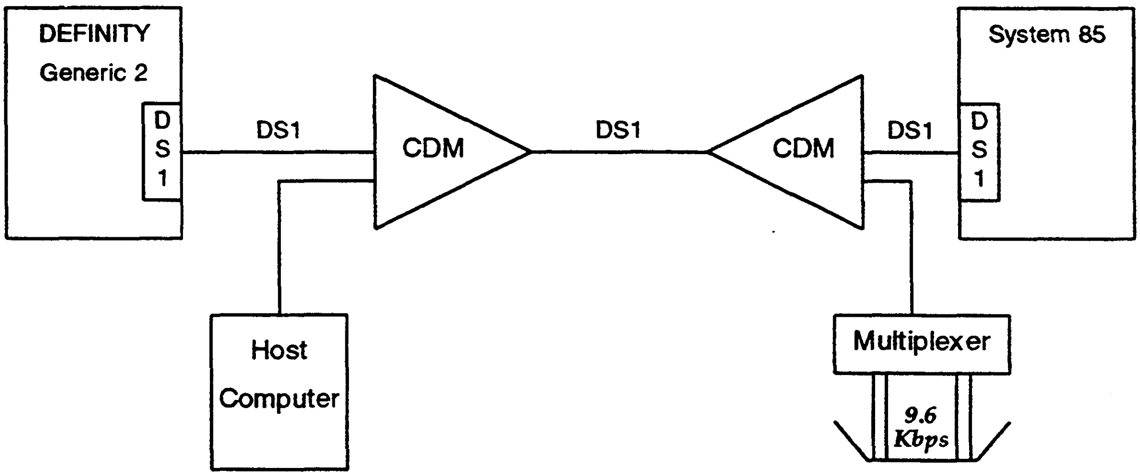

Figure 48-6. CDM Configuration

The CDM combines the partially filled DS1 Interface with any number of data ports. Each data port requires a channel on the DS1 carrier connecting the two CDMs.

DIMENSION System Application

For DIMENSION System applications, the CDM at the System 85 connects to a D4 Channel Bank. The D4 Channel Bank then routes the dedicated data ports to their destination while sending the switched voice and data channels over E&M tie trunks to the switch.

Dedicated Private Lines

The CDM can also be used to provide multipoint dedicated private lines. Figure 48-7 illustrates how this is done. Each output from a CDM is seen as a DS1 carrier. The DS1 signal contains any combination of switched voice/data channels and dedicated data port channels. This signal can be seen by another CDM as a dedicated data port input. The data terminals at Switch B and Switch C can access the host computer resident at Switch A using their own dedicated data link.

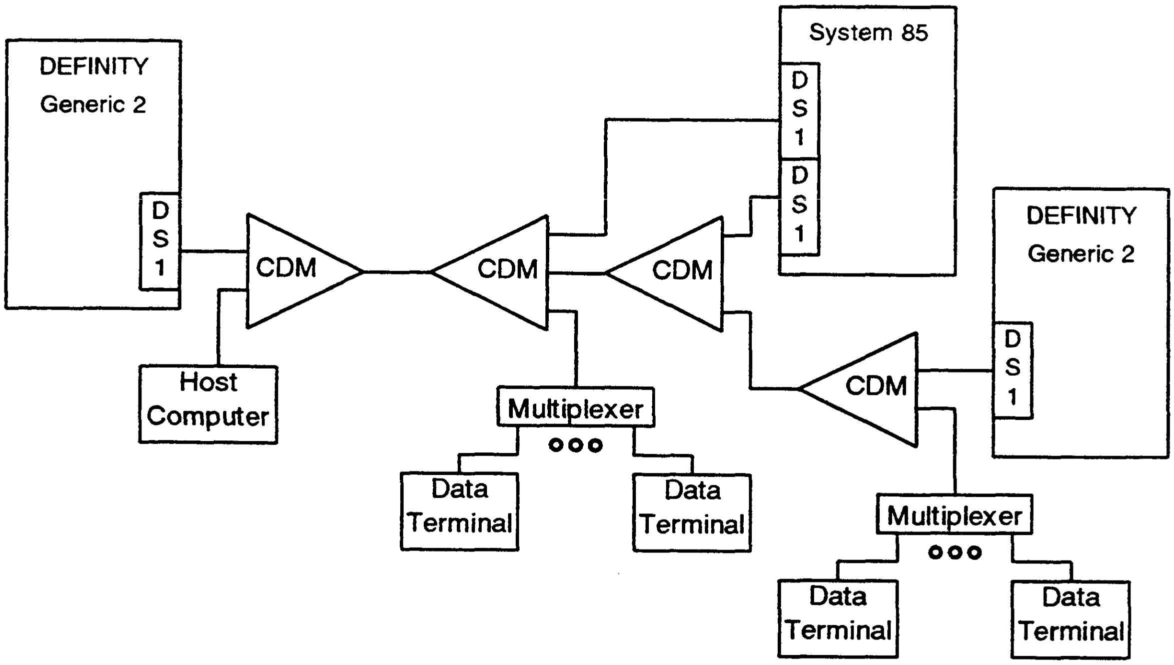

Figure 48-7. CDM Use With Multipoint Dedicated Private Lines

DCS (Distributed Communications System) Environment

CDMs can be used in a DCS environment, providing the DCS signaling information with direct access to a digital facility increases the signaling rate of the DCIU from 9.6 to 19.2 Kbps.

Mixing CDMs and CEMs

CDMs can also be used with CEMs on the same DS1 carrier to provide 24 or more point-to-point switched voice/data channels and dedicated data channels. In Figure 48-8, 40 of the possible 44 switched voice/data channels are combined onto one DS1 carrier by the CEM. The DS1 carrier output from the CEM has two full rate channels available for use by dedicated data channels. These are added at the data port inputs to the CDM.

DS1 Applications

56 Kbps Clear Channel Service: The ACCUNET Service Interface feature uses an adaptation of the DS1 Interface to provide access to and from this 56 Kbps clear channel public network service. This adaptation requires the use of a special data module (the MPDM/M1*) and a modified form of robbed-bit signaling. This application of DS1 is discussed in the ACCUNET Service Interface chapter of this manual.

DS1, Digital Tie Trunk Network: The DS1 Interface feature permits switched digital connections without a modem between digital facilities on two separate System 85 switches. Each DS1 channel functions as a digital tie trunk between the two switches. Alternate Voice/Data channels provide end-to-end digital connections and avoid the need

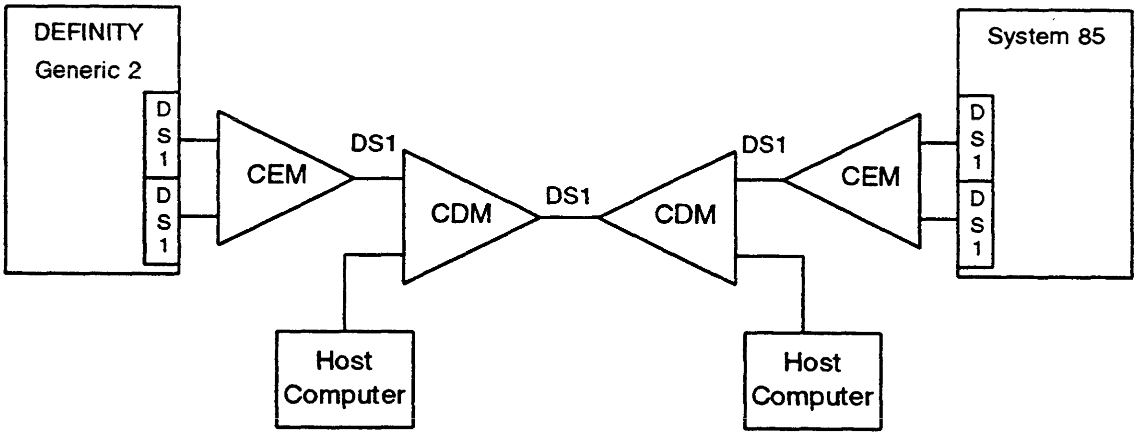

Figure 48-8. CEM / CDM Combined Use Configuration

for format conversion. This reduces hardware costs and makes the full functionality of the digital switching system available to calls between separate switching nodes.

DCS (Distributed Communication Service): A DCS network can be set up using DS1 AVD trunks between the DCS nodes. Such a network provides the same transparency for data traffic that is available to voice traffic over an analog DCS network. Such a network also avoids the need for Modem Pooling on calls between DCP facilities on separate nodes. This permits the use of significantly higher data rates. If a high volume of data traffic is expected, the added cost of AVD trunks (compared to voice-grade service DS1 trunks) may be overcome by the reduced requirement for Modem Pooling conversion resources. Such a system would derive added benefits from the higher data rates available.

Digital to Analog Tie Trunk Network: The DS1 Interface can also be used between a System 85 and Enhanced DIMENSION System. In this instance, the DIMENSION System switching node is analog so a conversion resource is required for data calls. A D4 Channel bank provides the DS1 interface at the DIMENSION System end of the digital tie trunk. The Modem Pooling feature provides the analog interface for the System 85. The conversion resources limit the data rates.

Data Features

The DS1 Interface feature is compatible with System 85 data features. Data calls using the DS1 tie trunk can use data support features such as Data Call Setup, Modem Pooling (where needed), and Host Computer Access.

Trunk Routing Features

System 85 trunk routing and support features such as AAR, ARS, Queuing, Route Advance, and the CDR call monitoring feature are also applicable to calls using the DS1 Interface feature.

Considerations

CEM (Channel Expansion Multiplexing) and Full Rate Data

Since full rate data channels cannot be compressed without destroying information, channel assignments in the CEM must be coordinated with the switch to ensure that only voice-band channels appear on the CEM channels designated for compression. Full rate data channels must be assigned to time slots in the incoming DS1 data stream that pass through the CEM uncompressed.

Channel—Trunk Arrangements

Channels on the same DS1 interface can be assigned to different trunk groups. Also, multiple DS1 interfaces can contribute trunks to the same trunk group. However, voice-grade service DS1 trunks and AVD service trunks cannot be mixed.

Tariff Concepts

When used for direct or bypass access to the AT&T Network, DS1 service can prove economically feasible even when considerably less than the full 23- or 24-channel service may be required.

Mixing Channel Types

In setting up Route Advance and AAR/ARS patterns for DS1 trunk groups, voice-grade and AVD trunk groups must not be mixed. A data call started on a voice-grade trunk group will receive Modem Pooling support. If a routing feature (for example, AAR) changes the call to an AVD trunk group, the conversion resource remains in the call. The terminating switch will not recognize the need for a conversion resource because the call comes in on an AVD trunk group. Modem Pooling support will not be provided and the call will fail. Conversely, calls started over AVD trunk groups and switched to voice-grade trunk groups will fail because the calling switch will not provide Modem Pooling and the terminating switch will. AVD trunk groups can overflow to voice-grade trunk groups only if the network does not carry data calls.

Hard and Soft Processor Swaps

Stable calls over DS1 trunk groups endure a hard processor swap when ESF (Extended Superframe) framing is used. However, calls cannot be placed over DS1 trunk groups during a hard processor swap.

Interactions With Other Features

The following features affect the operation of or are affected by the DS1 Interface feature.

ACCUNET Service Interface

The ACCUNET Service Interface feature uses the DS1 Interface feature as its trunking service. ACCUNET Service Interface is a specific application for the DS1 Interface feature.

Effective in January 1990, the ACCUNET Service Interface feature requires a more precise clocking and synchronization mechanism (Stratum 4, Type I) for subscriber switches with four or more ACCUNET Service Interface connections. The standard clocking and synchronization system, the SCS (System Clock Synchronizer), does not meet these new requirements. Therefore it is necessary for customers who use the ACCUNET Interface feature (with four or more service connections) to also use the stratum 3 Synchronization Clock adjunct.

Dedicated Switch Connections

The DS1 Interface circuits can be used for Dedicated Switch Connections. Each DS1 channel is associated with a software-type equipment location (that is, no real circuit exists for each channel, but System 85 does recognize the channels separately). This equipment location can be assigned as a semipermanent connection to a terminal location or to a DS1 channel equioment location. This allows "cross-connects" to be made between DSOs and DS1s similar to those provided by a DACs.

ISN (Information Systems Network) Interface

When the DS1 Interface feature with AVD (Alternate Voice / Data) trunks or ISDN—PRI trunks are used to link System 85 locations, end-to-end digital connections between ISN stations and remote locations are possible. These arrangements do not require support from the Modem Pooling feature.

ISDN—PRI (Integrated Services Digital Network—Primary Rate Interface)

The ISDN—PRI is similar in many respects to the DS1 Interface and uses an AVD (Alternate Voice / Data)-type carrier circuit like that used with the DS1 Interface feature. The signaling used with these two features is, however, different in that the DS1 Interface feature uses a bit-oriented signaling protocol while the ISDN—PRI feature uses message-oriented signaling. These two features are separate but compatible through the interworking function. That is, traffic that originated on one type of facility can be passed to the other type at a tandeming point. However, ISDN end-to-end connectivity and ISDN features and messaging is lost when this is done.

Last Number Dialed

The LND (Last Number Dialed) feature completely stores and redials the digits dialed during a DS1 call. The LND feature stores the DS1 access code and the following destination number.

Off-Premises Terminal

A single DS1 Interface can provide digital transmission facilities for up to 24 Off-Premises Terminals.

Remote Access

The Remote Access feature is compatible with the DS1 Interface feature. On System 85, Release 2, Version 3 switches, Remote Access via MEGACOM® service requires the use of DS1 facilities.

Tenant Services

The DS1 Interface is a partitioned feature on System 85. If the desired trunk type applies to DS1 service and the specific trunk group is partitioned, access to this trunk group can be dedicated to or shared by an extension partition.

Trunk Group Features

Features such as Route Advance and Queuing that provide more efficient access to trunk circuits can also be used for DS1 access.

Restricting Feature Use

Because the DS1 Interface appears as a trunk group to the system, any restriction that can be applied to a trunk group or to trunk-group access can be applied to DS1 trunks.

Hardware Requirements

For Traditional Modules:

The hardware required to support the DS1 Interface feature on a System 85 conventional module includes the following:

- DS1/73 Series Port Carrier

  The DS1/73 Series Port Carrier is designed specifically to accommodate the circuit boards used for DS1 Interfaces as well as 73 Series standard line and trunk circuit boards. Each carrier supports two boards.

- ANN11C or ANN11E, DS1 Interface Circuit

  The DS1 Interface Circuit (ANN11) is a single circuit pack that supports the full 24-channel interface for the T1 carrier. It provides both the multiplexer and demultiplexer function. The ANN11B was used with earlier versions of System 85 to provide voice-grade and AVD tie trunk circuits, and it supports the "proprietary" 24th channel signaling format of DMI—BOS. The ANN11C provides these functions and also provides the interface to the CO and Dedicated Switched Connections. The ANN11C is used to replace three SN228 circuit packs for the Off-Premises Terminal arrangement.

- TN380C, Module Processor

  The TN380C (Module Processor) must be used in place of the TN380B for all Release 2, Version 3 systems. It can optionally be used with earlier models. There cannot be a mix of TN380Bs and TN380Cs in a duplicated module. There can, however, be TN380Bs in one module and TN380Cs in another module within the same system. The TN380C provides the same functionality as the TN380B and also provides control signaling for Dedicated Switched Connections, control of Remote Groups, and control for DS1 CO interface circuits.

- D4 Channel Bank

  If an analog switch (DIMENSION System) is used in the DS1 networking arrangement, a D4 channel bank is required at the analog node. If data communications are to be carried over these trunks, Modem Pooling is required at each participating System 85.

- DDS Data Port Adapter

  For CDM applications involving DIMENSION Systems, the DS1 carrier terminates on the DIMENSION System via a D4 Channel Bank to provide a DDS compatible data link. In this instance, a DDS data port adapter for the CDM is required. The adapter allows an Office Channel Unit (OCU) to interface with the CDM, providing compatibility with the OCU in the D4 Channel Bank. The data ports are limited to 56 Kbps.

For Universal Modules:

The hardware required to support the DS1 Interface feature on a System 85 universal module includes the following:

- TN767, DS1 Interface Board

  The TN767, DS1 Interface board is used with the Release 2, Version 5, universal module. The TN767 contains 24 port circuits that are combined to form a single, 1.544 Mbps DS1/T1 link. The TN767 is functionally equivalent to the ANN-11 series circuit packs used with the conventional System 85 modules.

- TN555, DS1 Packet Adjunct Board

  The TN555 is required with the TN767 only when message-oriented signaling is needed (ISDN—PRI or DMI—MOS spans with a D-channel).

Optional Hardware:

Three hardware options are available to increase the functionality of the DS1 interface:

- Clocking Synchronization System

  Digital trunking requires a clocking synchronization system. While this is not optional as such, the form that it takes is optional. Two clocking synchronization systems are available:

  — TN463 SCS (System Clock Synchronizer)

  The SCS (System Clock Synchronizer), TN463, provides Stratum 4 (Type II) clocking and synchronization. The SCS is available to all System 85 and DEFINITY Generic 2 switches with the DS1 Interface feature. In a single-module system, the SCS resides in the module control carrier. In a multimodule system, the SCS resides in the TMS control carrier.

— External Stratum 3 Clock Option

This option is available for System 85, Release 2, Version 3 and later switches and for DEFINITY Generic 2 switches. It provides a local, external, Stratum 3 clock interface. This option consists of the following:

- TN2131, External Clock Interface Board

  Replaces the TN463 SCS (System Clock Synchronizer) Board and mounts in the switch cabinet and slot that would normally house the SCS. This varies depending on the switch vintage and configuration.

- Synchronization Clock, J58909A.

  This unit is available only in a stand alone, duplex (duplicated) configuration, mounted in an AUDIX small cabinet.

  This configuration is available in either an AC-powered or a DC-powered version.

- Connecting Cables

  The Synchronization Clock system requires the use of three special connecting cables. The specific cables required depend on the switch configuration

  Multi-Module TMS Control or Single Module Traditional Module Control

  H-600-260

  H-600-274

  H-600-293.

  Single Module Universal Module Control

  H-600-271

  H-600-274

  H-600-293.

- The CDM (Channel Division Multiplexer)

The CDM was first available with System 85, Release 2 and the DIMENSION Systems. The CDM enables the customer with DS1 applications where the switched voice and voice-band data requirements are less than 24 channels to use the extra channels for nonswitched (dedicated point-to-point) data transmission. Because the CDM is compatible with the D4 channel bank, a connection between a System 85 or DEFINITY Generic 2 and a DIMENSION System switch is also possible. The CDM supports both bit-robbed and common channel signaling and provides the following interfaces to the DS1 trunk facility.

- 4-wire E&M (analog tie trunks)
- RS-232 or RS-422 0 to 19.2 Kbps asynchronous data)
- RS-449 or V.35 (56 Kbps asynchronous data)
- RS-449 or V.35 (64 Kbps wide-band data)
- Digital Data System Data Port Adapter (2.4, 4.8, 9.6, or 56 Kbps).

For the 4-wire E&M interface, the CDM accepts an analog input signal and converts it to the digital format the DS1 trunk facility requires. This option is a cost effective alternative for the DS1 customer who needs only a few voice channels. The CDM can also provide channel bank functionality, host computer to host computer connections for high-speed dedicated data transmission, and DCIU to DCIU connections for DCS (Distributed Communications System) applications.

- The CEM (Channel Expansion Multiplexer).

The CEM compresses two DS1 signals, each with up to twenty-four 64 Kbps channels, into one DS1 signal containing up to forty-eight 32 Kbps channels. Only voice and voice-band data (up to 4.8 Kbps) can be compressed, but 64 Kbps channels of uncompressed data (this includes AVD channels) can be carried on the same DS1 trunk facility with compressed channels. Video and 5-, 8-, and 15-Khz program audio signals can also be carried at their normal bit rates without compression. Because a CEM must be provided at both ends of the DS1 facility, it can only be used for a point-to-point or tandem network configuration. The CEM is comptible with either bit-robbed or common channel signaling.

A hardware option provides a 384 Kbps Bundling signaling format for the CEM. This format uses one channel out of every 12-channel bundle to carry signaling information for the rest of the bundle, thus reducing the capacity of the CEM to 44 channels. However, these channels have an enhanced performance level over the normal in-band signaling format which may cause signal degradation when voice-band data calls tandem through one or more intermediate switches.

Feature Administration

The DS1 Interface feature is installed on a per-system basis. Administration within the system is on a per-trunk group basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

The DS1 Interface feature can also be administered using the Manager IV.

The following are the applicable administration procedure.

ADMINISTRATION PROCEDURES DIGITAL SERVICE (DS1) INTERFACE			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns voice terminal (line assignments) to DS1 ports and disables signaling for DSC applications.	Yes
100	1	Assigns trunk-group dial access code, trunk type, and dial access restriction. For System 85, R2 V3 and earlier, also assigns Route Advance. Applicable trunk-types include almost all "true" trunk types (actual switch to switch trunk types). Additionally, a Data-tone detector (trunk type 100) must be present (although not as part of the DS1 interface circuit).	No
100	2	For System 85 switches: Assigns the data rates, synchronization, duplex. For Generic 2 switches: Assigns Bearer Capability Class of Service to trunk groups.	No
100	3	Assigns alternate signaling types to trunk groups when a signaling type other than the default is needed.	No
101	1	Assigns trunk-group translations (for example, use of battery reversal, signaling, CDR) and assigns a trunk group as AVD (tie and OPX only).	No
103	1	Assign trunk-group translations for network trunks (for example, Data Protection, FRL, Bridge-on).	Yes
	1	Assigns trunks to a DS1 Channel, disable signaling for DSC administered channels (ones that use robbed bit signaling.).	No
150	1	Assigns tone-detector trunk circuits to trunk groups.	No
250	1	Assigns the SCS circuit to either the Module Control Carrier (single module system) or the Time Multiplexed Switch Carrier (multimodule system). If the Synchronization Clock is used, the SCS is not assigned (Field 11 is set to "0").	No
260	1	Assigns DS1 circuits to equipment locations and assigns signaling requirements and timing synchronization. If the Synchronization Clock (Stratum 3) is used, Fields 12 and 13 are set to "0."	No
290	1 & 2	Displays DS1 trunk interface and SCS circuit assignment: (equipment locations) and identification information.	Yes

The following is applicable TCM path name used with the AP 16.

TCM SCREEN DIGITAL SERVICE (DS1) INTERFACE	
PATH NAME	**PURPOSE**
terminal-change terminal equipment	Assigns an extension number to the DS1 Interface. The DS1 Interface must first be assigned to an equipment location by the installer.

Direct Department Calling

Description

The DDC (Direct Department Calling) feature provides an economical alternative to DID (Direct Inward Dialing) for departments that receive a high volume of incoming calls. Selected terminal users can be organized into a group and accessed by a listed directory number.

NOTE: The DDC feature is only available through Release 2, Version 1. This feature was replaced and enhanced by the EUCD (Enhanced Uniform Call Distribution) feature in Release 2, Version 2. The EUCD feature provides DDC functionality using the direct hunting option. The following feature description is meant to serve as a reference source for those using the earlier software packages.

A listed directory number links to a DDC group by associating the listed directory number with the extension number of the first terminal in the group. The controlling or primary terminal for the group is the first terminal providing control functions, such as call forwarding for the group.

The switch directs incoming calls for the listed directory number to a group queue. A linear hunt routine extends the queued calls to the first idle terminal. The hunt routine checks the controlling terminal first. If the controlling terminal is busy, the sequence checks the next terminal, etc. If all terminals are busy, the call remains in queue, and the hunt routine runs again in 2 seconds.

This feature reduces call-completion time and eliminates the need for attendant help on incoming calls. Calls to a DDC group can be via DID, non-DID, private switched network, dial repeating or automatic tie trunks, and from local terminals or attendant extended. All calls to the group extension, other than attendant originated, are initially directed to the queue associated with the group.

Each terminal (including the controlling terminal) in a DDC group can receive calls either as a group member or as an individual terminal. For calls within the switch, a unique group extension number (an associated extension number) identifies the group. For incoming calls, the type of trunk identifies the DDC group. For automatic incoming type trunks, the call routes to the assigned DDC group. For dial repeating type trunks, the call routes to the group dialed. This is similar to the way DID calls complete to individual terminals (including individual members of a DDC group).

When there is a delay in completing the call, an optional recorded announcement informs the calling party of the delay.

An individual group member can make his/her terminal available or unavailable for group calls. The controlling terminal can make the entire group unavailable for group calls.

Each voice terminal in a group is assigned a unique extension number. Calls to these extension numbers are serviced as normal voice terminal calls. Therefore, it is possible to call any individual group member directly.

Recorded Announcement

When there is a delay in completing a call, an optional recorded announcement informs the calling party of the delay.

This delay announcement can be repeated (Procedure 275, Word 4). When repeated, the calling party hears silence between cycles on the announcement machine.

Feature History and Development

This feature was first available on System 85 in Release 1. This feature was replaced in Release 2, Version 2 by the EUCD feature.

> **NOTE:** Refer to Appendix B for a tabular comparison of the various call distributors provided in Release 2 System 85 and DEFINITY Generic 2.

User Operations

The following are the user operating procedures for this feature.

To Busy-Out an Individual Group Member:

1. Go off-hook. [Dial tone]
2. Dial the DDC Terminal Busy access code. [Confirmation tone]
3. Go on-hook.

To Busy-Out an Entire Group From the Controlling Terminal:

1. Go off-hook. [Dial tone]
2. Dial the DDC Group Busy access code. [Confirmation tone]
3. Go on-hook.

Considerations

Group Limits

If the switch uses DDC ***and*** UCD (Uniform Call Distribution), the memory size of the switch and call traffic requirements determine the number of combined groups and voice terminals per group. The switch provides up to 28 groups with a maximum of 40 voice terminals per group. An extension number cannot belong to both a DDC and a UCD group; the groups are mutually exclusive.

Stop Hunt Option

A stop hunt option is available to group members. This options prevents calls forwarded to an individual extension from hunting.

Status Indicators

Status indicator lamps are available as queue warning level, trunk status, or system reload indicators. These units can be desk or wall mounted.

Queue Warning Lamp: A queue warning level lamp lights when the number of calls waiting in queue to be answered exceeds a preset warning level between 1 and 31.

Trunk Status Lamps: As many as 128 status lamps can be provided with the switch to display the status of trunks terminating to UCD or DDC groups. Each status lamp shows three separate trunk states: busy (lamp on), idle (lamp off), and alerting (fluttering lamp).

Automatic Reload Indicator: After a power interruption, an automatic reloading of switch translations from the tape into memory occurs. When this happens, the system reload indicator lights.

Effect of Automatic Reloading

When an automatic (or manual) reload occurs, all group members are unavailable for group calls. Each group member must dial the DDC terminal idle access code in order to receive group calls.

Shared Recorded Announcement

If the recorded announcement is provided and there is more than one UCD or DDC group assigned in the switch, all groups use the same recorded announcement.

When the 13A announcement set is used, the Intercept Treatment feature Recorded Announcement option and the delay recorded announcement for DDC can use separate channels on the same announcement machine.

Interactions With Other Features

The following System 85 features affect or are affected by the operation of this feature.

Attendant Call Waiting

An attendant call to a DDC group does not queue. The switch attempts to complete the call ahead of any calls that may be in queue. If no idle line is found in the group, the call waits on the controlling terminal if Attendant Call Waiting is provided. However, when an attendant places a call to a DDC individual terminal and that terminal is busy, the call waits on the busy individual terminal if Attendant Call Waiting is provided.

Attendant Direct Extension Selection With Busy Lamp Field

An attendant can use the appropriate DXS (Direct Extension Selection) buttons to place or extend calls to the listed directory number of a DDC group. However, since a group's queue is never really "busy", the BLF (Busy Lamp Field) lamps adjacent to these DXS buttons are never lit.

Automatic Callback

Any DDC group member can use the Automatic Callback feature. When Automatic Callback is activated toward a DDC group number, callback occurs only when the DDC controlling terminal and calling terminal become idle.

Busy Verification of Lines

The attendant can use the Busy Verification of Lines feature to check the busy/idle condition of a terminal in a DDC group. However, this feature cannot check for a line "made busy" to group calls. When Busy Verification of Lines is activated toward a DDC group number, only the controlling terminal lime is verified (no hunting takes place).

Call Coverage

Coverage cannot be assigned to an associated extension number of a DDC group. However, the switch allows the assignment of coverage to individual extensions of group members for calls addressed directly to that extension.

The switch cannot redirect a call to coverage after the DDC feature has distributed the call to a group member. Therefore, the DDC group must be assigned as the final point in the coverage path. When assigning a DDC group as the final point in a coverage path, the group number is the assigned coverage point.

CDR (Call Detail Recording)

On a DDC call, the trunk-identification number is recorded as the calling number. If the call is completed to an outgoing trunk, the dialed digits are recorded as the dialed number. If the call is to a voice terminal, the extension number is recorded as the dialed number.

Call Forwarding— Follow Me

Call Forwarding—Follow Me, when activated for a DDC group, routes all DDC calls to a designated terminal, the attendant queue, the centralized attendant queue, or to another UCD or DDC group's queue immediately after dialing. If a call is already queued when this feature is activated, the call remains in queue for 7 seconds before forwarding.

- Only the controlling terminal or attendant can activate or deactivate Call Forwarding—Follow Me for a DDC group. Other group terminals cannot activate or cancel forwarding for the group even if authorized by their class of service.
- Only calls to the DDC group number forward when Call Forwarding—Follow Me is active. Calls to an individual terminal or the controlling terminal number do not forward.

- The stop hunt option should be provided in the line class of service for each DDC group extension. Otherwise, a call forwarded to an individual DDC member's extension is treated as a call to the DDC group (hunting is used).

Call Waiting

When Call Waiting is assigned to an individual DDC group terminal, calls to the terminal are allowed to wait if the called terminal line is busy. The group terminal user can be connected to the waiting call by going on-hook; whereby the terminal is alerted and connected to the call upon answer. These calls have preference over DDC calls in queue waiting to be answered. Do not assign this feature to the controlling extension number of a DDC group.

DCS (Distributed Communications System)

In a DCS environment, direct attendant-calls and attendant-extended calls to another node are queued. However, for attendant-extended calls, the attendant does not receive confirmation tone to indicate that the queue has been entered.

Main/Satellite/Tributary

Extension numbers that use extension number steering cannot be the controlling extension for a DDC group.

Priority Calling

When activating Priority Calling toward an individual DDC group terminal, the call waits on that terminal. When activating Priority Calling toward a DDC group number, the call always waits on the controlling terminal (the call does not enter the group queue).

Restriction—Attendant Control of Voice Terminals

If any Attendant Control of Voice Terminals restriction (Outward, Terminal-to-Terminal Only Calling, Termination, or Total) is assigned to a DDC controlling terminal or a DDC group extension, then the restriction assigned is applied to the entire DDC group.

Restricting Feature Use

Performing the busy-out procedure removes the group member from the DDC group.

The controlling terminal can make the entire group unavailable for DDC calls by dialing an access code.

An attendant can restrict access to a DDC group by activating Controlled Termination Restriction for the group number.

Hardware Requirements

The DDC feature requires the following additional or special hardware:

- 30A8 system status indicator panel to display the queue warning status for eight DDC groups
- SN241 contact interface (eight circuits per SN241)
- SN231 auxiliary trunk for delay recorded announcement (four circuits per SN231)
- 13A announcement set to provide the recorded announcement (only one channel can be used for UCD / DDC) or KS-65270 digital announcer to provide the recorded announcement (single-channel announcement set).

or

- KS-65272 4-channel digital announcer to provide recorded announcements. One line circuit of an SN228B, SN229, or TN742 (eight circuits per pack), or one circuit of a TN746 (16 circuits per pack) must be provided for each K-65272 to support remote announcement recording.

Feature Administration

Assignment of the DDC feature is on a per-trunk group basis and on an extension class of service basis.

- The DDC feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

This feature cart also be administered using CSM (Centralized System Management).

The following are the applicable MAAP and SMT procedures.

MAAP and SMT Procedures — Direct Department Calling

Procedure	Word	Purpose	SMT
000	1	Administers the equipment location and class of service of each member's extension.	Yes
000	2	Assigns hunt-to assignments to each member's extension number.	Yes
001	1	Assigns associated extension number(s) to the primary extension.	Yes
010	1	Assigns UCD/DDC membership to an extension class of service and the stop hunt option to a UCD/DDC member's class of service.	Yes
025	1	Administers the characteristics of a UCD/DDC group.	Yes
025	4	Administers UCD/DDC trunk indicator lamps.	Yes
100	1	Administers the trunk type and trunk-group assignments for UCD delay recorded announcement trunk, contact interface trunk, and queuing trunk group. The applicable trunk-type encodes include: 6 Special queue 65 SN241 Contact Interface 68 DDC delay recorded announcement.	No
115	1	Assigns the UCD/DDC group terminations to incoming trunk groups.	No
150	1	Assigns the SN231 equipment location of a delay recorded announcement trunk to its trunk-group number.	No
155	1	Assign the SN241 equipment location of a contact interface trunk to its trunk-group number.	No
204	1	Designates the desired alphanumeric display for UCD/DDC calls that reach an attendant.	No
275	4	Assigns the UCD/DDC delay recorded announcement to the system class of service.	Yes

(Continued)

MAAP and SMT Procedures — Direct Department Calling *(Continued)*			
Procedure	**Word**	**Purpose**	**SMT**
350	1	Assigns the first digit of the feature dial access codes and extension number groups (if required).	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 51 UCD/DDC terminal busy 52 UCD/DDC terminal idle 54 UCD/DDC lamp test 58 UCD/DDC group busy 59 UCD/DDC group unbusy 70 UCD/DDC status toggle.	No
354	1	Administers groups of extension numbers.	No

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Direct Department Calling	
Path Name	**Purpose**
terminal-change terminal display-unit	Displays or prints the group members being monitored by a 106B display unit
terminal-change class-of-service attributes	Assigns UCD/DDC membership to an extension class of service and assigns the stop hunt option to a UCD/DDC member's class of service.
terminal-change extensions attributes	Assigns the extension class of service to an extension number, administers the associated extension numbers to the primary extension, and administers the hunt-to assignment to an extension.
terminal-change group call-distribution attributes	Administers the characteristics of a DDC group. Assigns "terminal" to the type-of-hunting field.
terminal change group call-distribution members	Adds or removes members to/from a DDC group.

Direct Inward Dialing

Description

This feature allows calls from the public network to connect to the dialed extension without attendant assistance. The Direct Inward Dialing (DID) feature can be used to access data stations, local attendants, Centralized Attendant Service attendants, and voice terminals at remote locations without attendant assistance. Extension number steering is used to access terminals at remote locations.

Feature History and Development

This feature was first available in System 85 with Release 1 and has remained unchanged since its introduction.

User Operations

The DID feature provides external access to System 85 and DEFINITY Generic 2 stations. There are no special user operations for this feature.

Considerations

Hard and Soft Processor Swaps

Stable calls from over DID trunk groups endure a hard processor swap. However, calls cannot be received over DID trunk groups during a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Bridged Call

The DID restriction is assigned to an extension class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When the DID restriction is assigned to a shared extension, the restriction applies to every image of the extension.

Call Detail Recording (CDR)

A DID caller attempting to place a call on a tandem switching basis through the switch cannot input the account code. If the call uses attendant assistance, the attendant can input the account code before extending the call.

Call Vectoring

If the DID restriction is assigned to a vector directory numbers (VDN's) class of service, this restriction ***is not applied*** to the VDN's vector processing. Even if this restriction is assigned to the class of service, DID calls are allowed to terminate to the vector.

Data Call Setup

A DID data call can be placed to a data station. If both analog and digital facilities are involved, Modem Pooling will be required.

Foreign Exchange Access (FX)

Using incoming (or 2-way) FX trunks, the DID feature can be used by a caller in the FX CO service area to reach a specific station on the System 85 or DEFINITY Generic 2 switch.

ISN (Information Systems Network) Interface

When an off-net voice terminal is used to set up a data call to the ISN Interface, stage one dialing can be performed from the voice terminal (the ISN Interface access code); however, stage two dialing must use keyboard dialing (that is, dialing the ISN endpoint).

Look-Ahead Interflow

At a sending switch, incoming DID calls can terminate to a VDN with a vector assigned that contains commands for Look-Ahead Interflow. When this is done, these incoming DID calls will interflow normally (that is, according to the commands in the sending and receiving vectors).

At a receiving switch, interflowed calls over ISDN—PRI trunk groups can enter the receiving switch on a DID basis (using digit-oriented routing). To provide this capability, the ISDN SETUP message that is sent by the sending switch contains the digits of the destination VDN at the receiving switch.

Modem Pooling

Modem pooling is required if DID calls over analog facilities are to be placed to a digital interfaced System 85 or DEFINITY Generic 2 data station.

Power Failure Transfer

A DID call cannot be received or used during the power failure transfer mode of operation.

Remote Access

When the switch is in the day mode and Remote Access is sharing LDN service, calls via the remote access number route to the attendant. Otherwise, Remote Access calls are treated like calls from a local (on premises) extension. In either case, the DID feature is not used.

Tenant Services

A DID trunk group can be dedicated to a specific extension partition or attendant partition. When this is done, the switch manager should accurately convey the numbering plan (within the partition that is associated with the trunk group) to the serving Central Office (CO) so that calls from the public network can be routed properly.

> **NOTE:** Using this method, if any DID calls are improperly routed to the wrong partition, the switch will return intercept treatment (reorder tone by default) to the calling party.

To minimize the consumption of DID trunk groups in a partitioned System 85 or DEFINITY Generic 2, the partitioning of DID trunks can also be set up at the serving CO. Under this arrangement, numerous small DID trunk groups at the CO can be administered ***to converge*** to a single large DID trunk group (containing as many as 255 discrete trunks) at the partitioned System 85. This large DID trunk group is assigned to Partition 0 and is shared by various partitions. When this is done, the system manager should coordinate the arrangement with the serving CO so that calls from the public network are properly partitioned.

> **NOTE:** With this type of trunking configuration, the System 85 or DEFINITY Generic 2 has no knowledge of the partitioning arrangement at the serving CO. So, if any DID calls are improperly routed out of the serving CO, the System 85 or DEFINITY Generic 2 would usually allow the call.

Restricting Feature Use

The DID restriction (Procedure 010, Word 3, Field 14) can be used to deny DID access to specific terminals through their class of service. This also prevents these terminals from receiving private network inward dialed calls.

An attendant can temporarily restrict selected stations from receiving DID calls with the Restriction—Attendant Control of Voice Terminals feature.

Hardware Requirements

The Direct Inward Dialing feature requires the following additional or special hardware

For Traditional Modules:

- Every DID trunk requires one circuit of an SN232 circuit pack (four circuits per SN232).

For Universal Modules:

- Every DID trunk requires one circuit of a TN753 circuit pack (eight circuits per TN753).

Feature Administration

Assignment of the DID feature is on a per-system basis.

On System 85 switches, this feature is administered using the Maintenance and Administration Panel (MAAP). The customer can partially administer this feature using the System Management Terminal (SMT).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

Administration Procedures — Direct Inward Dialing

Procedure	Word	Purpose	SMT
000	1	Assigns the class of service to an extension number.	Yes
010	3	Assigns DID restriction to a voice terminal class of service.	Yes
100	1	Assigns the dial access code and trunk type of a DID trunk group. The applicable trunk-type encodes are as follows: 30 Immediate start DID 31 Wink start DID.	No
101	1	Administers trunk-group characteristics for trunk groups administered in Procedure 100, Word 1.	No
150	1	Assigns the SN232 or TN753 equipment location of a DID trunk to its trunk-group number.	No
275	1	Specifies the amount of dial digits forwarded to the switch by the serving central office.	Yes
289	1	Administers the desired type of Intercept Treatment for calls from the public network.	Yes
350	1	Assigns the first digit of the trunk-group dial access codes (if required).	No

Direct Outward Dialing

Description

This feature allows a System 85 or DEFINITY Generic 2 terminal user to access the public network without attendant assistance. Direct access to the public network is rapid and efficient for terminal users and reduces the attendant's work load.

Feature History and Development

This feature was first available in System 85 with Release 1 and has remained unchanged since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Place a Direct Outward Dialed Call:

1. Go off-hook. [Dial tone]
2. Dial a Direct Outward Dialing trunk-group access code. [Second dial tone]
3. Dial the desired public-network number. [Call-progress tone]

Considerations

Access Code

The DOD access code can be from one to three digits. The single digit "9" is most frequently used for this access code.

Feature Availability

Availability of this feature is determined by the number of outgoing public-network type trunks available on the switch. Access can also be a function of other features, such as class of service and Facility Restriction Level, for the extension from which a DOD call is made.

Hard and Soft Processor Swaps

Stable calls over DOD trunk groups endure a hard processor swap. However, calls cannot be placed over DOD trunk groups during a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Data Call Setup

Data calls can be established using DOD if the Modem Pooling feature is also available.

Foreign Exchange Access

Using outgoing (or 2-way) FX trunks, the DOD feature is used by System 85 or DEFINITY Generic 2 users to call stations served by the FX CO over FX Access trunks.

Look-Ahead Interflow

from a sending (or tandeming) switch, calls can interflow over the public network using ISDN—PRI trunk groups assigned as DOD Trunk Types 17, 19, 22, 24, and 27. When this is done, these interflow calls will succeed if every public-network switch between the sending switch and the destination telephone number is ISDN—PRI-capable.

Modem Pooling

The Modem Pooling feature is used with DOD to place data calls to remote data stations over the public network.

Tenant Services

A voice terminal user in a partitioned switch is allowed to place calls outside the switch using a DOD trunk group that is either dedicated to or shared by the user's extension partition. If the user’s extension partition is not allowed to use the dialed trunk group, and if the switch cannot select an allowable alternate trunk group from the Route Advance list (if provided), the switch returns intercept treatment to the calling party.

WATS Access

Using outgoing (or 2-way) WATS trunks, the DOD feature is used by System 85 or DEFINITY Generic 2 users to call stations in the WATS Access service area over WATS Access trunks.

Restricting Feature Use

The attendant can use the following features to prevent a terminal user from accessing DOD:

- Attendant Control of Trunk Group Access
- Attendant Control of Voice Terminals.

To prevent a terminal user from accessing DOD, use the following restrictions:

- Terminal-to-Terminal Only Calling
- Origination restriction
- Outward restriction
- Miscellaneous Trunk restriction
- Code restriction
- Toll restriction.

Hardware Requirements

The Direct Outward Dialing feature requires the following additional or special hardware.

For Traditional Modules:

- Every DOD trunk requires one circuit of an SN230 circuit pack (four circuits per SN230).

For Universal Modules:

- Every DOD trunk requires one circuit of an TN747B circuit pack (eight circuits per TN747B).

Feature Administration

Assignment of the DOD feature is on a per-system basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — DIRECT OUTWARD DIALING			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to an extension number.	Yes
010	1	Assigns a Miscellaneous Trunk Restrictions group to a voice terminal class of service.	Yes
100	1	Assigns the dial access code and trunk type for a DOD trunk group. Route Advance is also administered as required. The applicable trunk-type encodes are as follows. Regular Central Office trunks 17 1-way outgoing DOD 18 1-way out DOD with party test 19 2-way Automatic incoming attendant-completing/DOD. Foreign Exchange trunks: 22 1-way outgoing DOD 23 1-way out DOD with party test 24 2-way automatic incoming attendant-completing/DOD. WATS (Wide Area Telecommunications Service) trunks: 27 1-way outgoing DOD 28 1-way out DOD with party test	No
101	1	Administers the characteristics of a DOD trunk group.	No
102	1	Assigns the trunk group (using the dial access code) to Miscellaneous Trunk Restriction Groups.	Yes
150	1	Assigns the equipment location of an SN230 or TN747B circuit pack to its trunk-group number.	No
202	1	Assigns the desired Direct Trunk Group Selection buttons with busy and warning levels to the attendant console(s).	No
289	1	Administers the desired type of Intercept Treatment for internal calls.	Yes
350	1	Assigns the first digit of the trunk-group dial access codes (if required).	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — DIRECT OUTWARD DIALING	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns a Miscellaneous Trunk Group to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Notes:

Display — Voice Terminal

Description

The Display—Voice Terminal feature provides a visual display of call related and message type information to voice terminal users who have a terminal with display capabilities. The specific form and nature of displayed information depends on a number of factors including the display device itself, and the source and nature of the displayed information.

Feature History and Development

The Display-Voice Terminal feature was first introduced in System 85, Release 1. Subsequent enhancements have included:

- Introduction of the DCS networking capability with Release 2, Version 1 extended the range of useful information available through the Display — Voice Terminal feature.
- Introduction of the AT&T Personal Terminal 510D provided a new terminal capable of using the feature. This terminal was first available in Release 2, Version 2 and can be retrofitted to earlier System 85 switches.
- Introduction of the Z300B Messaging Cartridge, for use with the 7404D VDS (Voice Data Station) and later with the 7406D Digital Voice Terminals. The Z300B Cartridges enables EIA terminals such as the 513 BCT to be use with the Display—Voice Terminal feature. The messaging cartridge was first available in Release 2, Version 3.
- The following enhancements were introduced with the System 85, Release 2, Version 3 switch:
  - Expansion of the Names list from 8,500 names to 32,767 available names.
  - Addition of a scrolling capability to provide for messages of more than 40 characters
  - Introduction of the 7407D IDT (Integrated Display Terminal) provided a reduced cost digital terminal with Voice—Terminal Display feature capabilities.
- System 85, Release 2, Version 4 enhancements included:
  - Introduction of the 7406D with Display provided a display terminal with a somewhat reduced functionality, but with built-in display capability at a reduced cost. These terminals are new with Release 2, Version 4.
  - Introduction of the PC Interface feature provides Personal Computers with the ability to access the Display—Voice Terminal feature. The PC Interface feature is introduced with Release 2, Version 4.
  - Introduction of the ISDN capability extended the range of potentially useful information available through the Display—Voice Terminal feature.

- With Generic 2, expansion of the ISDN capabilities increased both the variety of display terminals available and the source of displayable information. New display capable terminals include the BRI voice terminals 7506 and 7507, the PC/ISDN Platform (in Generic 2.1, Issue 2.0), and the 6500 ISDN Advantage (in Generic 2.1, Issue 3.0).

Sources of Displayed Information

Displayed information comes from a variety of sources including the following:

- The local terminal

  The information most commonly provided from the local terminal is the *called number* or *dialed digits display* while placing a call. This information is generated at the terminal and displayed while dialing is in progress. This includes the display generated by the *Redial* function and *Repertory Dialing* function on terminals with these capabilities.

- The Abbreviated Dialing feature

  Abbreviated Dialing provides display information in two cases. First is the *stored number display* for AD (Automatic Dialing) button stored numbers. Second is the *called number display* when a list stored number is used to place a call.

- The AUDIX (Audio Information Exchange) feature

  For MCS (Message Center Service) subscribers who are also AUDIX subscribers, AUDIX will provide MSC notification to display capable terminals when AUDIX messages are waiting to be retrieved.

- The ACD (Automatic Call Distribution) feature

  The ACD feature provides selected information displays that are of specific interest in an ACD environment. These include City/Queue and VDN of Origin displays, Start/End of Call displays, and Queue Status displays.

- The LWC (Leave Word Calling) feature

  The LWC feature will place display messages in the MCS mailbox when the user is a MCS subscriber and AP (encode 2) is assigned as the LWC destination in Procedure 000, Word 2, Field 9.

- MCS (Message Center Service)

  Provides an attended answering service that can take and relay messages, and provides displayable and retrievable text messages for subscribers with display capable terminals.

- The local switch

  Depending of translations in effect, the local switch provides both *calling* and *called party* display information (for local or on net calls) from the *Names Data Base* and incoming and outgoing trunk identification.

- In an ISDN environment:

  The distant switch (on an off-premises call)

  The distant switch can provide *calling* and *called party* name and number displays and *connected party* information on redirected calls for off-premises calls with an ISDN end-to-end connection (at the switch level). Note that in this case, the terminals do not necessarily need to be ISDN terminals.

The distant terminal

The distant terminal can provide *calling number* information (on off-premises calls) and *user-to-user* type information on ISDN end-to-end calls at the terminal level.

Types of Information Displays

Information displays can be categorized in a wide variety of ways. The two basic categories are:

- Call-Related Displays

  These include: incoming call identification such as calling party name, number, or incoming trunk identification. Outgoing call information such as dialed number, called party name, or outgoing trunk identification. Special call handling information such as calls redirected to coverage or forwarded and elapsed time of call.

- Noncall-Related Displays

  Noncall-related displays include: message retrieval and personal service displays such as date, time of day, and elapsed time other than for a specific call.

The exact form and content of messages, and to some extent the types of messages available, will vary depending on the version of switch being used the way the database is constructed, and the terminal (display device) being used. The examples shown in Tables 52-A and 52-B are typical of a 40-character such as the D401A Display Module. Other types of displays such as 20-character two line and 40-character two line may alter the form of some displays. For example, on a two line display the second line is often used to display supplemental information such as *elapsed time* of the current call while the first line displays call identification information such as *calling* or *called party* name or number. Also, on BRI display terminals with an associated data terminal, the second line of the display is used for messages relating to data calls in progress.

Network Call Displays: Message form and content may also vary on calls between network nodes in DCS (Distributed Communications Service) networks and ISDNs (Integrated Services Digital Networks) based on the different types of switches and network services involved. Note that with ISDN messages, in most cases, it is not the terminal that determines whether or not a message takes on the ISDN form, but rather the network. That is, an ISDN network message can be displayed on either a DCP or ISDN—BRI display capable terminal and will have the same form in either case.

Table 52-A shows displays of call related information while Table 52-B shows displays of typical non-call related information.

TABLE 52-A. Call Related Display Information

Type of Call	Display
Terminating Call Information	
Inside call	a=J D DOE
Attendant call	a=OPERATOR
Trunk call	a= OUTSIDE CALL (or)
	a=WATS
Call to Automatic Call Distribution Agent	a=R JONES to SALES DEPT 12 077
Call Forwarding	b=JACK D LOWE to RUTH JONES f
Call redirected via Send All Calls	d=OUTSIDE CALL to J F DAY s
Call redirected via Call Coverage Criteria	a=J BOGARD to T B ASHE d
Call answered using Call Pickup	a=S HOOVER to K S STEVENS P
Originating Call Information	
Call being dialed	b=572-- (digits as dialed)
Called Party Name	b=Anne H Smith
Trunk calls	b= OUTSIDE CALL (or)
	b=FX-Dallas
ACD Split	
Redirected Calls	
Call Coverage	b=C R Mills cover
Call Forwarding	b=RUTH JONES forward
Conference Calls, Station or Attendant	a=CONFERENCE

NOTE: Notations "a=", "b=", etc., within a display identify specific appearance on the terminal. That is, "c=" means the *third appearance* on the terminal. The ">" symbol shows that there is more to the message being displayed. Use the SCROLL button. Letter notations on the right of the display show reason for call redirection, e.g., "b" for Busy, "c" for cover, and "d" for Don't Answer.

TABLE 52-A. Call Related Display Information (Contd)

Terminating ISDN Call Information

Both ***NAME*** and ***NUMBER IEs*** (Information Elements) are available in the ISDN message set.	a=JONES, TOM R NXX NXX XXXX (Calling Party Name) (Calling Number)
If ***NAME IE*** is not available:	a=CALL FROM NXX NXX XXXX (Local Message) (Calling Number)
If the ***NAME IE*** is received but not the ***NUMBER IE:***	a=JONES, TOM R (Calling Party Name)
If neither ***NAME*** nor ***NUMBER IE*** is available:	a=OUTSIDE CALL (Standard Local Message)

Redirected Local Calls

If ***NAME IEs*** are available:	a=SMITH, MARY B TO JONES, TOM R d (Calling Party Name) (Called Party) (Redirection Type)
If calling party ***NAME IE is not*** available:	a=NXX NXX XXXX TO JONES, TOM R s (Calling Number) (Called Party) (Redirection Type)

Redirected Incoming Calls

A full set of ***CALLING PARTY IEs*** (both ***NAME*** and ***NUMBER)*** on a redirected call. Scrolling is necessary.	a=JONES, TOM R NXX NXX XXXX > a=TO THOMPSON, GEORGE C s

Outgoing Call Displays

Placing an Outgoing ISDN Call

While call is being dialed:	a=NXX NXX XXXX (Dialed Digit Display)
While call is progressing through the network:	a= OUTSIDE CALL (Standard Local Message)

Outgoing Call Displays

Once the Connection Is Complete

If both ***NAME*** and ***NUMBER IEs*** are available:	a=SMITH, MARY B NXX NXX XXXX (Answering Party Name) (Answering Number)
If ***NAME IE*** is not available:	a=ANSWERED BY NXX NXX XXXX (Local Message) (Answering Number)
If neither ***NAME*** nor ***NUMBER IE*** is available:	a=OUTSIDE CALL (Standard Local Message)

Notice that the ISDN displays are similar to displays available on other types of calls. The principal differences include 10-digit calling and called numbers. Also, the remote party name display ***may*** be available, even when the call is not over a DCS (Distributed Communications System) network.

TABLE 52-B. Non-Call Related Display Information

Type of Display	Display
Non-Call Related Information	
Review Stored Numbers	
Automatic Dialing	a=AD=94669613
Last Number Dialed	LND=912242863
Messaging Displays	
Leave Word Calling	Mr T H Thompson 10/16 ll:50a Call 3144
Message	You Have Voice Mail Call 4300
Message Retrieval	WHOSE MESSAGES?
	NO MESSAGES
Message Center Messages	T H DOE 32114 10/16 ll:40am 2 mtg with >
	M R Stone is confirmed for 2:00pm
Deleting a Message	DELETED
Error and Help Displays	
Temporariliy Blocked	MESSAGES UNAVAILABLE NOW - TRY LATER
Display Unit Locked	MESSAGES UNAVAILABLE , SET LOCKED
Time and Date Display	10:23 am FRIDAY SEPTEMBER 20, 1988
or	Feb 29, 1991 2:45 PM
Elapsed Time Display	a=S DOE 3:27:46

Display Modes and Functions

Five Display Modes and five special Display Functions are available for use with the Display—Voice Terminal feature. These modes and functions are administerable and can be assigned to feature/function buttons on the display device or voice terminal, or to administrable "dots" on the AT&T Personal Terminal 510D using Procedure 054, Word 4. In addition, two special security functions, LOCK and UNLOCK, are available. These are not assigned to buttons but rather are setup using dial access codes with Procedure 350, Words 1 and 2. The need for specific modes and functions is dependent on the applications that will be used at each display device. Therefore, with the exception of the basic operational modes and functions, not all will be applicable to each terminal and device.

Display Modes

- Normal Mode

  Provides call-related information for the call currently in progress on the user's terminal (active appearance). Information such as calling or called party identification and calling or called number is displayed. On an internal incoming call, the calling party's name is displayed while ringing.

- Inspect Mode

  Permits the user to display information for an appearance other than the current call in progress on the user's terminal. This mode is not normally used with the AT&T Personal Terminal 510D; the inspect function is provided separately on the PHONE screen.

- Message Retrieval Mode

  Accesses (or retrieves) messages for extension numbers having an appearance on the user's terminal.

- Coverage Message Retrieval Mode

  Allows a covering user to access (or retrieve) a principal's messages.

- Time of Day and Date Mode

  Allows the user to display the time of day, day of the week, and the date.

Display Functions

- Next Message

  Used with the two Message Retrieval modes to step from one message to the next.

- Scroll

  Used with the two Message Retrieval modes to view subsequent segments of messages that consist of more than 40 characters. This function is available only on Release 2, Version 3 and later switches.

- Delete Message

  Used with Message Retrieval mode to delete messages.

- Return Call

  Used with the two Message Retrieval modes to place a call to the party identified by the currently displayed message (for internal calls only).

- Elapsed Time

  Used to start or stop the elapsed timer on the display module. This function is implemented automatically by call control software, on some later model display terminals.

Security Functions

- Lock

  Used to prevent the unauthorized use of a display terminal or device while unattended. This function is activated by a dial access code (which in turn can be assigned to a button) and is primarily intended to block unauthorized message retrieval from a principals terminal.

- Unlock

  Used to deactivate the lock function. This function is implemented by a dial access code and should not be assigned to a button.

Display Capable Terminals and Devices

Voice Terminal Adjunct

One voice terminal adjunct, the D401A 40-Character Display Module, is available for use with the Display—Voice Terminal feature. Figure 52-1 shows the D401A Display Module. The D401A has a single line display capable of showing a maximum of 40 characters. It also has seven, administerable one-lamp feature/function buttons and one fixed function (On/Off) button. The administerable buttons can be used for any of the display modes or functions, or they can also be used for other features and functions assigned to the associated voice terminal.

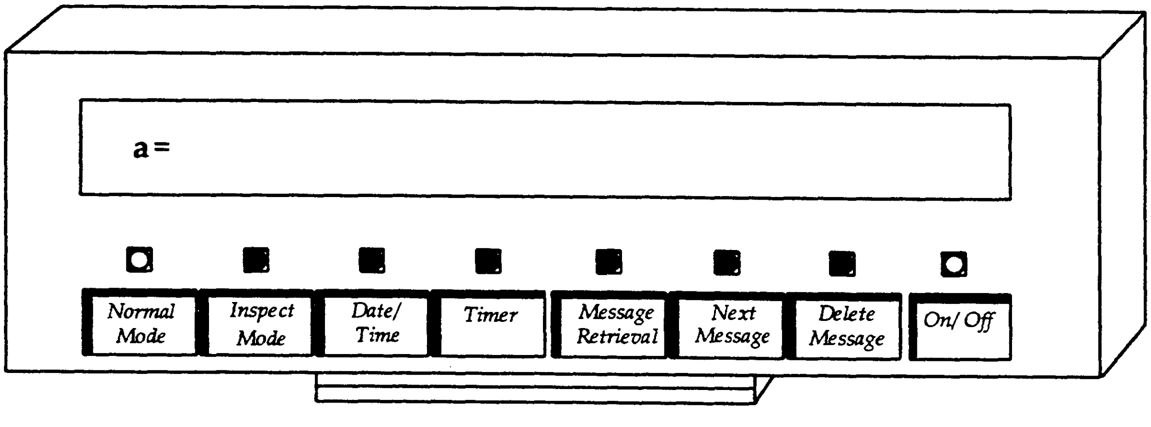

Figure 52-1. 40-Character D401A Display Module and Typical Button Assignments

The D401A is used to provide display capabilities to the 7405D voice terminal and the 7434 voice terminal. The 7405D voice terminal is classified as discontinued availability as of February 1990 and can no longer be ordered, however, terminals on hand can continue to be used.

Display Capable Voice Terminals

Several voice terminals are available with built-in display units. In general, the feature/function buttons on the voice terminal are used to provide whatever display mode and function controls available. In many recent models, basic display modes and functions such as Normal, Time and Date, and Elapsed Time, are provided automatically.

Built in display unit are generally of two varieties, two line displays with 40 characters per line or two line displays with 20 characters per line. Figure 52-2 shows a typical built-in two line display with 20 characters per line.

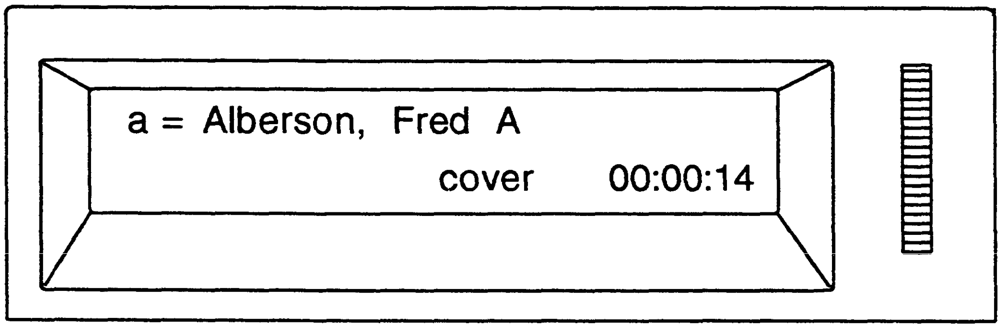

Figure 52-2. Two Line 20 Character Integrated Display

The following are voice terminals, available at time of publication, with a built-in display capability. Other voice terminals can be provided with a display capability (for example, the 7434 with D401A or the 7404D with data termmal and messaging cartridge).

- The 7406D Integrated Display Voice Terminal

  The 7406D With Display is a digital voice terminal with a built-in display. Note that there are two models of the 7406D, one (the -01A) with display and one (the -02A) without a display. The built-in display on the 7406D-01A, is a two line LCD (Liquid Crystal Display) unit with 20 characters per line.

- The 7407D IDT (Integrated Display Terminal)

  The 7407D is a digital voice terminal with a built-in display unit. The built-in display on the 7407D is a two line LCD unit with 40 characters per line.

- The CALLMASTER Digital Communications Terminal

  The CALLMASTER is a DCP multiappearance terminal designed for use in an ACD or Telemarketing rule. The CALLMASTER has an integral 2-line LCD display with 40 characters on each line.

- The 7506 ISDN MDT (Modular Display Telephone)

  The 7506 is an ISDN—BRI voice terminal with a built-in display unit The built-in display provides a two line LDC display with 24 characters per line.

- The 7507 ISDN IDT (Integrated Display Telephone)

  The 7507 is an ISDN—BRI voice terminal with a built-in display unit similar to the 7407D. The 7507 provides a two line LCD display with 40 characters per line.

Voice/Data Terminals

- The 510D

  This is an Executive Model data terminal with a built-in voice terminal. Keyboard and function keys, including Voice Terminal Display function keys, are provided by administrable "dots" on a touch-sensitive screen. The 510D is classified as limited availability and can be ordered only while supplies last. Terminals on hand can continue to be used.

- The 515 BCT (Business communications Terminal

  This is a multifunction data terminal with a built-in digital voice terminal. Voice terminal function keys, including the Voice Terminal Display function keys, are provided on the face of the terminal cabinet. The 515 BCT is discontinued availability and can no longer be ordered, however, terminals on hand can continue to be used.

- 513 BCT and EIA Terminals

  Standard EIA terminals, such as the 513 BCT, can provide a Display—Voice Terminal capability when combined with the 7404D VDS (Voice Data Station) terminal or the 7406D Voice Terminal supplemented with a Z300B Messaging Cartridge. The 7404D is classified as discontinued availability and can no longer be ordered, however, terminals on hand can continue to be used.

Personal Computers

- IBM Compatible PCs (Personal Computers)

  The AT&T PC 6300, AT&T PC 6300 Plus, and many other IBM compatible PCs (Personal Computers) can be used with the Display—Voice Terminal feature when an appropriate application of the PC Interface feature is also used. Several configurations are supported by the PC Interface feature including both DCP and configurations. See the chapter on the PC Interface feature in this manual for more information on these options.

- The 6500 ISDN Advantage

  The 6500 ISDN Advantage is a DCP interfaced PC specifically designed and developed for use in an ACD or Telemarketing role. The 6500 ISDN Advantage couples the 6538/39 PC with a Callmaster, 7406, or 7401 voice terminal to form the 6500 ISDN Advantage Work Station. A specially designed screen display that supports ACD or telemarketing needs. The screen display is shown in Figure 52-3.

```
  Display Info.    |------------------------------|       Auto Mode - OFF

                    Calling Party   Called Party

Selected Call ——→ |------------|  |------------|      Call
                                                    Duration

_a_  _idle_  (PF1)  |-----------|   |------------|  <_:_:_>  Conference  (PF5)
_b_  _idle_  (PF2)  |-----------|   |------------|  <_:_:_>  Transfer    (PF6)
_c_  _idle_  (PF3)  |------------| |------------|  <_:_:_>   Drop        (PF7)
_d_  _idle_  (PF4)  |------------| |------------|  <_:_:_>   Hold        (PF8)

     page  l of 3  __  CALLED  Number  Translation  Table  __  PF14 for more _

  Called      Target    Translation    Secondary  Translation   Screen   Trans
  Number      Window     Keystrokes        Ready  text          Coord    Play PF
  ______       ___      ___________    __________________       _ / _    ______
  ______       ___      ___________    __________________       _ / _    ______
  ______       ___      ___________    __________________       _ / _    ______
  ______       ___      ___________    __________________       _ / _    ______


 -a-    -b-    -c-    -d-    Conf    Xfer    Drop    Hold                 more
 PF1    PF2    PF3    PF4    PF5     PF6     PF7     PF8                  PF24
```

Figure 52-3. 6500 ISDN Advantage Display Screen

User Operations

The following are the user operating procedures for this feature.

Call Identification — Normal Mode

Using a 7407D lDT (Integrated Display Terminal), a D401A display module with 7434 voice terminal, or a 515 BCT:

— If display is not on:

Press **[ON/OFF]** . Display field lights.]

— If display is on, but in another function

Press **[NORMAL MODE]** . [Display field retums to Normal Mode function.]

Using a 510D:

1. Call identification is automatic if the PHONE screen is on in Release 2, Version2 and later switches. If the PHONE screen is not on,
2. Press the **[PHONE]** button on the terminal base. [PHONE screen appears], or
3. If PHONE Screen is active but in another display function or mode, press the **[NORMAL MODE]** dot on the touch sensitive screen. [NORMAL MODE identifier field brightens.]

Using a 513 BCT or other EIA terminal with a 7404D VDS and Z300B messaging cartridge:

— Call identification is automatic if in the Local Mode (equivalent to the ***Normal Mode*** on the D401A Display Module), and not retrieving messages or setting soft options, or in the DATA mode.

— If retrieving messages or setting soft options:

Enter the Exit Command by pressing the **[ESCAPE]** key on the data terminal keyboard. [Screen changes to the Local Mode function menu with calling party identification.]

— If in the DATA mode:

Enter the Local Command by pressing the **[ESCAPE]** key twice, or the **[BREAK]** key once, on the data terminal keyboard. [Screen changes to the Local Mode function menu with calling party identification.]

Call Identification — Inspect Mode

*To Display Information on a Ringing Incoming Call While Active on Another Appearance:**

Press **[INSPECT MODE]** [Display changes to show Information on Incoming Call.]

*To Display Information on a Nonringing Incoming Call While Active on Another Appearance:**

1. Press **[INSPECT MODE]**
2. Press the nonringing call appearance. [Display changes to show Information on Incoming Call.

To Answer an Inspected Call

— If not active on another call:

1. Press **[NORMAL MODE]**
2. Press the call appearance button.

— If active on another call:

1. Press **[NORMAL MODE]**
2. Press **[HOLD]**
3. Press the inspected call appearance.

---

* These procedures are not applicable to the 510D on Release 2, Version 2, and later switches. They do apply to any 510D used on switches prior to Release *2,* Version 2 if an Inspect Mode dot is administered

Message Retrieval

Message Retrieval Without Scrolling

— Using any display capable system except the 513 BCT (or EIA Terminal) with 7404D VDS and Z300B messaging cartridge:

1. Press **[MESSAGE RETRIEVAL]** . [Display showsRetrieval in Progress, Messages for (Name).]
2. Press **[NEXT MESSAGE]** (to display each message).

— Using a 513 BCT or EIA terminal with 7404D VDS and Z300B messaging cartridge from the Local Mode function menu screen:

1. Move the cursor on the function menu screen to the [messages] position.
2. Press the **[RETURN]** key on the data terminal keyboad. [Message Retrieval menu screen appears]
3. Type the letter **[N]** or press the **[RETURN]** key to display the Next Message.

Message Retrieval With Scrolling:

1. Press **[MESSAGE RETRIEVAL]** [Display shows Retrieval in Progress, the Messages for (Name).
2. Press **[NEXT MESSAGE]** (to display each message).
   If message is continued, display shows continuation mark ">" in last field.
3. Press **[SCROLL]** (to display additional segments of displayed message).

   If SCROLL is pressed when there is no next segment, the display will show the next message (same as NEXT MESSAGE button).

To Delete Displayed Message From File

— Using a 513 BCT with 7404D VDS and Z300B messaging cartridge:

Type the letter **[D]** [Screen shows DELETED.]

— Using any Other Display Capable Terminal:

Press **[DELETE MESSAGE]** [Display shows DELETED.]

*To Return a Call During Message Retrieval**

— Using a 513 BCT with 7404D VDS and Z300B messaging cartridge:

1. Go off-hook. [Dial tone]

* To return a call from the message retrieval mode, the extension number of the party to be called must be part of the message.

2. Type the letter **[C]** . [Call-progress tone]

— Using any other display capable terminal:

Press **[RETURN CALL]** . (Extension number must have appeared in the displayed message.)

To Exit Message Retrieval Mode

— Using a 513 BCT with 7404D VDS and Z300B messaging cartridge:

Enter the Exit command by pressing the **[ESCAPE]** key on the data terminal keyboard.

— Using any Other Display Capable Terminal:

Press **[NORMAL MODE]** .

Lock or Unlock Functions

To Lock the Terminal (Blocks Message Retrieval):

1. Go off-hook on an idle line appearance. [Dial tone]
2. Dial the LOCK access code. [Confirmation tone]

   This terminal cannot now be used to perform any Message Retrieval functions.

To Unlock the Terminal (Return Message Retrieval):

1. Go off-hook on an idle line appearance. [Dial tone]
2. Dial the UNLOCK access code. [Confirmation tone]

Display Time and Date

Using the 513 BCT with 7404D VDS and Z300B messaging cartridge on the LOCAL Mode function menu screen:

1. Move the cursor to the [time / date] box.
2. Press the **[RETURN]** key. [Current time, day of the week, month, date, and year, appears on the message line.]

Using any other display capable terminal:

Press **[DATE/TIME]** . [Display shows current time and date.]

To return to normal mode:

Press **[NORMAL MODE]** ,

or

Wait several seconds and the Time-Out function will return the display to the Normal Mode automatically.

Elapsed Time Timer

To start elapsed time display:

Press **[TIMER]** [Timer starts.]

To stop timer and clear time from display:

Press **[TIMER]**

Call Coverage Display Functions

To Access a Principal's Message File as a Covering User:

1. Press an idle appearance button.
2. Press **[COVER MSG RETRIEVAL]** . [Display shows WHOSE MESSAGES?]
3. Dial principal's extension number (when display shows: MESSAGES FOR [Name]:).
4. Press **[NEXT MESSAGE]** . [Message appears on display.]
5. If scrolling is available and the message is continued, press **[SCROLL]** .

To Access a Principal's Message File During a Call With the Principal:

1. Press **[HOLD]** .
2. Press an idle appearance button.
3. Press **[COVER MSG RETRIEVAL]** . [Display shows WHOSE MESSAGES?]
4. Dial principal's extension number.
5. Press the held appearance.

To Return the Call for a Principal:

(Originators extension number must have appeared in the message).

1. Press **[TRANSFER]** [Dial Tone]
2. Press **[RETURN CALL]** [Confirmation Tone]
3. Press **[TRANSFER]** (Principal is connected to message originator's terminal either in a ringing state or after answer.)

Message Sources:

Placing a message is not a function of the Display—Voice Terminal feature. For information or user operations in placing a message, see the appropriate messaging service feature (Unified Messaging, AUDIX, Leave Word Calling, or Message Center).

Considerations

Legal Consideration

Laws governing the use of calling and called number displays differ in different locations and are subject to change. It is the responsibility of the customer's switch administrator to understand and comply with the applicable regulations.

Scrolling

The D401A Display Modules and several voice terminals with built-in displays can show a maximum of 40 Characters on a line. Several voice terminals with built-in displays are limited to 20 characters on a line. When a SCROLL button is administered, stations can display messages of more than 40 (or 20) characters by pressing the SCROLL button. A message continuation symbol ">" appears in the last character field of the display if the message is continued. A long word continuation symbol "->" at the end of the display field means that the word shown on the display is continued in the next message segment. Stations without a SCROLL button will receive an advisory message like "Call Message Center XXXX" if a message of more than 40 characters is waiting. The same switch can support both scrolling and nonscrolling display stations. This function is assigned to stations on an individual basis.

ASCII Character Set

The characters that can be displayed include the full 96-character ASCII character set. This set consists of upper and lower case letters, the digits 0 through 9, and the more common punctuation marks and special characters. Table 52-C shows the characters that are available.

Switch Identification Database

The user can assign names to lines or trunk groups and load this information into the switch database. This is done using the MAAP (Maintenance and Administration Panel), SMT (System Management Terminal), or the TCM (Terminal Change Management) feature, or for Generic 2 switches, the DEFINITY Manager II (see Feature Administration). This allows the calling and called parties to be identified by name or by other identifiers such as department or orginization. The format for these names is optional; however, the maximum length of a name is 30 characters Due to overall memory space limitations, the average length of the assigned names cannot exceed 22 characters.

Truncation of Names

While names of 30 characters or less can be displayed on the 40-character display, there are situations where 2 names must be displayed at the same time (e.g., forwarded calls or coverage calls). This can result in messages that exceed the capacity of the display. To allow for this, the switch provides for the truncation of assigned names down to 15 characters. The default truncation assumes that the first 15 characters are the most significant.

TABLE 52-C. ASCII Character Set

	0	@	P	‘	p
!	1	A	Q	a	q
"	2	B	R	b	r
#	3	c	s	c	s
$	4	D	T	d	t
%	5	E	U	e	u
&	6	F	V	f	v
'	7	G	W	g	w
(	8	H	X	h	x
)	9	I	Y	i	Y
*	.	J	Z	j	z
+	;	K	[	k	{
,	<	L	\	l	|
-	=	M	]	m	}
.	>	N	^	n	~
/	?	O	_	o	PAD

Number of Name Records

As many as 8,500 names can be assigned to extension numbers and trunk groups in switches prior to Release 2, Version 3. In Release 2, Version 3, and later switches, up to 32,767 names can be assigned.

Periodic Compaction of Names Database

In switches with large Names Databases or where frequent changes are made to this database, it is recommended that the database be periodically compacted. This will reduce the likelihood of the Names Database overflowing the allocated memory. Compacting can be done using Procedure 012, Word 3 on the MAAP or SMT, or using the TCM feature on System 85 switches or the DEFINITY Manager II on Generic 2 switches.

Time-Out During Message Retrieval

While in either of the Message Retrieval Modes, the switch sets a timer between each retrieval action. If more than 5 minutes elapse between message retrieval actions, this timer will time out and terminate the session. This could cause the session to be dropped if a callback call initiated during a retrieval session lasts longer than 5 minutes.

Other Voice Terminal Modules

The D401A Digital Display module is compatible with the DTDM (Digital Terminal Data Module) and the Digital Function Key module. All three of these modules can be attached to a 7405D voice terminal at the same time. The 7434 voice terminal does not use either the DTDM or Function Key module. The Call Coverage module and the Display module cannot be installed on the same terminl at the same time. When attached to a 7405D or 7434 voice terminal, both of these modules occupy the same physical space.

Function Buttons

Display function buttons are administrable and can be assigned to any administrable feature buttons available. This applies to basic terminal buttons, buttons on the Function Button module, buttons on the Digital Display module, and administrable dots on the PHONE screen of the 510D with the following exceptions.

- Systems Prior to Release 2, Version 3

  The Normal Mode button has a fixed position on the display module on Release 2, Version 2 and earlier switches.

- The 513 BCT (or EIA Terminal)

  Mode and function selection is provided through screen cursor movement and keyboard entry on the data terminal. The following display modes and functions are available

  — Call Identification

  — Inspect

  — Retrieve Messages

  — Next Message

  — Delete Messages

  — Callback

  — Time / Date.

Message Security

The Lock / Unlock option prevents inadvertent or unauthorized retrieval of messages from a terminal with display capabilities. Dialing the Lock Message Retrieval access code "locks" the Retrieval Mode of the display. When this mode is locked, any press of the Message Retrieval button is ignored. Dkding the Unlock Message Retrieval access code returns the functionality of the Message Retrieval button.

Hard and Soft Processor Swaps

The contents of the Names Database are stored in a translation portion of switch memory. Therefore, these assigned names will endure a hard processor swap.

The Lock / Unlock option for message retrieval on display sets is stored in a status portion of switch memory. Therefore, when a hard processor swap occurs, every display voice terminal will be in the locked mode after the hard swap is finished.

Voice terminal display units do not operate during a hard processor swap. The Display—Voice Terminal feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

When AD (Automatic Dialing) or list access is used on a terminal with display capability, the stored number is displayed. If the names data base is loaded for the number, the associated name immediately replaces the number on the display.

Security of Stored Numbers: If the stored number includes a security code, such as a password or authorization code, this code will also be displayed unless it is protected in the list by the ***suppress function code.*** If the suppress function code is used, the protected characters (except "*" or "#") will be replaced on the display by "s" indicating a character has been supressed. The suppress function code is available on Version 3 and later switches.

For Release 2 switches prior to Version 3, the manual digit entry function can be used to protect security codes. When manual digit entry is used, an underscore "_" is displayed rather than the digits entered. *To Check Stored Numbers*: Beginning with the Release 2, Version 3 Issue 1.2, if a user presses an AD button while the voice terminal is on-hook, the display shows the stored number for the button as "AD= stored number."

Beginning with Release 2, Version 4, if the AD button points to a list stored item (system, group, or personal list), the contents of the list item are displayed.

Differences Between Terminal Types: A noticeable difference occurs in the operation of the Display—Voice Terminal feature *between BRI and DCP terminals* when the Abbreviated Dialing feature is used to place an *outgoing trunk call.*

- With DCP terminals, the dialed number display appears immediately whether the call is dialed from the terminal or the Abbreviated Dialing feature is used to place the call.
- With BRI terminals, the dialed number display appears immediately when the call is dialed from the terminal or the **Redial** button is used. However, when the Abbreviated Dialing feature is used to place an *outgoing trunk* call, there is a noticeable delay (up to 10 seconds or mom) before the dialed number display appears.

AUDIX (Audio Information Exchange)

During message retrieval, the display will notify the user whenever there are also AUDIX messages to be retrieved. If this notification contains the AUDIX associated extension number [or VDN (Vector Directory Number)], the user can press the RETURN CALL button to easily access the AUDIX system.

ACD (Automatic Call Distribution)

When queue-status displays are assigned in the ACD environment, these displays use 8 characters (6 digits and 2 spaces) of the available 20 or 40 characters. Further, these eight characters overlap with the source and destination fields on the 40-character display. Therefore, unless the source and destination names are fairly brief, these names are more likely to be truncated when queue-status displays are enabled.

Call Coverage

If a multiappearance voice terminal user who is busy on call receives a call on another appearance, the new call can redirect to coverage after the don't answer interval. However, when this occurs, the covering user (equipped with a display) to whom the call redirects will receive a redirection notification of "b" (for busy) rather than "d" (for don't answer).

DCS (Distributed Communications System)

Outgoing internode calls in a DCS: The displays shows an outgoing trunk call. Either the digits that were dialed or name of the trunk group, if assigned, is displayed.

Incoming internode calls in a DCS: The display shows the name of the calling party as though the call had originated on the local switch if all switches involved in the connection (the calling and called nodes and all tandem nodes) are System 85, Release 2, Version 2 or later. If other types of switches or other versions of the System 85 switch appear in the connection the specific information shown will vary depending on the types of switches involved.

ISDN—PRI Displays

When both DCS and ISDN—PRI are used in an ETN (Electronic Tandem Network) configuration, the Display—Voice Terminal feature uses the DCS display transparencies instead of the ISDN displays.

ISDN—BRI (Basic Rate Interface)

The Display — Voice Terminal feature works on ISDN—BRI voice terminals in the same way as it does for other display capable terminals. The only user noticeable differences are:

- For the Message Retrieval and Scrolling functions, the response time between displays is longer
- For outgoing trunk calls, the display is initially controlled by the local (terminal) interface. For information displays generated by a source other than the local interface (for example the local or a distant switch), a noticeable delay will occur before information is displayed. For example, on outgoing trunk calls where the Abbreviated Dialing feature is used to place the call, a delay of several seconds may occur before the dialed number display appears.

ISDN—PRI (Primary Rate Interface)

The Display—Voice Terminal feature works with the ISDN—PRI feature through the interworking function.

For DCP terminals with a display capability, message type information provided by ISDN is passed to the Display—Voice Terminal feature software and made available to System 85 and Generic 2 terminals with the display capability. Information displayed is not ISDN direct, but this fact is not readily apparent to the user.

For ISDN—BRI terminals with a display capability, DCP or switch originated information is passed to the ISDN software where it is converted to the appropriate ISDN message format for display on the ISDN—BRl terminals.

Last Extension Dialed

When using the Last Extension Dialed feature to place a call, the called extension appears on the display unit. If the names database is loaded for the called extension, the associated name immediately replaces the number on the display.

Last Number Dialed

When using the Last Number Dialed feature to place a call, the called number appears on the display unit. If the names database is loaded for a called extension, the associated name immediately replaces the number on the display.

If a user presses the LND button while the voice terminal is on-hook, the display shows the stored number for the button as "LND=stored number."

Look-Ahead Interflow

With the excception of the "called party's name", an ACD agent at a receiving switch who answers a Look-Ahead Interflow call (with a display set) receives the normal display for incoming ACD calls. Some sample displays are shown in Table 52-D.

The first field identifies the calling party. If the call routes over DCS facilities, this field contains the calling party’s name or number as provided in the DCS message. Otherwise, this field contains the calling party’s name or telephone number if provided in the Look-Ahead SETUP message. If not, this field contains the name assigned to the incoming trunk group at the receiving switch.

The second field identifies the called party. This field contains the ***original*** called party’s name as provided in the Look-Ahead SETUP message.

Also, if the Look-Ahead Interflow call routes over DCS facilities, the last character of the display shows the "reason for redirection" as provided in the DCS message.

TABLE 52-D. Look-Ahead Interflow Display Information

Type of Call	Display
ISDN call	`a=R JONES to DETROIT CLAIMS`
ISDN call	`a=212-281-7733 to DETROIT CLAIMS`
Default	`a= DETROIT TG to DETROIT CLAIMS`
DCS call	`a=RUTH A JONES to DETROIT CLAIMS      b`

Remote Access

The Voice Terminal Display for an incoming Remote Access call is "OUTSIDE CALL." This display appears even if a different name is assigned to the trunk group in Procedure 012.

Hardware Requirements

The Display—Voice Terminal feature requires one of the following combinations of additional or special hardware.

- Voice / Data Terminals:
  - — The 510D, AT&T Personal Terminal [LA (limited availability)]
  - — The 515 BCT (Business Communications Terminal) [DA (discontinued availability)]
  - — 513 BCT (Business Communications Terminal) or other EIA Terminal with 7404D (DA) or 7406D Voice Terminal and Z300B Messaging Cartridge
- Stand-Alone Voice Terminals:
  - — The 7405D Digital Voice Terminal (DA) with D401A Display Module
  - — The 7406D Integrated Display Voice Terminal
  - — The 7407D IDT (Integrated Display Terminal)
  - — The 7434 Multiappearance Digital Voice Terminal with D401A Display Module
  - — The CALLMASTER Digital Communications Terminal
  - — The 7506 BMT (Basic Modular Terminal)
  - — The 7507 BMT (Basic Modular Terminal)
- Personal Computers
  - — The AT&T PC 6300, 6300 Plus, and other IBM compatible PCs (Personal Computers) can be used with the Display—Voice Terminal feature when a display capable application of the PC Interface feature is also used.

— The 6500 ISDN Advantage

- Applications Processor

  An AP (Applications Processor) with the MSC (Message Center Service) feature is also required if the Scrolling function of the Display—Voice Terminal feature is to be used.

Feature Administration

Assignment of the Display—Voice Terminal feature is on a per-terminal basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Communications System Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — DISPLAY — VOICE TERMINAL			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to an extension number.	Yes
012	2	Administers the name for the extension or trunk group assigned in procedure 012, Word 1.	Yes
012	3	Compacts the Names Database software table to free as much memory space as possible to add additional names.	Yes
051	1	Administers the voice terminal equipment required for display functions and assigns the LOCK and UNLOCK option to the display capability.	Yes
054	4	Assigns the desired display feature buttons to a 7404D, 7405D, 7407D, 510D, or 515 BCT voice terminal, or to a D401A display module.	Yes
261	1	Assigns scrolling capability to AP Messaging	Yes
350	1	Assigns the first digit of the feature dial access code for the LOCK and UNLOCK feature.	No
350	2	Assigns the feature dial access codes. Applicable encodes are as follows: 68 LOCK Message Retrieval 69 UNLOCK Message Retrieval.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — DISPLAY — VOICE TERMINAL	
PATH NAME	**PURPOSE**
terminal-change names compact	Compacts the Names Database software table to free as much memory space as possible to add additional names.
terminal-change names extension-names	Assigns a name to an extension number or a group of extension numbers.
terminal-change names trunk-group-names	Assigns a name to a trunk group or a set of trunk groups.
terminal-change terminal buttons	Assigns the desired display feature buttons to a 7405D and D401A display module, or the 510D, or 515 BCTs. Assigns an automatic message waiting lamp to a voice terminal.
terminal-change terminal equipment	Assigns a display module to a 7405D voice terminal. Also, use this screen to assign the LOCK and UNLOCK option to a voice terminal with display capabilities.

Distributed Communications System

Description

The DCS (Distributed Communications System) is a service designed to meet the needs of customers with telecommunication requirements that exceed the capacity of a single switch. Using a DCS allows the customer to operate and control multiple switches as if they were one switch.

A DCS is not a network in and of itself. Rather, the DCS software provides a messaging overlay to either a Main / Satellite or an ETN (Electronic Tandem Network). Therefore, a DCS is subject to the same intrinsic capabilities and limitations of the type of underlying network used.

A DCS can contain from 2 to 64 separate switches, or nodes, depending on the type (such as, DEFINITY Generic 1 or 2, System 85, System 75, or DIMENSION System) and version of switches used. These nodes can be located in the same building scattered around a metropolitan area, or spread across the country. The DCS can include all nodes of a network or part of a larger network (Figure 53-1).

Figure 53-1 shows a DCS with a single gateway switch to another larger network. Other arrangements are possible. Each node within the DCS could have its own access trunks to the other network if desired. However, the single node gateway arrangement will often be more efficient and economical.

Feature History and Development

The DCS feature was first available with System 85 in Release 2, Version 1. Subsequent enhancements have included the following

- Expanded dialing plan from 4-digit to 5-digit dialing (introduced in Release 2, Version 2)
- Unrestricted 5-digit dialing with extension number portability (first available in Release 2, Version 3)
- ES (Enhanced Services) message set for remote messaging (first available in Release 2, Version 4)
- Unrestricted 4-digit dialing with extension number portability (first available in Release 2, Version 4, Issue 1.3).

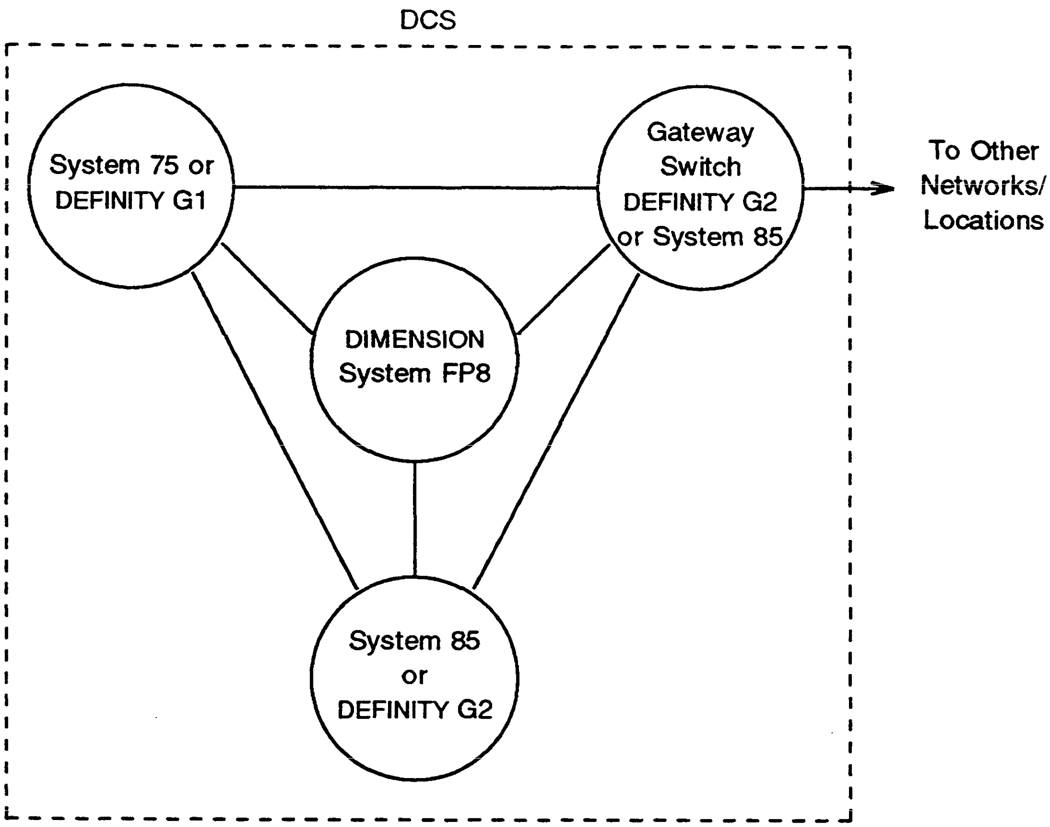

Figure 53-1. A DCS Cluster Connected to Another Network

General Concepts

Feature Transparency

Feature transparency is the additional characteristic that DCS provides for a Main / Satellite or ETN network. A feature is transparent when it works the same (from the user's point of view) whether the terminals involved are located on the same node or on different nodes. For example, in a non-DCS network, a terminal user can activate Priority Calling only to another terminal on the same switch. In a DCS, however, a terminal user can activate Priority Calling to any unrestricted terminal on any node in the DCS.

NOTE: Features may be transparent in some DCS arrangements and not in others. All switches that can connect to a DCS do not have the same capabilities and some features work differently on one type of switch than they do on another. For this reason, a feature that may be transparent when every switch is a System 85, Release 2, Version 4 or a DEFINITY Generic 2 can lose transparency when any other type of switch (such as a System 75) or another version of the System 85 switch is involved.

The DCS Cluster

A DCS Cluster consists of two or more switches interconnected in such a way that a limited set of features is transparent to every user. The terms DCS and Cluster can be used interchangeably or in combination to refer to a Distributed Communications System network.

Mixed Clusters

As shown in Figure 53-1, a DCS Cluster can include different types of switches. The same DCS can include different generics of the DEFINITY switch, different versions of the System 85 Release 2 switch, DIMENSION System switch (FP8), or the System 75 switch. Some interactions and special limitations apply in "mixed" clusters. For example, a System 75 node prior to RI V3 can only serve as an end point or terminating node in a DCS. This reference manual addresses the System 85, Release 2 switches and the DEFINITY Generic 2 switch. Differences will occur when other types of switches are used in a DCS. Some of these differences are noted under Considerations and Interactions with Other Features.

Data Paths

Each node in a DCS Cluster must have a data path to every other node. Data paths allow the switch processors to exchange call processing information, and it is this exchange of information that enables the network to function as one switch. The DCIU (Data Communications Interface Unit) uses *network channels* within DCIU *links* to provide these data paths.

The DCIU Link

The DCIU link, in its DCS role, is a hardware connection between DCIUs on separate switches. Each System 85 or DEFINITY Generic 2 node in a DCS must have a DCIU and must be connected through the DCIU, either directly or indirectly, with every other node in the DCS. The DCIU link is sometimes called a *Messaging Link.* It provides a full duplex, synchronous data path between DCIUs. The DCIU links carry call processing information between nodes over logical channels.

Logical Channels

A logical channel is a virtual circuit that is a segment of the data stream carried on a DCIU link. It is called a *virtual circuit* because it is defined by software (or firmware) rather than consisting of discrete hardware components. A logical channel carries a packet or frame of data. Packets and frames are essentially the same thing within the BX.25 environment. Technically, a frame contains some additional header and trailer information that is used by the BX.25 protocol but does not otherwise affect the information content of the packet. For the Switch Link (the connection between the DCIU and the local switch processor), logical channels are called ports. Each link (Release 2, System 85 or DEFINITY Generic 2 DCIUs) supports up to 64 software-assigned logical channels. If either end of a link connects to a DIMENSION System switch, the link is limited to 20 logical channels.

Data paths may be direct (from the originating switch to the receiving switch) or indirect (pass through intervening nodes when necessary). Indirect data paths can use up to three DCIU links (one between each pair of switches or processors in the path).

Limiting Factors

Each switch is limited in the number of DCIU links that it can support for DCS use. Factors that limit the number of DCIU links available include the following

- The type of switch (DEFINITY Switch, System 85, System 75, or DIMENSION Switch). Different types of switches support a different number of DCIU links.
- The number of APs or adjunct processors (such as AUDIX units) connected to the switch. Each AP and adjunct processor uses a DCIU link.
- The type of DCIU linkage (direct or indirect) used.

The DCIU

The DCIU is a special purpose processor. It is located in the Common Control Cabinet of the System 85 or DEFINITY Generic 2. The DCIU operates as a packet switch. It receives call processing information from the switch processor, assembles this information into discrete packets, and then routes these packets of data to the appropriate distant switch processor over a DCIU link logical channel pair.

A ***link logical channel pair*** is a specific logical channel (or port) on a specific link, and is identified in software during DCIU administration. The DCIU also receives packets from distant switch processors and routes this information either to the local switch processor or to another distant processor acceding to internal instructions contained in the DCIU firmware and software.

DCIU Releases

Two DCIU releases are available, and both can be used in the same cluster.

- ***Release 1 DCIU***

  The Release 1 DCIU provides up to 4 links with 20 logical channels each and is used by DIMENSION System FP8 switches.

- ***Release 2 DCIU***

  The Release 2 DCIU provides up to 8 links with up to 64 logical channels each and is used by System 85, Release 2 and DEFINITY Generic 2 switches. On the Release 2 DCIU, 19 of the switch link ports are predesignated for DCS application.

Both releases support DCS connections; however, Release 2 DCIUs provide an alternate routing capability while the Release 1 version does not. Alternate routing is described later in this chapter.

The release 2 DCIU has a total of nine communications links (ports). There are eight external data links (1 through 8) that connect to other DCIUs. There is also one internal link (link 0), that connects to the local switch processor. Link 0 is also referred to as the *switch link* to distinguish it from the outside DCIU links. The most significant difference between the switch link and the DCIU links is the protocol used. The switch link uses DMA (Direct Memory Access), while the external DCIU links use the BX.25 transmission protocol.

Logical channels on different links are connected to each other within the DCIU via network channels. These DCIU network channels can be thought of as electronic patch cords. The Release 1 DCIU has 64 network channels while the Release 2 DCIU has 256.

Network Channels

The DCIU passes data between links along internal paths (paths within the DCIU itself) called network channels. Network channels are set up in administration (Procedure 257, Word 1). Network channels establish the association between logical channels on two different links.

When a data packet is received, it is routed via the network channel to the appropriate logical channel (or port) on the outgoing (from the DCIU) link. For example, data intended for the local switch is routed to a port on the switch link (Link 0), and data intended for a distant switch is routed to the appropriate logical channel and DCIU link. For data packets intended for a distant switch, this can work differently depending on what release DCIU is involved.

- In a Release 1 DCIU, network channels are given fixed assignments. That is, a specific network channel always associates packets received on a particular link logical channel or port with a specific logical channel on the link leading to the distant switch. This arrangement is called a Fixed Network Channel or PVC (Permanent Virtual Circuit).
- In Release 2 DCIUs, the packet routing concept is modified to allow both fixed channels and Alternate Routing. With Alternate Routing, the DCIU selects a network channel based on a destination routing code contained within the header information. The DCIU can select from up to two alternate network channels to reach a distant switch if the link for the primary network channel is not available for some reason.

The operation and functions of the DCIU is described in detail in Appendix H of this manual.

DCIU Linkage Options

Two options are available for DCIU linkage configuration: direct and indirect linkages. Direct linkage uses a direct (no hop) data path between nodes. lndirect linkage uses two or three links (one or two hops) between nodes. The term "hop" refers to a data path that passes through an intervening DCIU (where the logical channels are connected via a DCIU network channel) before reaching its destination. This is similar to the tandeming process for tie trunks except that DCIU messages are not sent into the switch for processing. Hop is used to distinguish between data paths and communications paths.

Indirect Linkage

Figure 53-2 shows a DCS cluster that uses indirect linkages to achieve a minimum linkage configuration Link minimization provides the minimum essential data paths needed for DCS service. At optimum, this means that there will be one, and only one data path between any two nodes. Notice that there are no direct data links available between Nodes 1 and 3, 1 and 4, or between Nodes 3 and 4. All data paths between Nodes 1, 3, or 4 are indirect. They must pass or hop through the DCIU at Node 2.

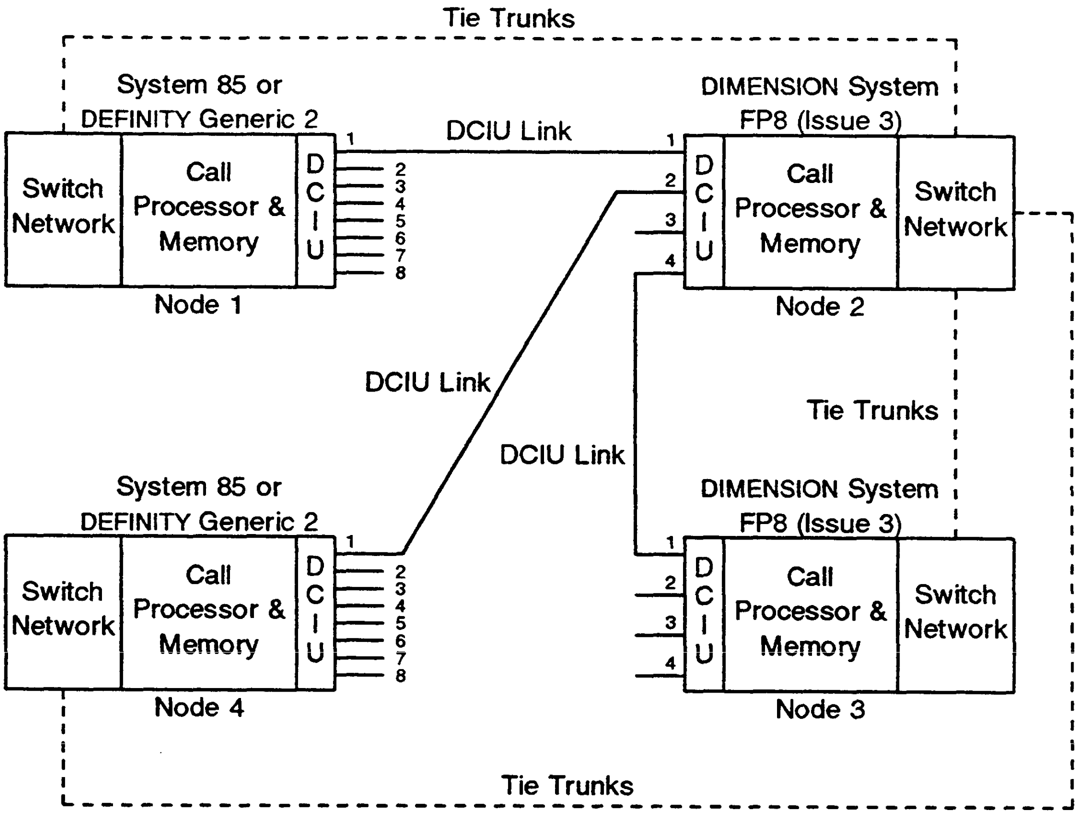

Figure 53-2. Link Minimization DCS Configuration

Because this arrangement requires less hardware, it is less expensive than direct linkage. It also uses fewer link connections at each DCIU (except the hub in a star configuration) leaving more connections available for other applications. However, indirect linkage is less reliable than direct linkage. For example, failure of the data link between Nodes 2 and 4 (Figure 53-2) will cause loss of transparency between Node 4 and all other nodes in the network.

Direct Linkage

Direct linkage provides "no hop" data paths between nodes in a DCS. A fully implemented direct linkage configuration would provide a no hop data path between every node in a DCS. Figure 53-3 shows a DCS that uses direct linkage for most of its data paths (4 out of 6).

In a large or highly disbursed network, fully implemented direct linkage is not always practical and usually some combination of direct and indirect linkage is used. In the example shown in Figure 53–3, if the data link between Node 1 and Node 2 fails, transparency is lost only between these two nodes. If ***alternate routing*** were available, transparency could be maintained even between Nodes 1 and 2.

The problems with direct linkage are that for a system with three or more nodes, direct linkage is more expensive than a link minimization arrangement; direct linkage can limit

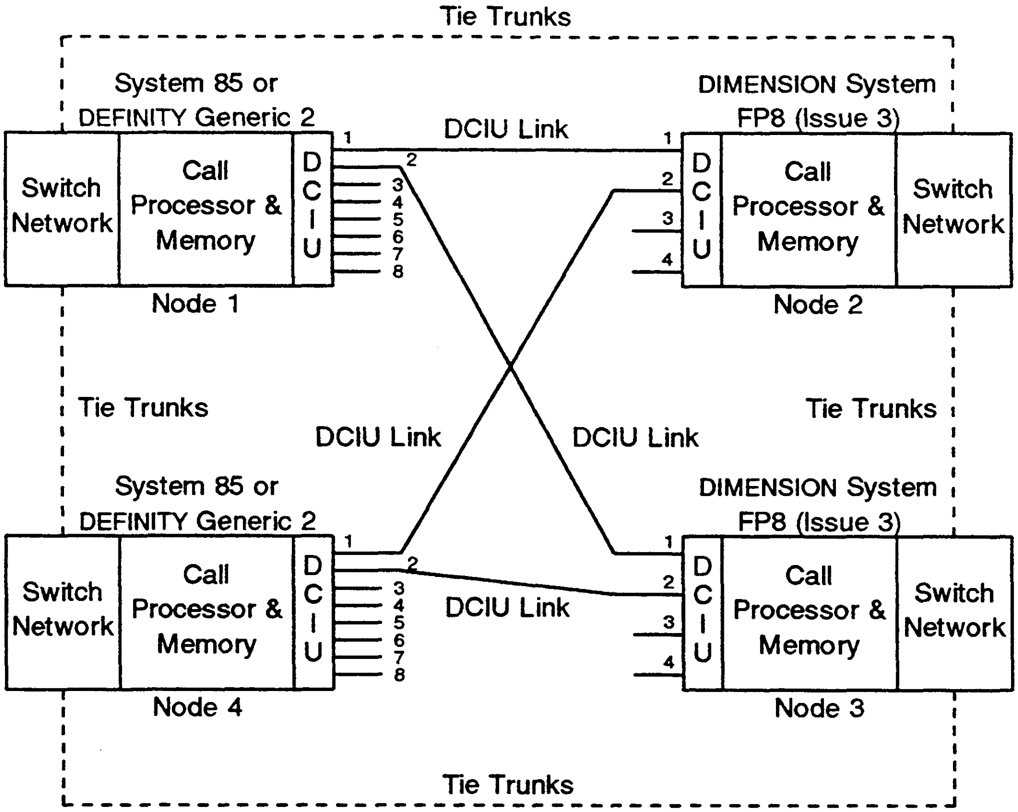

Figure 53-3. Direct Linkage DCS Configuration

the number of nodes that can be included in a DCS; and direct linkage uses DCIU connections that may be needed for other applications.

Alternate Routing of DCIU Messages

Alternate routing is assigned on a per-network channel basis and provides a way for DCIU messages (DCS applications) to bypass a failed DCIU link. Alternate Routing is ***available only on data paths where Release 2 DCIUs are used at both ends.*** The use of link minimization configurations tends to limit or eliminate the effectiveness of alternate routing.

Network Channels

Alternate routing is enabled for a network channel using switch administration Procedure 257, Word 1. When a network channel is assigned alternate routing, the second link and logical channel (Component B) is filled with dashes. The alternate routing network channel is then assigned a ***destination routing code*** in Procedure 257, Word 2. In effect, the destination routing code takes the place of the second logical channel link pair for alternate routing network channels. When alternate routing is used, the packet header includes a destination routing code and postage.

Destination Routing Code

The destination routing code identifies the node for which a packet is intended. It must be common throughout the network. That is, a given destination routing code must identify the same destination switch and port from any node in the system. The destination routing code is used at each alternate routing DCIU to select up to three routes (a primary and two alternate routes) that can be used from that DCIU to reach the destination switch. At the DCIU serving the destination switch, only one route (the primary) is provided that routes the packet to the designated port on the switch link. Alternate routing network channels are assigned to destination routing codes at each alternate routing node using Procedure 257, Word 4.

Postage

Postage is the system's way of keeping track of the number of hops that have been used. A data path is limited to a maximum of two hops. When the DCIU receives a packet that is using alternate routing it decrements the postage; and if the result is negative, the message is discarded (the 2-hop limit has been used up). If the result is positive, the DCIU selects a network channel from the routing table that serves the destination routing code of the packet.

Fixed Network Channels

Alternate routing paths can use fixed routing network channels or ***PVCs (Permanent Virtual Circuits)*** at hops or interveininf nodes. When an alternate routing path passes through a node that uses a PVC, it works just like a nonalternate routing data path. That is, it passes from the incoming link logical channel to a specific, predesignated outgoing link logical channel, without the local DCIU making any route selection decisions. However, ***alternate routing packets must originate and terminate on alternate routing nodes.*** The additional header information used for alternate routing (destination routing code and postage) cannot be modified by a fixed network channel switch.

Routing on Failure

Alternate routing uses a technique called routing on failure. This technique uses the primary data path (first-choice routing) until a link failure is encountered. If this happens, the switch can select an available alternate route serving the same destination routing code. Each destination routing code is assigned a primary route and one or two alternate routes at each node that uses alternate routing. This assignment is made during DCIU administration.

Alternate Routing Example

Figure 53-4 shows an example of a mixed DCS cluster that uses alternate routing. The DCIU messaging paths are labeled in *italics*. Each path is labeled with the node numbers that it serves. That is, *"**12**"* is the path between Node 1 and Node 2. Fixed paths (paths that do not use alternate routing) are shown as two digit numbers while alternate routing paths are labeled as three digit numbers, with the third digit representing the sequence number in the alternative routing scheme. That is, the primary path between Node 2 and Node 3 is labeled *"**231,**"* while the second choice (first alternate) path is labeled *"**232.**"*

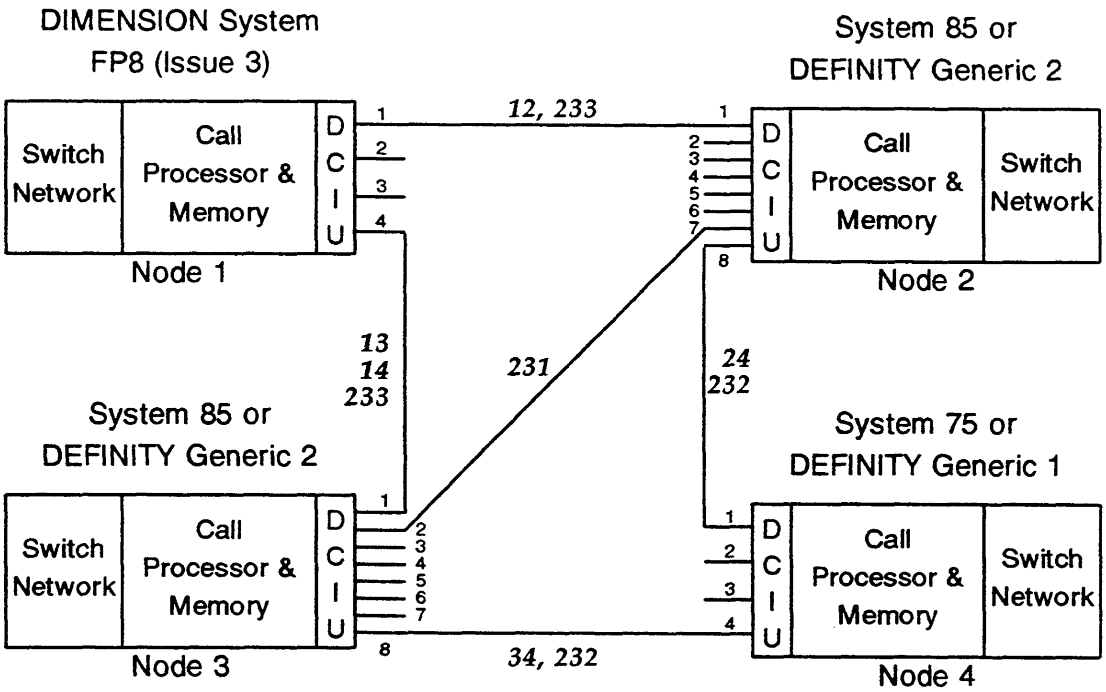

Figure 53-4. Alternate Routing in a DCS Arrangement

Fixed Routing Paths

Data paths originating or terminating on Nodes 1 and 4 are set up using fixed routing or PVCs (Permanent Virtual Circuits). Node 1 is a DIMENSION System FP8 switch and Node 4 is a System 75 (or DEFINITY Generic 1) switch. Neither DIMENSION switches, nor System 75, nor Generic 1 switches are capable of alternate routing. All data paths that either originate or terminate at either Node 1 or Node 4, must use fixed routing or PVCs. These fixed routing paths are as follows:

- Path ***12*** is a direct linkage path between Node 1 and Node 2.
- Path ***13*** is another direct linkage path between Node 1 and Node 3.
- Path ***14*** is an indirect linkage path that uses the same link as path 13 between node 1 and node 3. It then hops through the DCIU at Node 3 and goes on to Node 4 over the same link that is used by Paths ***34,*** and ***232.***

  Note that here, the word "link" refers to the physical link and not the virtual circuit on that physical link. Different paths using the same physical link will use different virtual circuits on that physical link.
- Path ***24*** is another direct linkage path between Node 2 and Node 4.

If any of the links carrying these fixed network channel paths fail, transparency for the connected nodes is lost on every fixed network path that uses that link. That is, if the link between Node 1 and Node 3 goes down, DCS transparency is lost between Node 1 and Node 3, and between Node 1 and Node 4.

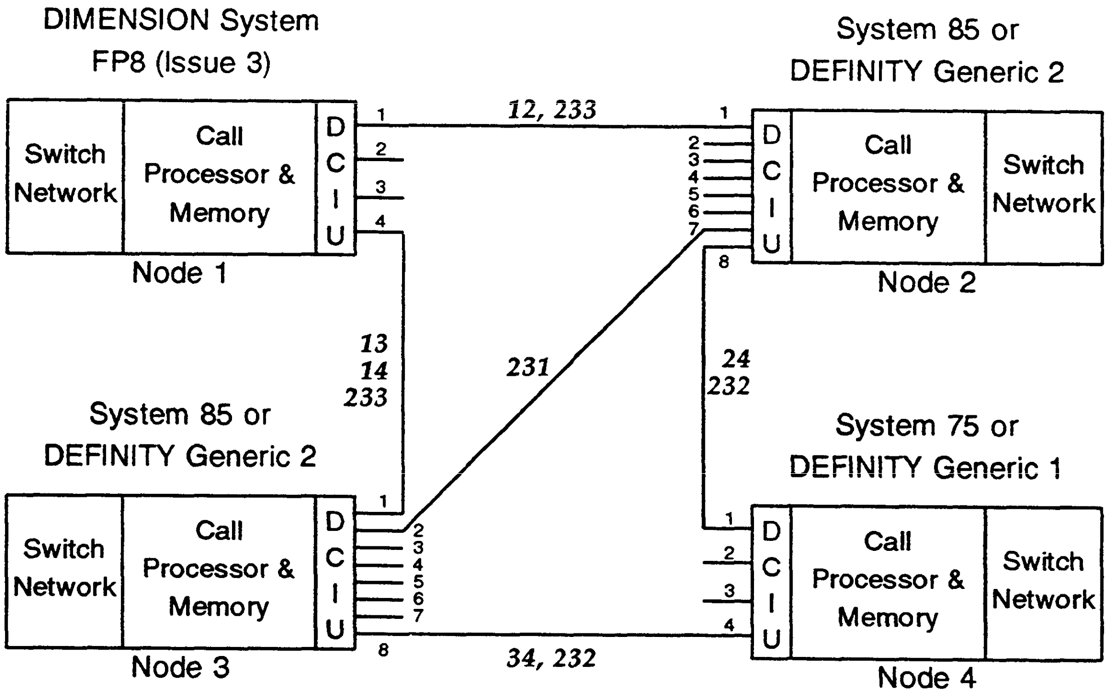

Figure 53-4. Alternate Routing in a DCS Arrangement

Figure 53-4 is repeated above for ease in referencing.

Alternate Routing Paths

Both System 85, Release 2 and DEFINITY Generic 2 switches use the Release 2 DCIU. The Release 2 DCIU is required for alternate routing. In the example in Figure 53-4, only Node 2 and Node 3 can use alternate routing because they are the only nodes that use the Release 2 DCIU. The alternate routing paths are set up as follows:

- Path ***231*** is the primary path and connects directly between Node 2 and Node 3.
- Path ***232*** is the first alternate route. This is an indirect linkage path that uses the link between Node 2 and Node 4 (the same link that is used by path ***24),*** hops through the DCIU at Node 4, and uses the link between Node 3 and Node 4 (the same link used by paths ***34).*** The hop through the DCIU at Node 4 uses a permanent virtual circuit and does not involve Node 4 or its DCIU in any routing decisions.
- Path ***233*** is the second alternate route. This also is an indirect linkage path that uses the link between Node 1 and Node 2 (the same link that is used by path 12), hops through the DCIU at Node 1, and uses the link between Node 1 and Node 3 (the same link used by paths ***13*** and ***14).*** Again, the hop through the DCIU at Node 1 uses a permanent virtual circuit and does not involve Node 1 or its DCIU in any routing decisions.

Note that the same set of paths ***(231, 232,*** and ***233)*** is administered in the same way at both Node 2 and Node 3. That is, path ***231*** is the same channel on the connecting link from both Node 2 and Node 3.

Fixed Network Channel Hops

Fixed network channel hops (or PVCs) can be use on data paths in an alternate routing pattern. In the above example, alternate routing paths with fixed network channel hops are used as the alternates rather than the primary path. That is, path ***231*** is tried first. Only if path ***231*** fails, is path ***232*** attempted. If both these paths fail, then path ***233*** is tied. If a direct linkage route (no hops) is available it should normally be used as the primary path, regardless of the type of hop used on alternate routes.

The only real problem with using a fixed network channel hop (or PVC) is that postage is not decremented on the packet. By passing through a nonalternate routing node (Node 1 or Node 4), it is possible to exceed the 2-hop limit. In the simple network used for this example, there is no real problem. The 2-hop limit could only be exceeded by going into an infinite loop. For an infinite loop to occur in a DCIU alternate routing arrangement, there must be at least three alternate routing nodes. However, in a large and complicated network, this could be a meaningful problem that can only be avoided by careful engineering.

Data Path Looping

The possibility of entering a logical infinite loop can exist in certain alternate routing arrangements. This is because ***alternate routing data paths are not fixed but are established dynamically*** at each alternate routing DCIU as the packet passes from one link to the next. Figure 53-5 shows a DCS arrangement where infinite looping could occur.

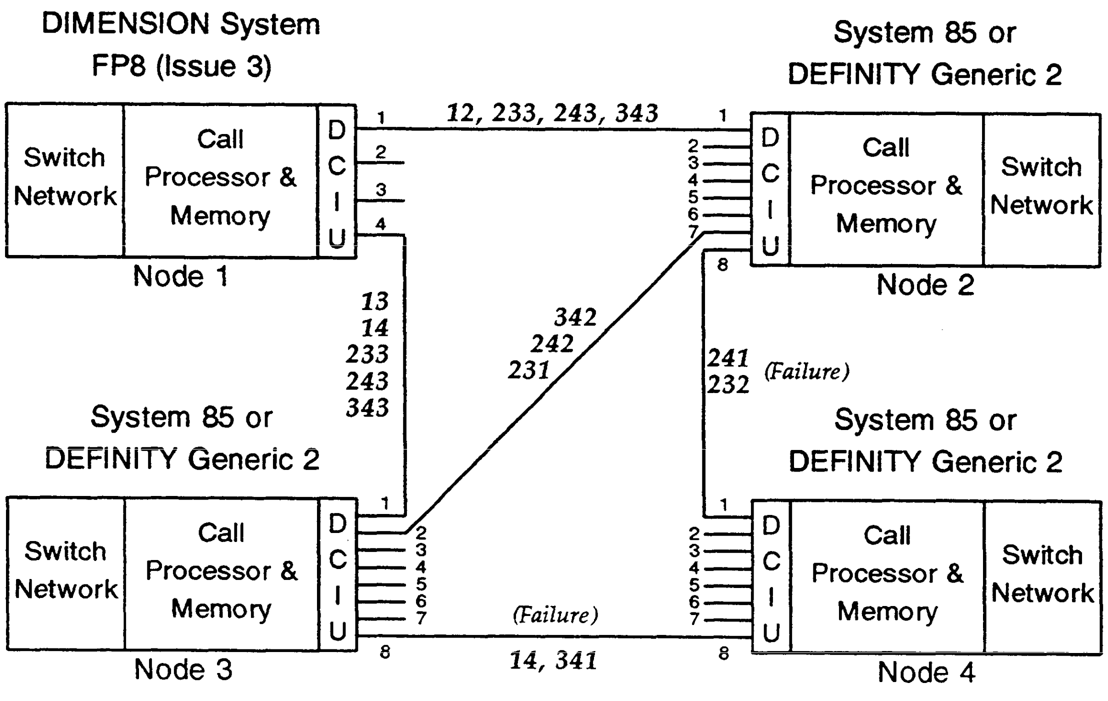

Figure 53-5. Alternate Routing with Looping

This configuration is remarkably similar to that used in the previous example. The only difference is that in this example, Node 4 is an alternate routing node. This results in three alternate routing nodes and three sets of alternate routing paths. The failure states shown in this example are are unlikely to occur simultaneously. This scenario is used only to illustrate data path looping in a simple setting.

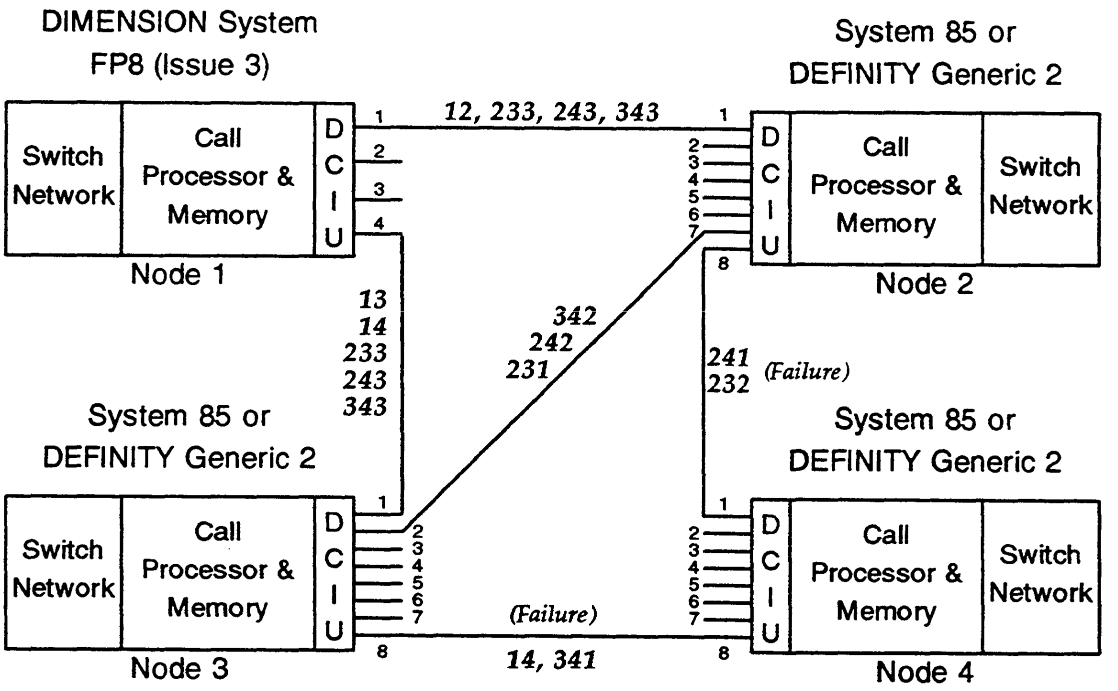

Figure 53-5. Alternate Routing with Looping

Figure 53-5 is repeated for convenience in referencing.

The paths shown in the example in Figure 53-5 are set up as follows:

- *Node 1:*

  The paths to and from Node 1 are set up in exactly the same way as they were in the previous example. That is, all paths to and from Node 1 are fixed routing paths. This is because Node 1 is a nonalternate routing node (Release 1 DCIU). Paths ***12*** and ***13*** are direct linkage paths between Node 1 and Nodes 2 and 3 respectively.

  Path ***14*** is an indirect linkage route that follows the link to Node 3, hops through the DCIU at Node 3 and then follows the link between Node 3 and Node 4.

  Notice that path ***14*** "hops" through the DCIU at Node 3. This is fixed network channel (or PVC) routing, even though this DCIU is alternate routing capable. This path cannot use alternate routing at any point, because it originates (or terminates) on a nonalternate routing DCIU. The presence of an alternate routing capable DCIU along the path makes no difference. As long as either the origin or destination of the packet is a nonalternate routing DCIU, the path cannot use alternate routing at any point.

- *Node 2:*

Path ***12:***

Path ***12*** is the same as described for Node 1 above, a direct linkage fixed network channel path between Node 1 and Node 2.

Path ***231:***

Path ***231*** is the first choice route between Node 2 and Node 3. This is a direct linkage route and as long as it is in service, will be used for DCIU messaging between these nodes.

Path ***232:***

Path ***232*** is the first alternate route (second choice route) for DCIU traffic between Node 2 and Node 3. This path will not be used unless path ***231*** fails for some reason. In terms of the ultimate destination (Node 3), this is an indirect linkage route that passes through the DCIU at Node 4 before continuing on to Node 3. However, it is not administered that way.

Because this is an alternate routing path that is processed through an alternate routing capable DCIU (at Node 4), it works differently than the fixed network channel case. The path from Node 2 to Node 4 is a direct linkage path between these two nodes. This path actually terminates a Node 4. The path designation ***232,*** does not appear on the link between Node 3 and Node 4. When the DCIU message from Node 2 reaches the DCIU at Node 4 the destination routing code for the message is examined. Routing from Node 4 is based on the destination routing code and path assignments at Node 4 (path ***341***). In this case, the first choice route is over the link between Node 3 and Node 4. This route is also a direct linkage path, connecting Node 3 and Node 4. Note that in this case, the "postage" is decremented.

Path ***233:***

Path ***233*** is the second alternate route (third choice) for DCIU traffic between Node 2 and Node 3. This is another indirect linkage route. It uses the link between Node 1 and Node 2, and then hops through the DCIU at Node 1. This is a fixed network channel hop that does not involve Node 1 or its DCIU in a routing decission. from Node 1, path ***233*** uses the direct link to Node 3 (the same link that is used by paths ***13*** and ***14).***

- Nodes 3 and 4

The alternate routing paths from Nodes 3 and 4 work exactly like those from Node 2. That is, the most direct route is selected as the first choice. Alternate routes all pass through at least one intervening DCIU. Any route that hops through Node 1, uses a fixed network channel and does not decrement postage at that hop. Any route that passes through an alternate routing DCIU uses the destination routing code and selects the next link based on the alternate routing translations at that DCIU. At alternate routing DCIUs postage is decremented. Path ***343*** passes through both a fixed network channel connection and an alternate routing DCIU.

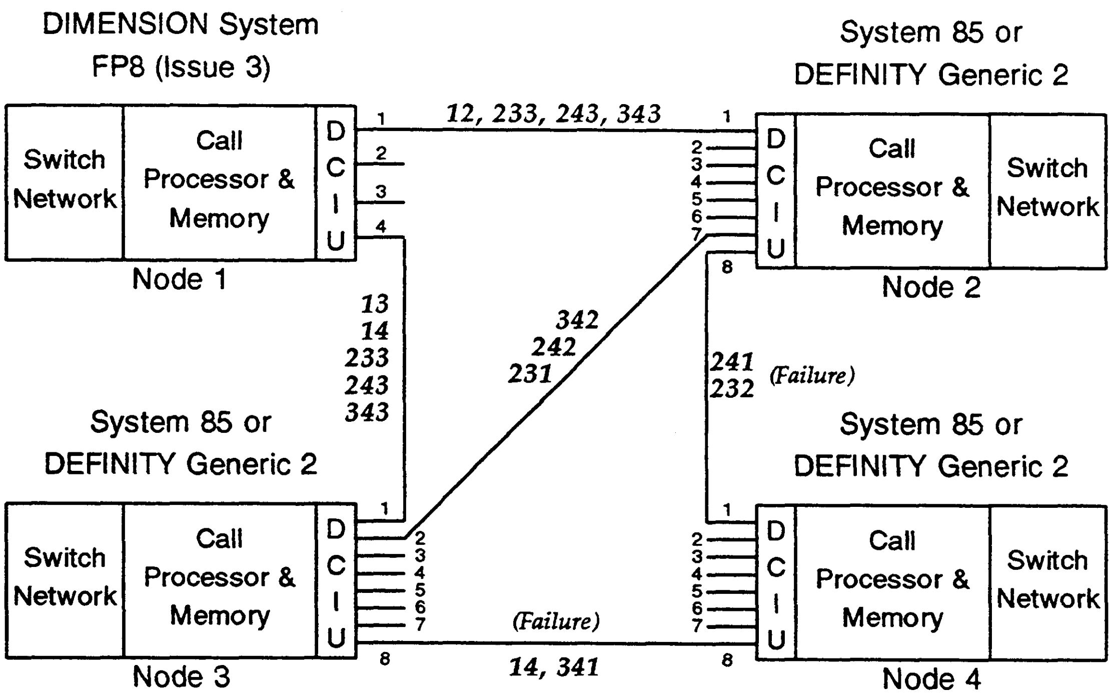

Figure 53-5. Alternate Routing with Looping

Figure 53-5 is repeated for convenience in referencing.

Path ***343:***

This path is a true indirect linkage path in that it hops through Node 1 using a fixed network channel connection. Alternate routing is not used at this point (note that path designator 343 appears on both the link between Node 1 and Node 3 and the link between Node 1 and Node 2). When a packet from Node 3 reaches Node 2, the route terminates and the destination routing code is evaluated at Node 2 using alternate routing. At this point, postage was not decremented at Node 1 but is decremented at Node 2. Routing evaluation at Node 2 selects path ***241*** for the next connection (assuming that this path is available). Following this path, a packet traveling between Node 3 and Node 4 passes through two intervening DCIUs but postage is decremented only once.

The Failure Case

With the arrangement shown in Figure 53-5, the data paths can enter a logical infinite loop if the physical links between Nodes 2 and 4, and between Nodes 3 and 4 fail. With these two links down, a packet from Node 2, destined for Node 4 attempts to route as follows:

- The initial attempt is to route over the first choice, path ***241.*** Finding this route in a failed state, the second choice (path ***242)*** is used. This takes the packet to Node 3.

- At Node 3, the destination routing code is evaluated and a route to Node 4 is selected. Again, the first choice route (path ***341***) is not available and the second choice (path ***342)*** is used. This path takes the packet back to Node 2.
- At Node 2, the first choice (path ***241)*** is still in a failed state and the second choice (path ***242)*** is again used, taking the packet back to Node 3.

The packet in the above example has entered a logical infinite loop. It can readily be seen that unless something is done to stop it, this packet will bounce back and forth between Node 2 and Node 3 like a ping pong ball.

Stopping the Infinite Loop

The best way to deal with the infinite looping data path problem is to avoid it by careful network engineering. However, the postage field in the packet header provides a mechanism to end an infinite data path loop automatically if this should occur.

Postage Method

As mentioned earlier, when an alternate routing capable DCIU receives an alternate routed packet, it decrements the postage; and if the resulting value is negative, discards the packet. In this way, after a packet has tandemed through two alternate routing DCIUs the value of the postage field is zero. Even if a packet enters an infinite looping data path, the packet will be discarded (and the data path ended) when the postage runs out. Of course, when this happens transparency for the associated call is lost.

Point of Origin Only Alternate Routing

Another simple method of avoiding the infinite loop is to use alternate routing only at the point of origin of a packet. This is the method used in the first example (Figure 53-4), although its use there was dictated by the physical constraints of the network rather than a matter of design intent. With point of origin alternate routing, all hops through intervening DCIUs use fixed network routing channels, regardless of whether or not alternate routing is available at the intervening DCIU. In this way, a data path will never double back on itself to form an infinite loop. However, with this method, postage is not decremented and it is the responsibility of the designer to insure that the two hop limit is enforced.

DCIU Ports

Fixed DCIU Port Reservations

Prior to Release 2, Version 4 System 85 switches, the uses for DCIU ports were fixed. That is, each DCIU port can be used for only one application. Each DCIU port is reserved for a specific use in the factory before the system is shipped. These reservations cannot be changed and these ports cannot be used for any other purpose. The same ports must be used for the same purpose on every DCIU. (Refer to Appendix H.)

For example, on a System 85, Release 2, Version 3 switch, DCIU port number 11 can only be used for Leave Word Calling on AP number 2. DCIU port number 11 has been permanently reserved for the Leave Word Calling feature on AP 2. If there is no AP number 2 on a given switch, or if AP 2 does not have a leave word calling function, this DCIU port cannot be used.

Flexible DCIU Port Reservations

Flexible DCIU port reservations was introduced in System 85, Release 2, Version 4. Flexible port reservations provides a degree of flexibility not available with earlier versions. It also results in administrative requirements not previously placed on users or service support personal. With flexible port reservations, if a particular requirement (for example AP 2) does not exist, a port is not reserved for that requirement. If there is no AP 2, ports 9 through 13 (those ports designated for AP 2) can be used for another purpose if needed. This flexibility is further enhanced by the shared use of adjuncts in a DCS which was also new with System 85, Release 2, Version 4 switches.

Port Reservation Versus Port Assignment

Assignment

Assignment is done at the customer facility. Assignment amounts to turning on the function for which a DCIU port has been reserved. Assignments are made during switch administration. The assignment of an application to a DCIU port is limited to the application for which that port has been reserved.

Reservation

Port reservation establishes in the switch processor memory the identity of the device that will be at the distant end of the DCIU link. In Release 2, Version 3 and earlier switches, reservations are made in the factory and are fixed. For these switches, every like DCIU port is reserved for the same application. In Release 2, Version 4 switches, port reservations are flexible and this flexibility extends to the assignment process.

Flexible port reservations allows a port that is not needed for a particular function to be used for an alternative purpose. With flexible DCIU port reservations, the assignment of DCIU ports is also flexible. However, once the port is "reserved" for a particular application, the port must either be assigned to that application, left unassigned, or the reservation must be changed before a different assignment can be made. Port assignment flexibility is based on the DCIU port reservation process rather than the DCIU port assignment process.

Reserved or Unreserved Port Status

Because DCIU ports on Release 2, Version 4 switches do not have standard reservations, some new terms and concepts are needed. The "reserved" or "unreserved" status of a port affects administration. DCIU ports can still be reserved at the factory before the switch is shipped. This is done based on the TRACS (Translations Recovery Additions Conversions System) input. However, ports do not default to standard reservations. If not pm-reserved, the DCIU ports are ureserved. Separate administration is needed to set up DCIU port reservations. Whether or not ports are pre-reserved, separate administration is still needed to assign (or turn on) each DCIU port to its reserved application.

Standard Reservations and Assignments for APs

For local APs, DCIU port reservations and assignments are still somewhat fixed. That is, if APs are to be used on a particular switch, standard DCIU port reservations and assignments mus be used. However, these standards are not assigned by default. The standard reservations and assignments must be administered just like any other port reservation and assignment. If a local AP is not required, these ports can be used for other purposes, but other ports cannot be used by a local AP. The standard ports for local APs are:

AP 1 Ports 1 through 8

AP 2 Ports 9 through 13

AP 3 Ports 14 through 18

AP 4 Port 19, and ports 30 through 33

AP 5 Ports 37 through 41.

The administration of the DCIU and its various applications is a complex topic and is treated separately and in considerable detail in Appendix H of this manual.

ES (Enhanced Services) Message Set

Beginning with System 85, Release 2, Version 4, the ES (Enhanced Services) message set is used for remote messaging in the following cases:

- Messages between AUDIX adjuncts and remote switches
- Messages between switches (switch-to-switch links).

N-digit Message Set: On System 85, Release 2, Version 3 and earlier switches, MCS (Message Center Service) messages are sent using the n-digit message set. This format is used for communication between the local switch and the AP.

On Release 2, Version 4 and later switches, messages sent ***from*** an AP to a remote switch are converted to ES format by the local switch and routed to the destination switch. Messages originating from an AUDIX adjunct are sent in ES format only.

Messages ***from*** a remote switch are sent in the ES format between switches. If the destination adjunct is an AP, the messages are converted to the n-digit format by the local switch before being sent on to the AP. If the destination is an AUDIX Adjunct, conversion is not necessary. The ES messaging capability provides several enhanced feature transparency operations. For example it is ES messaging that makes Centralized Messaging possible.

Centralized Messaging

Centralized messaging allows users in a DCS to access AUDIX and MCS (Message Center Service) without requiring each node in the DCS to have an AUDIX adjunct or Message Center AP. The operations of centralized messaging are generally transparent to the user.

Centralized Messaging Examples

The following examples show how Centralized Messaging can be used in a DCS network under a variety of conditions. These examples assume that every node in the DCS uses a Release 2, Version 4 or later switch. Earlier versions ***do not*** support centralized messaging.

- Mesaging between a local switch and adjunct
- Messaging from a remote switch to an adjunct
- Messaging from a remote switch to an adjunct requiring one or more DCIU hops
- Messaging from an adjunct to a remote switch
- Messaging from an adjunct to a remote switch requiring one or more DCIU hops

Local Adjuncts

Messaging between the Switch and local adjunct is handled the same as for Release 2, Version 3 and earlier switches. Messages between the switch and an AUDIX adjunct are in the ES format. Messages between the switch and a local AP are in the n-digit format.

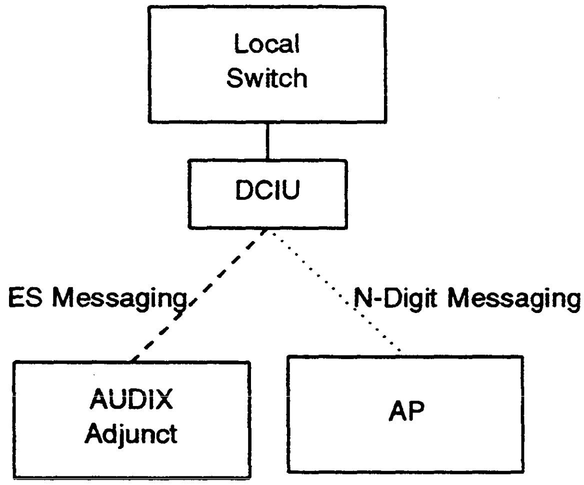

Figure 53-6. Local Adjunct Configuration

In the arraingement shown in Figure 53-6, only one (local) DCIU is used in the data path.

Remote Switch to Adjunct

Figure 53–7 shows a two node DCS arrangement with centralized messaging. When the remote switch sends a message to a centralized adjunct, the following occurs:

- The message is sent in ES format between switches.
- The local (centralized messaging) node receives the message and routes it to local adjuncts as follows:
  - — Messages to an AUDIX adjunct are sent in the ES format. This is the same format in which they are received from the remote switch.
  - — Messages to an AP are converted to the n-digit format and then sent to the local AP.

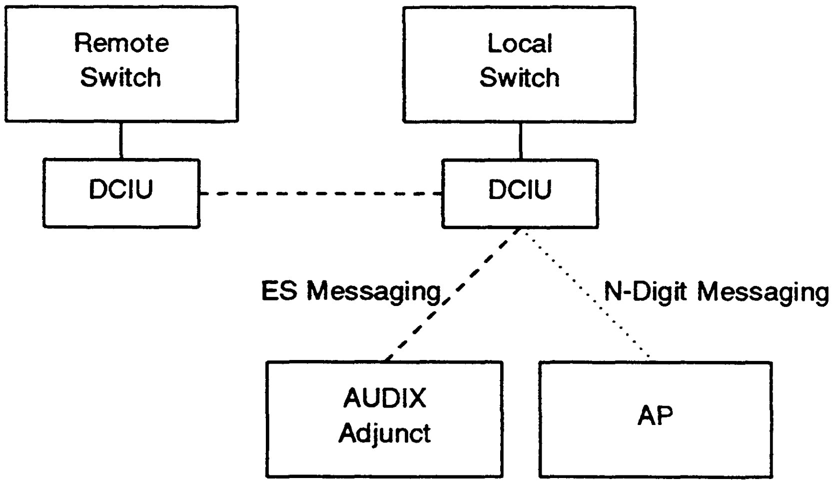

Figure 53-7. Remote Adjunct Configuration

In this example, two DCIUs are involved in the data link One at the remote switch where the message is originated, and one at the centralized messaging location. At the centralized messaging location, the same DCIU that supports the DCS link (between switches) supports the links to the local adjuncts.

Remote Switch to Adjunct Using Intervening DCIU

Figure 53-8 shows a three node DCS with centralized messaging. When the remote switch sends a message through an intervening switch the following occurs:

- The message is sent in ES format from the originating switch to the intervening node.
- The intervening switch receives the message and examines the addressing information contained in the message packet header. The addressing information shows that the message should be routed to the centralized messaging location.
- The message is routed to the centralized messaging switch through the appropriate ES port.
- The centralized messaging node receives the message and routes it to the appropriate local adjunct as in the previous example.
  - — Messages for an AUDIX adjunct are sent on in the same format in which they are received. No conversion is needed.
  - — Messages for an AP are converted to the n-digit format and then sent to the AP.

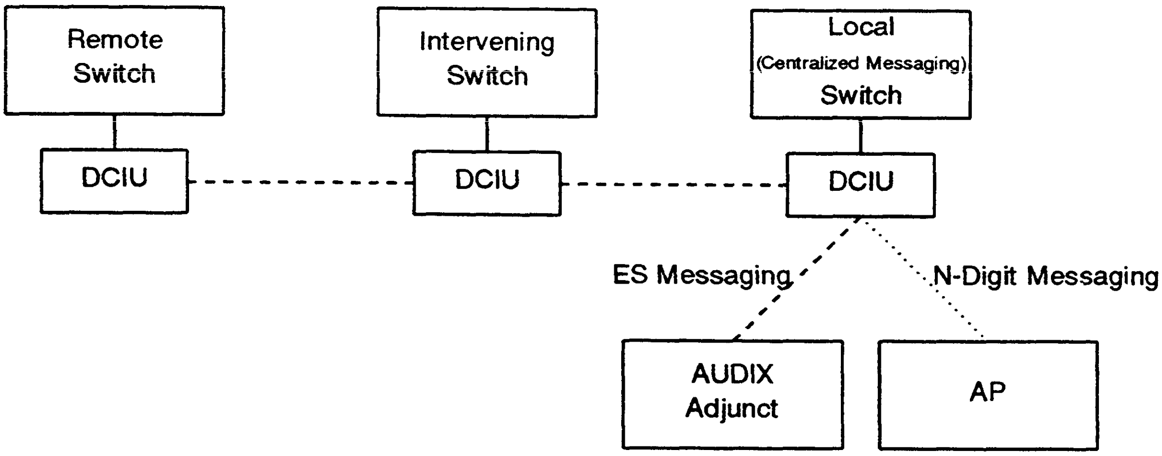

Figure 53-8. Remote Adjunct Configuration With a DCIU Hop

In this example, three DCIUs are involved. One at the originating switch, one at the intervening switch, and one at the centralized messaging switch. At the centralized messaging switch, the same DCIU supports the DCS links to other DCS nodes and the links to the local adjunct processors.

DCIU Hop: The process of passing a DCIU message through an intervening DCIU is called a ***hop.*** This is to distinguish it from the ***tandeming process*** used to pass trunk traffic through an intervening switch. The DCIU link is separate from and can follow a different route than the trunk path used by associated trunk traffic. That is, a DCIU message will not necessarily follow a path that parallels the trunk path for associated calls. Many DCIU messages, particularly in a centralized messaging arrangement, will not have associated trunk traffic.

Adjunct to Remote Switch

In Figure 53-9, an AP or AUDIX connected to Node 1 routes a message to Node 2. The following occurs:

1. If the message originates from an AUDIX adjunct, Node 1 reads the destination machine number and determines the message should be routed to another switch.

   or

   If the message originates from an AP, the message routes by extension number and is converted from n-digit format to ES format.
2. Node 1 routes the message to Node 2 over the appropriate ES port.
3. Node 2 receives the message.

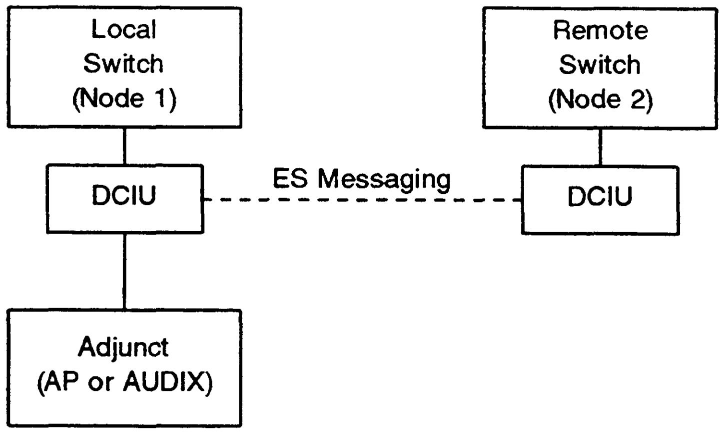

Figure 53-9. Adjunct to Remote Switch Configuration

Adjunct to Remote Switch Using Tandem Switches

In the next example (Figure 53–10), an adjunct (an AP or AUDIX) is connected to a remote switch via an intervening (tandem) switch.

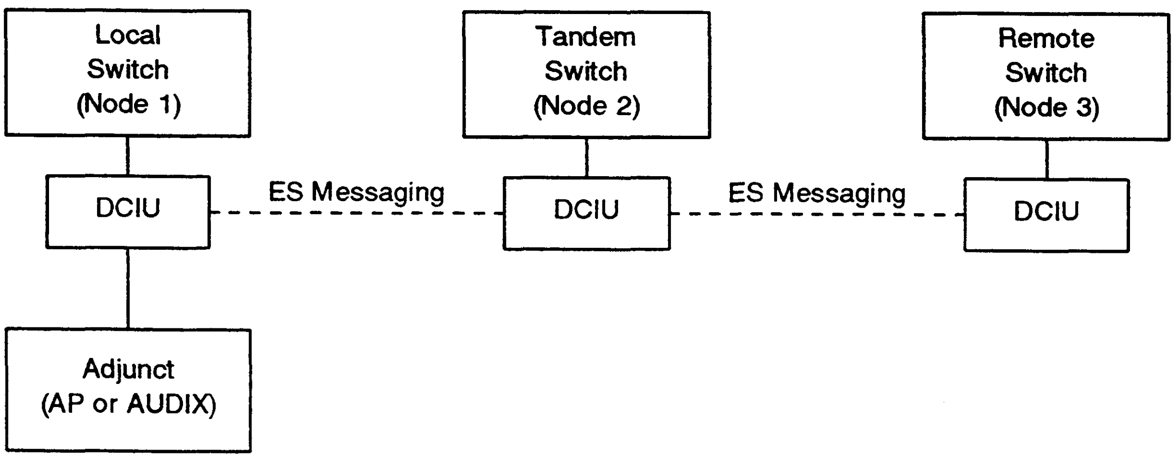

Figure 53-10. Adjunct to Remote Switch Configuration With DCIU Hop

In this case, when the adjunct (connected to Node 1) routes messages ***through*** Node 2 to Node 3 (an intervening switch in the message path), the following occurs:

1. If the message originates from an AUDIX adjunct, Node 1 reads the destination machine number and determines the message should be routed to another switch.

   or

   If the message originates from an AP, the message routes by extension number and is converted ***from*** n-digit format ***to*** ES format.

2. Node 1 routes the message to Node 2 over the appropriate ES port.
3. Node 2 receives the message and determines the message should be routed to Node 3.
4. Node 3 receives the message.

Adjunct Configurations

In a DCS environment, configurations of different switch releases can cause loss of transparency since different versions of the same switch type and different switch types may not possess the same capabilities. The following examples show network arrangements of System 85 and DEFINITY Generic 2 switches with different levels of transparency.

AUDIX

The first example (Figure 53-11) doesn't use Centralized Messaging. Any software version that supports AUDIX can be used if each switch is connected to an AUDIX Adjunct by a direct (local) DCIU network channel. AUDIX capabilities can be provided locally at each switch and data transfer across DCIU links is not involved. System 85, Release 2, Version 4 and DEFINITY Generic 2 capabilities are not transparent with this arrangement.

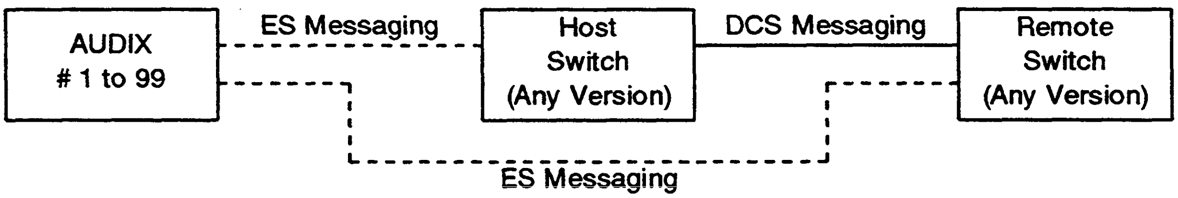

Figure 53-11. Remote AUDIX Without Centralized Messaging

In the second example (Figure 53-12), any switch that supports AUDIX can be used as a remote switch without its own network channel if the destination AUDIX network adjunct number is set to "1". This restriction occurs because older switches default the adjunct number in the ES message to "1".

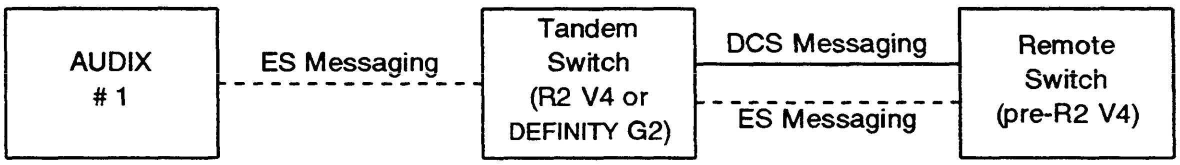

Figure 53-12. Remote AUDIX for Pre-Release 2, Version 4, Switches

In the third example, if the network AUDIX number is not "1", only Release 2, Version 4 and later switches can be used for the remote switch without a direct network channel to AUDIX (see Figure 53–13).

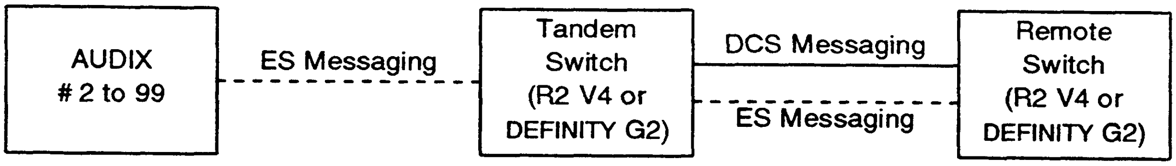

Figure 53-13. Remote AUDIX Configuration with Release 2, Version 4, Switches

MCS (Message Center Service)

The configuration shown in Figure 53–14 does not use Centralized Messaging. In this example, MCS can only support DCIU connections to one switch. Only the host switch can pass messaging information to MCS. Any System 85 or DEFINITY Generic 2 switch that supports MCS can use this configuration with the following variations in transparency::

- Switch versions before System 85, Release 2, Version 2 allow only direct calls to MCS that look like external calls.
- Release 2, version 2 handles direct and forwaded calls transparently.
- Release 2, Version 3 sends calls to MCS based on the call coverage specified.

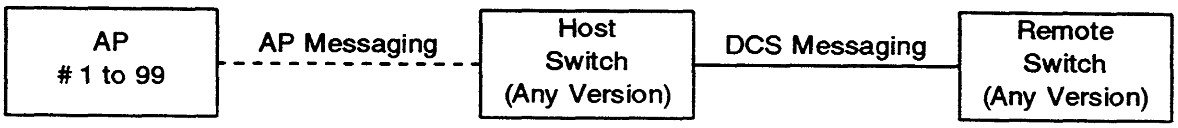

Figure 53-14. Message Center Service Without Centralized Messaging

With the configuration used in the next example, shown in Figure 53-15, MCS is fully transparent.

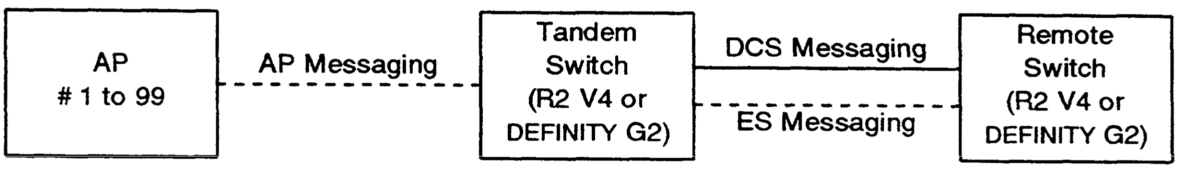

Figure 53-15. Message Center Service With Centralized Messaging

Messages can be sent from the remote switch to the MCS adjunct or to the remote switch from the MCS adjunct by tandeming through the host switch. Additional capabilities that are also available include:

- Remote automatic message waiting lamps are updated.
- Remote LWC (Leave Work Calling) is possible.
- Message retrieval is supported.

Uniform Numbering Plan

As with any automatically switched or direct dial networking system, a DCS requires a uniform coordinated numbering plan. In a DCS, network wide coordination of the numbering plan is critical.

In a conventional (non-DCS) ETN (Electronic Tandem Network) without Extension Number Portability, each node is assigned a unique switch number or location code. Each of these separate nodes then has its own separate set of extension numbers While the separate nodes must conform to the same numbering rules, duplication in extension numbers between nodes is allowed. This is because the switch numbers distinguish between like extension numbers on different nodes. A caller on one node of a conventional ETN who wants to call an extension on a different node dials the required access code, the location code for the remote node, and the extension number.

In a DCS, placing the call is simpler; however, the uniform numbering plan can be considerably more complex A caller in a DCS places a call to an extension on a remote node in exactly the same way that a call is placed to a local extension. The caller simply dials the extension number, and the switch does the rest. The caller does not need to know where the extension is located or the location code for that node. With DCS, the switch must be able to use the extension number to determine where the desired extension is located. Within the DCS, the same extension number cannot be duplicated on two separate switches. TMs must also work for calls coming into the DCS from an outside switch, either from a private network other than the DCS itself or from the public network.

Dialing Plans

A DCS dialing plan must use either a 4-digit or a 5-digit numbering system. To callers, every switch (dialing plan) within the DCS must appear the same. That is, if callers dial a 4-digit extension number to reach a local extension, they must be able to reach any extension on the DCS by dialing a similar 4-digit extension number.

NOTE: While a DCS must use a 4- or 5-digit numbering system, DCS is not a requirement for these numbering plans to work.

4-Digit Dialing

The principal disadvantage to a 4-digit plan is that it severely limits the number of extensions available in the network. A 4-digit numbering scheme cannot contain more than 10,000 numbers (ranging from 0000 to 9999). In practice, the numbers that can be used for extensions would be limited to much less than 10,000 (generally 4,000 to 5,000).

Extension Number Steering

A DCS that uses a 4-digit dialing plan can also use Extension Number Steering. Extension Number Steering is a switch attribute (administered using Procedures 350 and 354) that allows a number, in the same form as a 4-digit extension number, to invoke a feature or dial access code. Many users find this method of accessing features and trunks easier to learn and use than dialing the basic codes themselves. In a 4-digit dialing plan, Extension Number Steering can use the first one to four digits of the dialed extension number to identify the destination node. If the destination node is identified as the local switch, the call process normally to the local extension. If the destination node is a remote switch, Extension Number Steering produces the dial access code and trunk-group number needed to route the call to the remote node. In a 4-digit dialing arrangement, this is done as in a Main / Satellite arrangement. In fact, the DCS is configured as a Main / Satellite arrangement when a 4-digit dialing plan is used.

While a Main / Satellite arrangement is a satisfactory configuration for many small to medium-sized organizations, there are some inherent advantages to an ETN. These include the ability to use the AAR (Automatic Alternate Routing) feature or the WCR (World Class Routing) feature,* which are not available in a Main / Satellite arrangement. These features provide call routing capabilities (to a distant node) that are more flexible and more efficient than those available through Extension Number Steering. (However, in a Main / Satellite arrangement, some degree of alternate routing can be realized using the Route Advance feature.)

Unrestricted 4-Digit Dialing

Unrestricted 4-digit dialing was introduced for System 85, Release 2, Version 4, Issue 1.3 and is available only on these and later switches. This system records every extension number in the numbering plan in memory routing tables on each switch. These memory routing tables contain pointers that show where each extension number is currently located. This allows an extension number on any switch in the system to begin with any digit not otherwise restricted. This method makes implementing the Extension Number Portability feature easier.

Unrestricted 4-digit dialing is useful for smaller DCS configurations where the full capacity of unrestricted 5-digit dialing is not needed. (See the description of "Unrestricted 5-Digit Dialing.")

5-Digit Dialing

The 5-digit dialing options are designed for customers with a need for more than 10,000 numbers in a single system or dialing plan. Two forms of 5-digit dialing are possible:

- Prefix digit dialing
- Unrestricted 5-digit dialing.

* The AAR feature is available on System 85 and DEFINITY Generic 2.1 switches. The WCR feature is available on DEFINITY Generic 2.2 switches.

Prefix dialing employs Extension Number Steering or the AAR feature. Prefix dialing is used with DIMENSION System switches (FP8, Issue 3), System 75 switches, and System 85 switches prior to Release 2, Version 3. The unrestricted form of 5-Digit Dialing is available on System 85, Release 2, Version 3 and later, and Generic 2 switches.

Prefix Digit 5-Digit Dialing

The prefix digit form of 5-digit dialing uses the first one or two digits of the 5-digit extension number to route calls to the appropriate node. This is done either with Extension Number Steering or the AAR or WCR feature. Figure 53–16 shows a DCS that uses this type of 5-digit dialing.

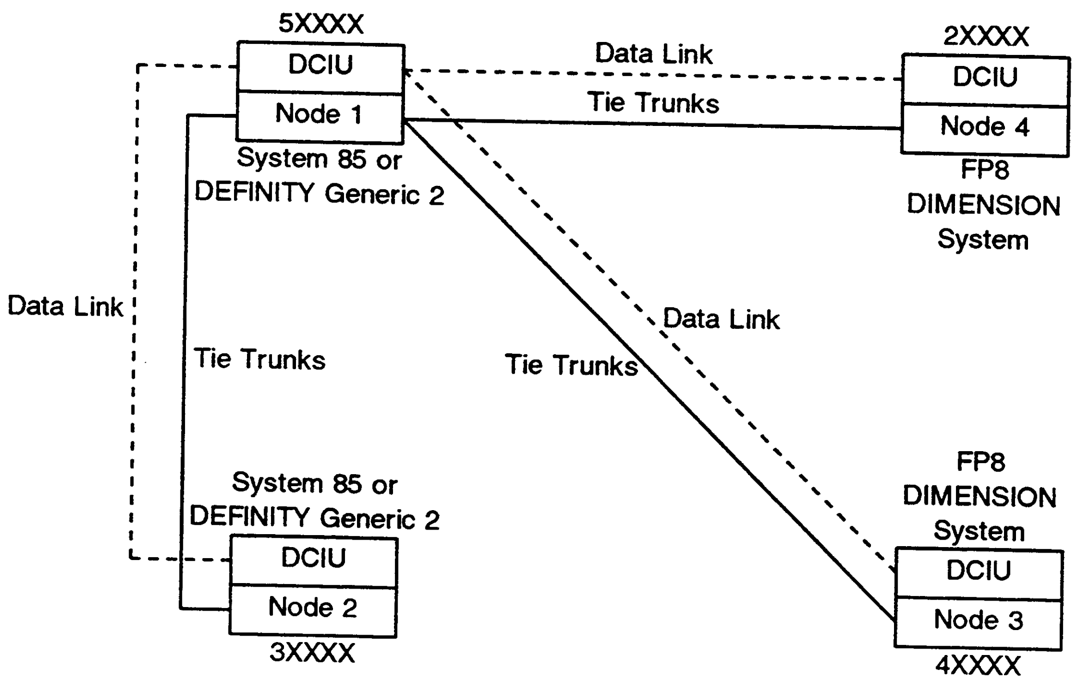

Figure 53-16. Prefix Type 5-Digit Dialing DCS Configuration

With prefix dialing the switch actually uses a 4-digit numbering plan and either discards the first dialed digit or uses it only for interswitch routing identification. If Extension Number Steering is used, it works in much the same way as with 4-digit dialing except that an additional digit is available for node identification. If the call is on the home or local switch, the prefix (first digit) is deleted, and the call is routed to the local 4-digit extension. If the call is for a remote switch, the prefix and second digits are used to identify the node where the desired extension is located. Either Extension Number Steering converts the prefix to the appropriate trunk-group dial access code or the prefix plus the extension number is converted to a 7-digit number, (including a location code) and the call is routed via the AAR or WCR feature. Either method (Extension Number Steering or location code routing) can be used, and both methods can be used within the same DCS.

In Figure 53-16, the "Xs" in the extension numbers stand for any digit 0 through 9. The first wo digits in each extension number must identify a specific switch or node.

NOTE 1: This does not mean that both the first and second digit must be unique to a single switch. For example, in Figure 53-16, all four nodes could use the same second digit as long as no two switches use the same first digit. Conversely, the first digit could be duplicated in different nodes as long as the second digit is different.

NOTE 2: For DIMENSION switches and System 85 switches prior to R2 V3, neither the first nor the second digit ("X") can be "0."

In the example in Figure 53-16, the digit "5" in 5XXXX identifies Node 1, the digit "2" in 2XXXX identifies Node 4, and so on. When a caller on Node 1 dials 22133, Extension Number Steering identifies the 22 as a call for Node 4. The switch then deletes and inserts the necessary digits for the dial access code and trunk-group number (or 7-digit number) to route the call to extension 2133 on node 4. This system is available for DIMENSION System (FP8), System 75, DEFINITY Generic 1, System 85, and DEFINITY Generic 2 nodes.

Unrestricted 5-Digit Dialing

Unrestricted 5-digit dialing was introduced on System 85, Release 2, Version 3 and is available only on these and later switches. This system records every extension number in the numbering plan in memory routing tables on each switch. These memory routing tables contain pointers that show where each extension number is currently located. This allows an extension number on any switch in the system to begin with any digit not otherwise restricted. This method makes implementing the Extension Number Portability feature easier.

One benefit derived from unrestricted 5-digit dialing is that the full line capacity of Release 2, Version 3 and later switches can be used. In a 4-digit dialing plan and in a 5-digit dialing plan where prefix digits are used, the numbering plan itself limits any switch to a maximum of 10,000 numbers. In an unrestricted 5-digit dialing plan, this arithmetic limitation is removed, and up to 32,000 extensions (depending on other applications) can be assigned to a single switch. However, a DCS Cluster is not limited to all prefix or all unrestricted 5-digit dialing. Figure 53-17 shows a DCS Cluster using both unrestricted and prefix type 5-digit dialing. For DCS Clusters using u.restricted 5-digit dialing on every node, a line capacity of 100,000 is possible. For users with a high local area line requirement, this could mean a significant savings by using fewer switches to meet the requirement.

As a simplified example, a company has a need for 43,000 new lines in a metropolitan area. Using switches with a maximum line capacity of 10,000 or less (or a 4-digit dialing plan), it will require 5 switches to meet this requirement; if a basic 4-digit dialing plan is used, a DCS arrangement would not be possible. Using a 5-digit dialing plan, this requirement can be met by 2 or 3 switches (depending on other applications).

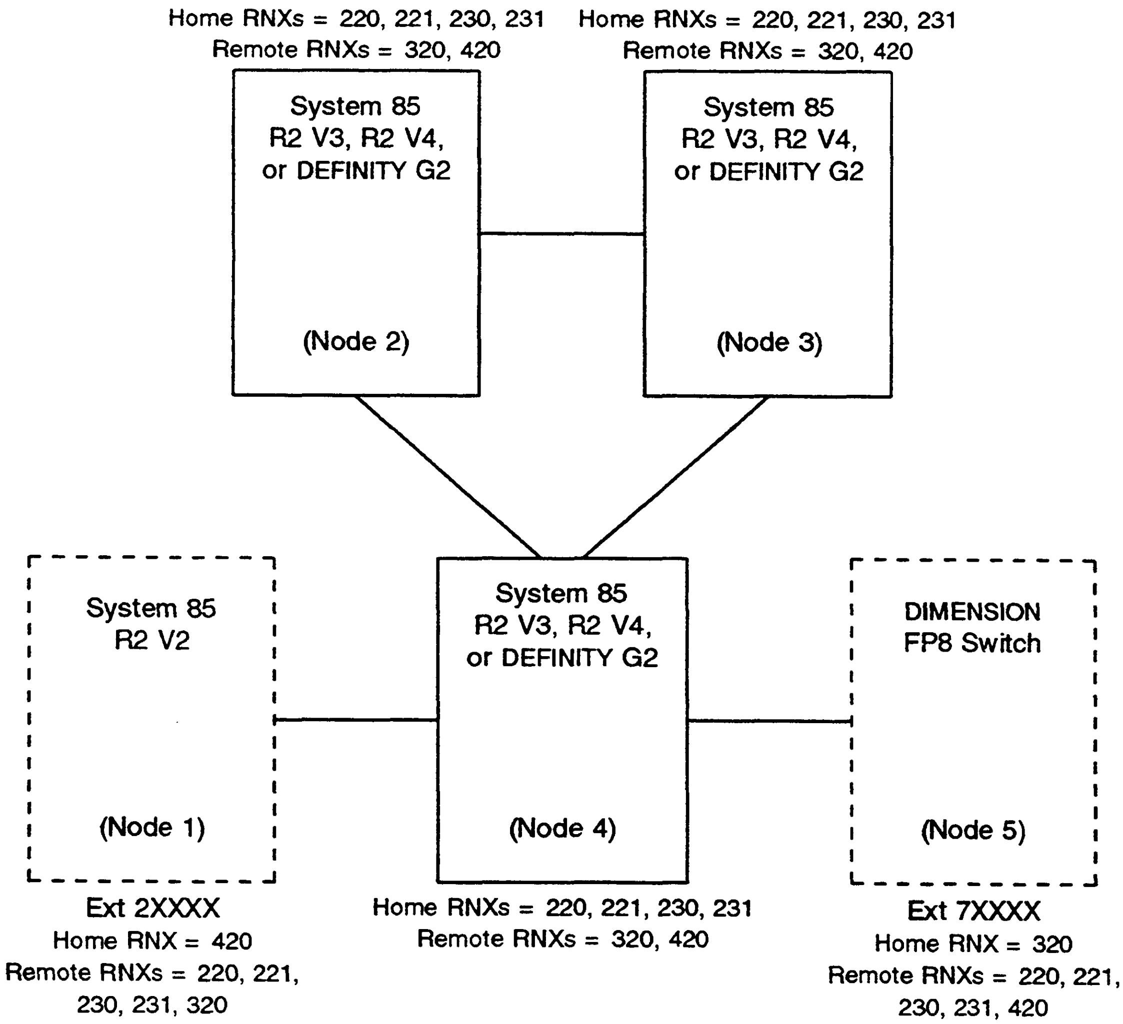

Figure 53-17. Unrestricted 5-Digit Dialing in a DCS Configuration

Figure 53–17 shows that switches using unrestricted 5-digit dialing and those using prefix digit 5-digit dialing can be used in the same DCS cluster. The unrestricted 5-digit dialing nodes form a subnetwork within the DCS. Within this subnetwork, the Extension Number Portability feature operates normally. However, portibility between this subnetwork and the other nodes is limited. See the Extension Number Portability feature for a more detailed discussion of this arrangement.

Other Applications of 5-Digit Dialing

Although 5-digit dialing was initially developed to meet the needs of the DCS customer with more than 10,000 extensions, 5-digit dialing can be usedoutside a DCS. For example:

- Extension Number Portability

The Extension Number Portability feature requires that an unrestricted 5-digit dialing plan be used for full, 2-way portability.

- ETN (Electronic Tandem Network)

  An ETN without DCS or an ETN location outside the DCS cluster can also use 5-digit dialing. The 5-digit dialing capability does not require a data link between locations.

Constraints

Some restrictions must be observed. Both 4-digit and 5-digit dialing plans cannot be used in the same cluster. For a cluster with a 5-digit dialing plan, private-network calls can be routed using location code routing (the AAR or WCR feature) or using the Main / Satellite feature. With a 4-digit dialing plan, System 85 switches in a cluster must be configured and administered like a Main / Satellite complex, and private-network calls are routed using Extension Number Steering. A more detailed discussion of the DCS feature, ETN networking and Main / Satellite arrangements can be found in Reference Manual, ***AT&T IS Network and Data Services*** (555-025-200).

User Operations

The DCS feature is software operated and controlled. There are no user operations as such associated with this feature.

Considerations

4-Digit Dial Access Codes

Beginning with R2 V4, System 85 and DEFINITY Generic 2 can provide 4-digit dial access codes. However, in a DCS, the following features do not function with 4-digit dial access codes if these features are used between nodes.

- Attendant Control of Trunk Group Access
- Attendant Direct Trunk Group Selection
- ACA (Automatic Circuit Assurance).

Adjunct Processors

Applications Processors and AUDIX (Audio Information Exchange) systems require DCIU links to the local switch. In Release 2, Version 3 and earlier switches, the ports for these links are permanently reserved. Beginning with Version 4, flexible port reservations allows DCIU ports to be used for other purposes if not needed for local adjuncts. Therefore, the use of local adjuncts on System 85, Release 2, Version 4, or DEFINITY Generic 2 switches reduces the possible number of direct linkage connections to other DCS nodes.

Centralized Messaging

Centralized messaging allows users in a DCS to access AUDIX and MCS (Message Center Service) without requiring each node in the DCS to have an AUDIX adjunct or Message Center AP. The operations of centralized messaging are generally transparent to the user.

Centralized messaging is available only when every node in the DCS uses System 85, Release 2, Version 4 or DEFINITY Generic 2 switches. Earlier versions do not support centralized messaging.

Dial Access Codes

Trunk group dial access codes for remote trunks are limited to 3 digits.

Dialing Plans

The dialing plan must be consistent throughout a DCS cluster. That is, every extension number must have either four or five digits.

- 4-Digit Dialing Plan:
  - Every switch can be arranged in a Main / Satellite configuration.
  - Every switch (beginning with R2 V4, Issue 1.3*) can be arrainged in an ETN configuration using location code routing.
- Unrestrictd 5-Digit Dialing Plan:
  - Every switch can be arranged in a Main / Satellite configuration.
  - Every switch (beginning with R2 V3†) can be arrainged in an ETN configuration using location code routing (AAR or WCR feature).
- Prefix Digit 5-Digit Dialing Plan:
  - The first two digits (ten-thousands and thousands digits) of an extension number must identify a particular node within the cluster.
  - More than one node may have the same first or second digit as long as both digits are not duplicated on any two nodes.‡

Extension Class of Service

Extension class of service assignments should be identical at each node. That is, extension class of service 1 should have the same permissions and restrictions on every node of a DCS cluster.

Inhibited IEs in Procedure 100, Word 3

For ISDN—PRI trunk groups that also serve as DCS trunk groups, Field 8 of Procedure 100, Word 3 must be set to "0". Otherwise, an FRL (Facilities Restriction Level) TCM (Traveling Class Mark) is not sent with each DCS call, and these calls will fail. However,

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* DIMENSION System Switches and earlier System 85s cannot use unrestricted 4-digit dialing.

† DIMENSION System and System 85 Switches earlier than release 2, Version 3 cannot use unrestricted 5-digit dialing.

‡ If more than one node uses the same first two digits, these switches must be connected to each other in a Main / Satellite arraingement.

since a DCS trunk group is a private-network trunk group, setting the Optional IE Inhibited field to "0" does not result in additional tariff charges for the trunk group.

"Network Trunk" and "Main/Tandem" Fields in Procedure 103

For System 85 and Generic 2.1 switches, fields 4 and 5 of Procedure 103 must be assigned as "1" to route DCS calls through an ETN configuration with feature transparency. This is to force TCMs to be sent. When the Network Trunk field, Field 4, is set to "0" for a DCS trunk group, DCS calls will fail. When the Main/Tandem field, field 5, is set to "0" for a DCS trunk group, DCS feature transparency is lost. This is not necessary for Generic 2.2 switches.

MCS (Message Center Service)

Centralized Message Center Service: Centralized Message Center Service is available for System 85 R2 V4 and DEFINITY Generic 2 switches. With centralized service, message center service is fully transparent.

Release 2, Version 3 Switches and Mixed Clusters: For DCS clusters using System 85, Release 2 switches earlier than Version 4, basic message center services such as, receiving incoming calls and taking messages for a principal, function in a transparent manner. However, if the principal is not located on the same node as the message center AP, the message waiting lamp cannot be used to alert the principal that a message has been received. Also, Unified Messaging Services, Demand Print, and Display-Voice Terminal retrieval functions are not available if messages are not stored on the node where the principal's extension is located.

Network Channel Priority

Network channels are assigned a priority (either high or low) in Procecduer 257, Word 1. This priority is used by the DCIU any time that contention exists for message processing. If contention occurs, high priority network channels are serviced first. In assigning priority to network channels, consider that a message on an alternate routing network channel or one that is being hopped through the DCIU from an incoming DCIU link to an outgoing DCIU link has been in transit for some time already and may be in danger of timing out

Trunk and Trunk Group Numbering

Tie trunks connecting two nodes of a DCS must be assigned the same trunk-group number and trunk numbers at both nodes. For example, a tie trunk (connecting nodes A and B) assigned as Trunk Group 55, Trunk 12 at Node A must also be assigned as Trunk Group 55, Trunk 12 at Node B.

Trunk Group Overflow Restriction

If a DCS cluster uses AAR or WCR to route private-network calls, private-network trunk groups should not overflow to public-network trunk groups. When overflow to the public network occurs, feature transparency is lost and the user will perceive inconsistent operation.

Restricted Trunk Types

The following trunk types should not be used to interconnect the nodes of a DCS cluster:

- 2-way automatic in and dial repeating out (Trunk Type 38)
- 2-way automatic in and automatic out (Trunk Type 39)
- 2-way dial repeating delay dial in and automatic out (Trunk Type 45)
- 2-way immediate start both ways (Trunk Type 72).

Interactions With Other Features

Feature Transparency

The backbone of the Distributed Communications System is transparent feature operation. A feature is transparent when the user cannot detect any difference between the way a feature works in the DCS environnent and in the single switch mode.

Unfortunately, full feature transparency is not always possible. This is because some switches that can function in a DCS have different capabilities, and the same features work diferently on these switches. Full transparency cannot be assured unless every switch in the DCS is of the same type and version.

> **NOTE:** Unless otherwise stated, these interactions are based on the use of System 85, Release 2 and DEFINITY Generic 2 switches. Interactions with other types of switches in the same DCS network may differ.

Abbreviated Dialing

Abbreviated Dialing can be used to dial an extension number in a DCS, whether the extension is local or on a remote node. However, Abbreviated Dialing does operate across a DCS cluster. That is, attributes of the Abbreviated Dialing feature such as system or group lists cannot be used between nodes of a DCS.

Attendant Call Waiting

Transparent for System 85, Release 2 and DEFINITY Generic 2 switches.

Attendant Control of Trunk Group Access

When used on tie trunks between nodes of a DCS, this feature has the effect of disrupting transparency for all features (including normally transparent features) between these nodes. Instead of the call progressing as expected (according to the transparent feature invoked), the call routes to the attendant queue. To control a trunk group from a distant node, the attendant must use a Direct Trunk Group Selection button.

Attendant Direct Extension Selection With Busy Lamp Field

The DXS (Direct Extension Selection) portion of this feature is compatible with the DCS feature as long as the dialing plan is limited to 4-digits. Attendant DXS or Extended DXS cannot be used in a DCS where 5-digit dialing is used. Also, DXS does not work properly in a DCS where one or more of the nodes is an Enhanced DIMENSION System switch.

The BLF (Busy Lamp Field) on attendant consoles does not operate transparently in a DCS environment. The BLF does not display the busy / idle status of extensions residing in different DCS nodes. The BLF only displays the busy / idle status of extensions residing in the same node as the attendant console.

Attendant Display

For the calling number display attribute of the Attendant Display feature to be useful, attendant consoles must be of the 8-character alphanumeric display type. On a console with a 4-character alphanumeric display, only the first (leading) four digits of a 5-digit extension number are shown.

The class of service and ICI (Incoming Call Identification) displays work transparently with either 4-character or 8-character attendant consoles.

AUDIX (Audio Information Exchange)

The basic AUDIX function is transparent in a DCS. That is, a caller can be routed to AUDIX and leave a message without regard to what switches are involved in the connection. Other functions, (such as, Leave Word Calling on AUDIX, call back, and call transfer out of AUDIX) will differ depending on what versions of the switch are used.

AAR (Automatic Alternate Routing)

A DCS subnetwork can be administered as either a Main / Satellite arrangement or as an ETN (Electronic Tandem Network). If a DCS is administered as an ETN, the AAR feature is used on System 85 and Generic 2.1 switches, for internode call routing.

However, DCS feature transparencies ***are not*** provided when a user (within the DCS subnetwork) dials the AAR dial access code + RNX + XXXX to call a party in another DCS node. DCS feature transparencies are ***only*** provided when the calling party dials an extension number [including "soft" Call Coverage extension numbers, DID LDNs, associated extension numbers, and VDNs (Vector Directory Numbers)] or an attendant dial access code.

Automatic Callback

The Automatic Callback feature is transparent for System 85, Release 2 and DEFINITY Generic 2 switches. However, when Call Forwarding is involved, transparency may be disrupted. If the forwarded extension is called from a remote switch, the Automatic Callback call is placed from the forwarded-to extension. However, if when the forwarded-to extension is available, if the forwarding extension is no longer forwarded, the forwarding extension (the originally called extension) will place the callback call. Also, when the user who is establishing an Automatic Callback call to a remote DCS extension is called back, the remote extension could become busy on a new call while callback call is being set up. The Automatic Callback call originator hears busy tone if this happens and Automatic Callback is cancelled.

ACD (Automatic Call Distribution)

In a DCS environment, direct attendant-calls and attendant-extended calls to an ACD or EUCD split (or a UCD or DDC group) on another node are queued like any other incoming ACD call. The attendant does not receive confirmation tone to show that the queue has been entered.

ACA (Automatic Circuit Assurance)

In a DCS environment with as many as 40 nodes, attendant consoles can be concentrated at one node. When an attendant at this node activates or deactivates the ACA feature, the feature is activated or deactivated for every switch in the DCS. In a DCS environment, the operation of ACA is similar to the operation in a non-DCS environment. However, the initial alphanumeric display to identify a faulty trunk is different. In addition to the usual information, the initial display shows dial access code of the tie trunk involved in the ACA referral call.

In a DCS, trunk groups that require the use of the ACA feature across DCS nodes cannot have 4-digit dial access codes.

Busy Verification of Lines

Transparent for System 85, Release 2 and DEFINITY Generic 2 switches.

Call Coverage

Coverage points in a coverage path for a principal must be located on the same node as the principal.

However, if Call Forwarding—Follow Me is active for a group coverage point (an ACD split or a dummy ACD split), calls will cover to the local split's queue and then can ***interflow*** to a distant DCS node according to call forwarding. (This effective DCS coverage operation is primarily recommended to divert coverage calls to a centralized AUDIX or Message Center split.)

The calling terminal from a remote System 85 or DEFINITY Generic 2 node is treated as a local caller. That is, the call covers according to internal coverage criteria, and single-burst ringing is applied. This is not true when a DIMENSION System switch is involved.

The Call Coverage feature is not transparent between DIMENSION System (FP8, Issue 3), System 85, or DEFINITY Generic 2 nodes. The Caller Response Internal and Coverage Tone are not given. Also, the call covers according to external coverage criteria even though the call is internal to the DCS environment. All coverage points in a coverage path must be located on the same node (DIMENSION System) as the principal.

CDR (Call Detail Recording)

A data item is provided in the CDR record that contains the DCS node number. Therefore, if necessary, subsequent accumulation of records from numerous switches carry an identifying field from which a multinode call can be reconstructed. This field contains the value of the node number assigned in Procedure 275, Word 3.

Call Forwarding—Busy and Don't Answer

Not transparent on System 85 or Generic 2 switches.

Call Forwarding—Don’t Answer

Not transparent on System 85 or Generic 2 switches.

Call Forwarding—Follow Me

If a call is forwarded from Node A to Node B, activation of Leave Word Calling is denied toward the originally called voice terminal on Node A. ACD and EUCD splits can use Call Forwarding—Follow Me between nodes of a DCS (Overload balancing, also known as Interflow, is available for this purpose). This includes the use of ACD splits as gateways to the Message Center and AUDIX features.

Call Vectoring

A route to command can be used in vector processing to route calls to a distant DCS node. Beyond this functionality, no DCS transparency is provided. System 85 and DEFINITY Generic 2 do not provide message correspondence over DCIU links that is unique to Call Vectoring. Queue to main split and check backup split commands can only be applied to ACD splits within the local DCS node.

Given these limitations, route to commands could be used to route calls to an associated extension number or VDN at a distant DCS node. The distant associated extension number or VDN can, in turn, terminate to a centralized AUDIX or Message Center queue.

Call Waiting

Transparent for System 85, Release 2 and DEFINITY Generic 2 switches.

CAS (Centralized Attendant Service)

Attendants serving several DCS nodes can all work at one location and still provide a wide range of services. For attendant features to work transparently, there must be at least one direct tie trunk between the attendants’ location and each unattended node.

Conference—Three Party

Transparent for System 85, Release 2 and DEFINITY Generic 2 switches.

DDC (Direct Department Calling)

Not transparent. Interactions are the same as the ACD interaction.

Direct Trunk Group Selection

Transparent for System 85, Release 2 and DEFINITY Generic 2 switches.

Display—Voice Terminal

Outgoing internode calls in a DCS: The displays shows an outgoing trunk call. Either the digits that were dialed or name of the trunk group, if assigned, is displayed.

Incoming internode calls in a DCS: The display shows the name of the calling party as though the call had originated on the local switch if all switches involved in the connection (the calling and called nodes and all tandem nodes) are System 85, Release 2, Version 2 or later. If other types of switches or other versions of the System 85 switch appear in the connection the specific information shown will vary depending on the types of switches involved.

ISDN—PRI Displays

When both DCS and ISDN—PRI are used in an ETN configuration, the Display-Voice Terminal feature uses the DCS display transparencies instead of the ISDN displays.

Extension Number Portability

This feature works in a DCS environment but does not require a DCS. The ENP feature applies to System 85, Release 2, V 3 and V 4, and DEFINITY Generic 2. In a mixed DCS cluster, full portability is limited to a subset of the full cluster called the Portability Subnetwork. See the Extension Number Portability feature for more details.

When Extension Number Portability is used in a DCS, it is not readily apparent where a particular extension number resides. Therefore, users should be aware that nontransparent features may not function when activated toward ported extension numbers.

EUCD (Enhanced Uniform Call Distribution)

Not transparent. Interactions are the same as the ACD interaction.

ISDN—PRI (Primary Rate Interface)

A DCS arrangement can be set up using either ETN (Electronic Tandem Network) or Main / Satellite, trunking arraingements. The ISDN—PRI feature is compatible with ETN trunking arrangements but not with the Main / Satellite confirmation.

This does not mean that a DCS with a Main / Satellite configuration cannot be connect to an ISDN. It does mean the ISDN facilities cannot be assigned as Main / Satellite trunks (Trunk Types 70 to 78) between the Main and Satellite switches. The reason for this is that a Main / Satellite arrangement uses Extension Number Steering. Extension Number

Steering cannot handle ISDN message processing. Either the AAR (Automatic Alternate Routing) and ARS (Alternate Route Selection) features with generalized route selection, or the WCR (World Class Routing) feature, are needed by ISDN call processing.

In a DCS, ISDN—PRI trunking can be used between nodes that are arranged as an ETN. Conventional (non-ISDN) trunking must be used for any Main / Satellite connections. ISDN service can be provided to the Main (assuming the Main is a Release 2, Version 4 switch) but not between the Main and a Satellite switch (unless ETN is also provided between these switches using a parallel trunk group). Interworking at the Main will take care of call processing services for ISDN calls originating (or terminating) on a Satellite station. Figure 53-18 illustrates the various DCS / ISDN—PRI connectivity arrangements.

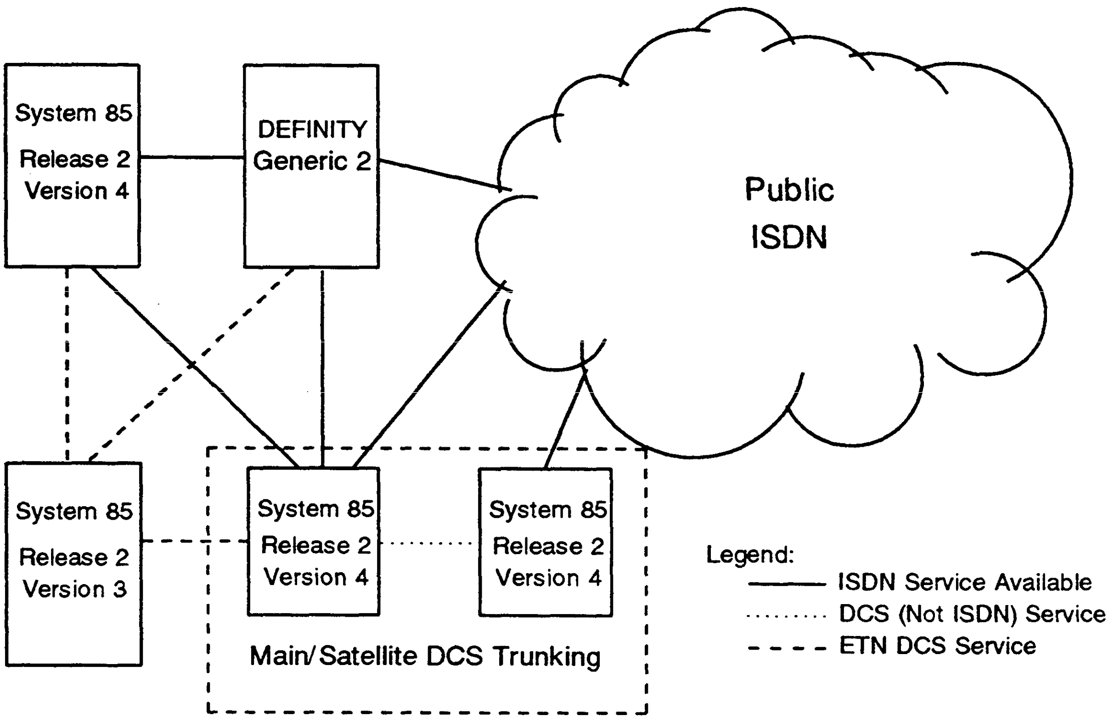


Figure 53-18. DCS / ISDN Network Connectivity

Intercept Treatment

When an incoming call from another switch in the DCS subnetwork arrives at the local System 85 or DEFINITY Generic 2, the Treatment feature considers the source of the call to be private, not internal.

Last Extension Dialed

Last Extension Dialed is not strictly a DCS transparency. However, the Last Extension Dialed feature does operate across a cluster of switches that use a 4- or 5-digit numbering plan.

Last Number Dialed

Last Number Dialed is not strictly a DCS transparency. However, the Last Number Dialed feature does operate across a cluster of switches that use a 4- or 5-digit numbering plan.

LWC (Leave Word Calling)

For System 85 switches prior to Release 2, Version 4, a message can only be retrieved from the node where it is stored. The message indication sent by the System 85 (i.e., light the MESSAGE lamp) can only be sent to a voice terminal on the same switch. A message waiting signal cannot be sent to another node. If the attendant extends a call to an extension on another node, Leave Word Calling cannot be used (also see Call Forwarding—Follow Me and Transfer).

For DCS networks with all System 85, R2 V4 or DEFINITY Generic 2 nodes, the ES message set provides centralized messaging and feature transparency for messaging services including LWC.

Leave Word Calling is not transparent to or from a DIMENSION System, FP8, Issue 3 switch.

LWC on the Switch and Global Retrieval

Global retrieval of LWC messages is not a DCS transparency. Within a DCS cluster, each node can have a global retriever who can access messages stored within the local switch's memory. However, global retrievers cannot access LWC messages outside the local node.

Look-Ahead Interflow

Look-Ahead Interflow can divert calls over DCS facilities that are set up to use ISDN—PRI facilities within routing patterns. To implement Look-Ahead Interflow over DCS facilities, an ETN configuration using 4- or 5-Digit UDP is required, or ENP converts the dialed extension to an RN(X) + XXX(X).

This type of DCS arrangement is required for routing Look-Ahead Interflow calls to a centralized AUDIX or Message Center split.

To prevent the loss of transparency for DCS calls over ISDN—PRI trunk groups, ISDN glare resolution does not allow the Network side to negotiate a channel for the User side after a glare condition. Instead, the Network side blocks the call and sends the User side a message to return reorder tone. If the User side is the sending switch in a Look-Ahead Interflow call, this switch, upon receiving the message, will retry the vector step (if the final effective step) at 2-second intervals. Otherwise, the sending switch will continue vector processing with the next sequential vector step.

Malicious Call Trace

If a malicious call originates from a voice terminal from within the DCS network, the ICI display on the controlling attendant console displays the extension number of the calling voice terminal. Beyond this functionality, DCS transparency is not provided for the Malicious Call Trace feature.

A voice terminal user cannot activate Malicious Call Trace unless an attendant is in the same DCS node as the user. Also, a third party cannot activate the feature unless the third party is in the same node as the user.

When Malicious Call Trace is activated in a DCS environment, every System 85 or DEFINITY Generic 2 attendant in the local DCS node is alerted to trace the call.

Override

The Override feature works only on the local switch in a DCS. If directed toward an extension on a distant DCS node, the caller attempting to use override receives intercept treatment.

Priority Calling

Transparent for System 85, Release 2 and DEFINITY Generic 2 switches.

Restriction—Attendant Control of Voice Terminals

Once applied, Attendant Control of Voice Terminals restrictions work in a transparent manner from the voice terminal user's perspective. However, these restrictions must be activated and deactivated by a local attendant. Pooled or centralized attendants cannot activate or deactivate these restrictions for voice terminals on a distant DCS node.

Restrictions—Voice Terminal Restrictions

This feature is transparent in a DCS environment from the voice terminal user's point of view. That is, a voice terminal restriction, such as inward restriction, applies to the DCS rather than just to the local switch. These restrictions are applied through class of service and must be applied uniformly throughout the DCS. They must be separately administered to each switch within the DCS.

Ringing—Distinctive Ringing

Transparent for System 85, Release 2 and DEFINITY Generic 2 switches.

Tenant Services

A partitioned System 85 or DEFINITY Generic 2 can serve as an endpoint in a DCS cluster. As long as the partitioned switch is serving as an endpoint, an extension partition and an optional attendant partition within the partitioned switch can access the tandem node over dedicated trunk groups. However, a partitioned System 85 or DEFINITY Generic 2 should not serve as a tandem node. When this is done, the tandem does not provide trunk-to-trunk partitioning to the public or private network.

When DCS is used in conjunction with Tenant Services, be careful to ensure that the tenant's organization can access the dedicated DCS trunk group at both ends. Otherwise, DCS feature transparency is degraded. As an example, when a voice terminal user at Node A activates Call Forwarding toward a voice terminal in Node B, the activation is accepted by Node A with confirmation tone. However, if both ends of the tie trunk cannot be accessed by the same tenant organization, a call to the user at Node A will fail durning the forwarding process. Meanwhile, the calling party receives the usual forwarding display and the forwarding party receives ring-ping.

Allowing a single partitioned System 85 or DEFINITY Generic 2 to serve as an endpoint in multiple DCS clusters can be difficult to implement. Within the partitioned switch, each extension partition must have unique extension numbers. Moreover, the endpoint partition's extension numbers must likewise be unique in its DCS cluster. Also, the DCS tie trunks must have the same trunk-group numbers and trunk numbers at both ends.

Transfer

After an internode call is transferred, activation of Leave Word Calling is denied.

Trunk Group Busy/Warning Indicator to Attendant

Transparent for System 85, Release 2 and DEFINITY Generic 2 switches.

Trunk Verification—Attendant

This feature is transparent for System 85, Release 2 and DEFINITY Generic 2 switches. However, the procedures for verifying distant trunks are slightly different than the usual verification procedures (see The Trunk Verification—Attendant feature.) When trunks on a remote switch are involved, only the first 99 trunks can be verified. Also, remote trunks with DACs of more than three digits cannot be verified.

UCD (Uniform Call Distribution)

Not transparent. Interactions are the same as the ACD interaction.

WCR (World Class Routing)

The DCS feature is fully compatible with the WCR feature. When DCS is in effect, either WCR or the Main/Satellite feature must be used for call routing.

The DCS feature provides feature transparency for a limited set of switch features. When used with the WCR feature, both messaging and call routing are based on either ENP (Extension Number Portability) or UDP (Uniform Dial Plan) routing. DCS calls are dialed using an extension number. Calls dialed using a network DAC (standard WCR dialing procedures) are not provided with DCS transparency.

For DCS calls, WCR route selection gives preference to DCS trunks. If DCS trunks are not available, non-DCS trunks are selected and transparency is lost. DCS trunks do not need to be located in an earlier preference for this selection process.

Hardware Requirements

Specific hardware requirements for the DCS feature include a DCIU with appropriate interconnecting cables at each switch and appropriate modems. The following DCIUs are available:

- DCIU, Release 1. Applicable to DIMENSION System switches and System 85 switches earlier than Release 2, Version 2.
- DCIU, Release 2. Applicable to DEFINITY Generic 2 and System 85 switches beginning with Release 2, Version 2.

Feature Administration

Assignment of the DCS feature is on a per-system basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal), TCM (Terminal Change Management) feature, or FM (Facilities Management) feature.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
DISTRIBUTED COMMUNICATIONS SYSTEM

PROCEDURE	WORD	PURPOSE	SMT
010	1, 2, 3, 4	Assigns features and restrictions (including Call Forwarding —Off Net and Miscellaneous Trunk Restriction groups) to a class of service for a voice terminal.	Yes
100	1	Administers the trunk type and queuing parameters for a trunk group.	No
101	1	Administers trunk-group characteristics for trunk groups administered in Procedure 100, Word 1. (Touch-tone in and out should be assigned to non-ISDN DCS trunk groups in Fields 6 and 7.)	No
101	3	Assigns trunk group prefixing for DEFINITY Generic 2.2 switches.	N/A
102	1	Assigns trunk groups (using dial access codes) to Miscellaneous Trunk Restriction groups.	Yes
103	1	Administers network trunk-group parameters including: FRL, authorization code requirements, incoming tie trunk network association, and for Generic 2.1 and earlier, prefixing requirements.	Yes
104	1 & 2	Administers Main/Satellite parameters to a DCS cluster with a 4- or 5-digit dialing plan.	No
116	1	Administers DS1 and ISDN trunk assignments.	No

(Continued)

ADMINISTRATION PROCEDURES DISTRIBUTED COMMUNICATIONS SYSTEM (Continued)			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
150	1	Assigns the equipment location of an analog trunk to its trunk-group number.	No
202	2	Assigns the desired Direct Trunk Group Selection buttons to the attendant console(s) and sets the BUSY/WARNING level for a trunk group.	No
256	1	Administers the DCIU link characteristics.	No
256	2	Administers BX.25 level 2 timers and counters for the DCIU Link.	No
256	3	Administers BX.25 level 3 timers and counters for the DCIU Link.	No
257	1	Administers the components, priority, and alternate routing status of the DCIU network channels.	No
257	2	Administers DCIU ports for the network channels.	No
257	3	Administers DCS node and trunk-group assignments.	No
257	4	Administers the alternate routes associated with a destination map.	No
258	1	Copies translation changes made using Procedures 256 and 257 to working (i.e., machine-used) tables.	No
258	2	Refreshes the DCIU temporary translation (scratch-pad) tables before using Procedures 256 and 257.	No
275	1	Assigns DCIU to the system class of service.	Yes
275	3	Associates the first digit of a 5-digit dialing plan with a branch (switch) number for a DCS cluster in systems earlier than R2 V3. In R2 V3, assigns DCS switch number and number portability.	Yes
276	1	Assigns DCS to the feature-group class of service.	No
350	1	Assigns the first digit of the dialing plan for both dial access codes and extension numbers. It also specifies the number of digits required.	No
350	2	Assigns the feature dial access codes.	No
354	1	Administers blocks of extension numbers for a 4-digit or 5-digit dialing plan. A 4-digit dialing plan uses extension number steering; therefore, set Field 3 to "1" (extensions to dial access codes only). In 5-digit dialing plans for R2 V3, this procedure is also used to identify the node where the extension number resides in a portability subnetwork.	No

Because of major changes introduced in Release 2, Version 4 some new or significantly changed administration procedures are required for the DCS feature. The following apply specifically to System 85, Release 2, Version 4 and DEFINITY Generic 2.

ADMINISTRATION PROCEDURES DISTRIBUTED COMMUNICATIONS SYSTEM, Release 2, Version 4 and Generic 2			
PROCEDURE	WORD	**PURPOSE**	**SMT**
256	1	Assigns DCIU link characteristics. With flexible DCIU port reservations, in R2 V4 and Generic 2, ranges for Field 8, Destination Machine Number are as follows: Range: Use: 1 to 7 AP 1 to 8 AUDIX 1 to 63 DCS Node.	No
257	5	Administers the DCIU port reservations.	No
257	6	Associates the ES port assignments with network numbers for APs, AUDIXs, and DCS nodes.	No
261	1	Correlates the internal AP or AUDIX number (Fields 1 and 2) with the network AP or AUDIX number (from 1 to 99, in Field 7).	Yes

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — DISTRIBUTED COMMUNICATIONS SYSTEM	
PATH NAME	**PURPOSE**
terminal-change system parameters	Assigns the local prefix digit (the first digit in the 5-Digit Dialing Plan). This parameter is on the "access-codes" screen.

The following is the applicable FM path name used with the AP 16.

FM SCREEN — DISTRIBUTED COMMUNICATIONS SYSTEM	
PATH NAME	**PURPOSE**
facilities-mgmt dciu link-assignments	Generates a report of the DCIU link assignments that includes the endpoint type (Applications Processor or switch) and the data rate for each link.

DCIU Administration

Administration of the DCIU involves some complex considerations. For a detailed discussion of this issue, and some typical administration recommendations, see Appendix H of this manual.

Enhanced Uniform Call Distribution

Description

The EUCD (Enhanced Uniform Call Distribution) feature permits incoming trunk calls, local voice terminal calls, and attendant-extended calls to terminate to an idle voice terminal in a group of terminals. The answering positions appear as normal voice terminals to the switch and are also able to place and receive calls in the usual manner. This feature can provide an economical alternative to Direct Inward Dialing for departments that receive a high volume of similarly natured calls. Selected terminal users (agents) can be organized into a split (collection of agents) to allow for balanced call distribution to the agents.

The EUCD call distribution routines serve as the gateway to Message Center, with the Message Center agents residing in an EUCD split. This balances the call distribution to Message Center agents. All of the following EUCD functions can be applied to EUCD splits containing Message Center agents.

The EUCD call distribution routines also serve as the gateway to AUDIX (Audio Information Exchange), allowing effective use of the AUDIX facilities. Appropriate EUCD functions (such as, recorded announcements and music on hold for callers) can be applied to the AUDIX gateway.

To assist in conceptualizing this feature, a sample EUCD arrangement is depicted in Figure 54-2.

Feature History and Development

The EUCD feature was provided in Release 2, Version 2 to enhance the functionality provided by the UCD (Uniform Call Distribution) and DDC (Direct Department Calling) features.

The EUCD feature was itself replaced and enhanced by the ACD feature in Release 2, Version 3. The following feature description is meant to serve as a reference source for those using the Release 2, Version 2 software package.

> **NOTE:** Refer to Appendix B for a tabular comparison of the various call distributors provided in System 85, Release 2 and DEFINITY Generic 2.

General Usage

A published number is linked to an EUCD split by associating the published number with the extension number of the first terminal in the split. The controlling or primary for the split (the split supervisor's terminal) is the first terminal in the split's list.

The switch directs incoming calls for a published number to the split's queue. Two kinds of hunting are available to extend the call to an idle agent linear (also called direct) and circular. Linear hunting always starts with the first agent (split supervisor) and hunts toward the last member. Circular hunting starts where the hunting process left off during the previous scan and continues through the list of agents. After checking the final member of the list, circular hunting again returns to the first member of the list and continues in a circular fashion. Linear hunting is used for applications where a priority series of answering positions is desired. Circular hunting is useful for applications, such as order taking or Message Center, where an evenly balanced call distribution is necessary.

NOTE: Linear hunting provides the call-distribution algorithm that was formerly provided by the DDC (Direct Department Calling) feature. Circular hunting provides the call-distribution algorithm that was formerly provided by the UCD (Uniform Call Distribution) feature.

Each terminal (including the split supervisors terminal) in an EUCD split can receive calls either as a split member or as an individual terminal. For internal calls, unique EUCD split extension numbers, called associated numbers, identify the split. For incoming calls, the EUCD split is associated with incoming trunks. For automatic-in type trunks, the call routes to the assigned EUCD split. For dial repeating type trunks, the call routes to the split dialed. This is similar to the way DID calls complete to individual terminals (including individual EUCD split members). One of the associated numbers and any number of the incoming trunk groups may receive a priority designation. Priority calls are placed at the top of the queue or behind priority calls which are already in queue. A nonpriority call enters the queue behind all other calls.

An example of priority queuing is shown in Figure 54-1. Within this figure, the priority calls are ***lettered*** and the nonpriority calls are ***numbered.*** One associated extension number (the maximum) and two trunk groups are delivering incoming calls to the end of the priority portion of the queue. Meanwhile, two associated extension numbers and two trunk groups are delivering incoming calls to the end of the nonpriority portion of the queue. Each call is distributed from the head of queue (currently, Call A) to an available agent in Split 29.

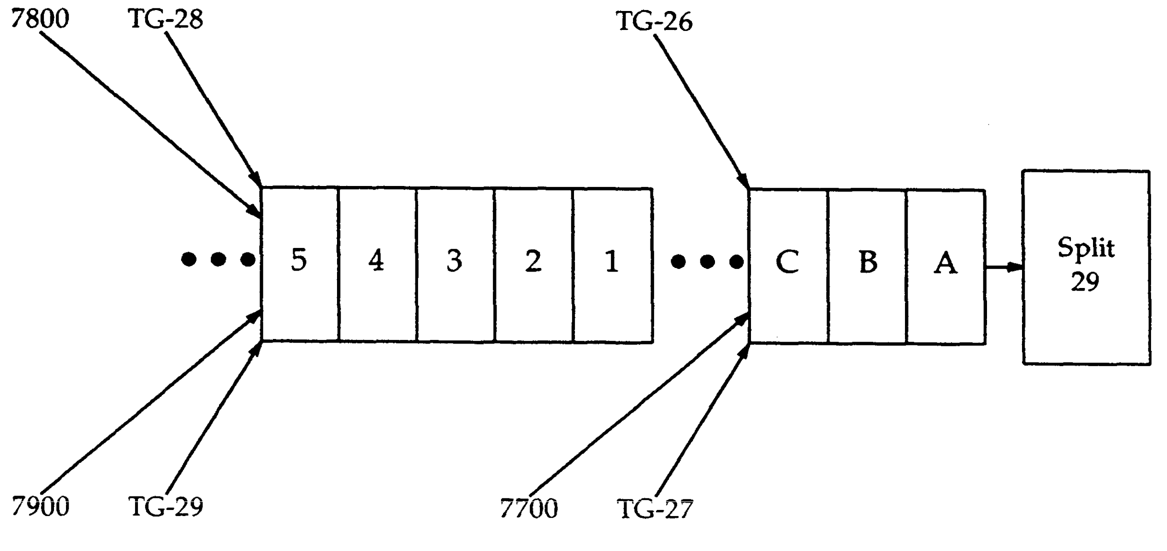


Figure 54-1. EUCD Priority Queuing

Methods of Routing Incoming EUCD Calls

As previously mentioned, there are two ways to route incoming EUCD calls to the System 85 switch:

- Dial-repeating type routing

  Using this method, an associated extension number's digits are passed through the serving switch (usually, the serving Central Office) and to the local System 85 in a similar manner to the way that DID calls are routed. As the associated extension numbers digits are analyzed by the System 85's call-processing software, the dialed number is recognized as a number that terminates to a specific split's queue. In turn, the call-processing software gives control of the incoming call to EUCD processing.

- Automatic-in type routing

  Using this method, an EUCD call is recognized by the serving switch (usually, the serving Central Office) as a call that is routed to the local System 85 over a specific trunk group. In turn, the local System 85 accepts the call from over the incoming (or 2-way) trunk group and recognizes the call as assigned to terminate to a specific split's queue. This method of routing resembles "non-DID routing" to the attendant queue. In fact, "attendant-completing" trunk groups are often used for automatic-in type routing of EUCD calls. Automatic-in type trunk groups are assigned as type "attendant-completing in" or "automatic in" in Procedure 100, Word 1 and then assigned to terminate to an EUCD split in Procedure 115.

  The trunk types that can be assigned to terminate to an EUCD split in Procedure 115 include:

  — 16 = CO 1-way in attendant-completing

  — 19 = CO 2-way attendant-completing in/DOD out

  — 20 = CO 2-way with party test attendant-completing in/DOD out

  — 21 = FX 1-way in attendant-completing

  — 24 = FX 2-way attendant-completing in/DOD out

  — 25 = FX 2-way with party test attendant-completing in/DOD out

  — 26 = WATS 1-way in attendant-completing

  — 35 = TIE 1-way in automatic

  — 38 = TIE 2-way automatic in/dial repeating out

  — 39 = TIE 2-way automatic in and out

Outgoing Calling by EUCD Agents

This EUCD feature description primarily focuses on EUCD from the perspective of incoming calls. However, especially when EUCD is applied to address the needs of a telemarketing organization, agents can also devote significant effort placing outgoing calls (for example, to solicit sales). In this environment an EUCD manager is also concerned with the costs of outgoing calling. Although the EUCD feature does not directly address these concerns, a number of coresident features on System 85 do.

As previously mentioned, EUCD agents in addition to receiving calls distributed from a split's queue, are able to place and receive calls in the usual manner. Given this ability, EUCD agents have the same access as other System 85 users to the least-cost-routing capabilities of the ARS (Automatic Route Selection) and AAR (Automatic Alternate Routing) feature. Indeed, the switch administrator can control the costs of EUCD outgoing calling with the ***same*** methods that costs are generally addressed on System 85, R2 V2.

Lamp Monitoring of EUCD Agents

Display units (106B) can be provided to show the current activity of 20 EUCD agents. For each agent, the unit displays one of the following agent states.

- The agent is available to handle an EUCD call. [First lamp (top)]
- The agent is handling an EUCD call. [Second lamp]
- The agent is engaged in after call work. [Third lamp]
- The agent is in the Aux-Work mode. [Fourth lamp]
- The agent is not engaged in work-related activity (receiving a personal call or placing a call). [Fifth lamp (bottom)]
- The agent's position is in the Unstaffed mode. [All lamps off]

Answer Supervision

Answer supervision is a signal sent by switch to the serving CO (Central Office) indicating that an incoming call has been answered. Upon receiving this signal, the originating CO (generally) begins tracking toll charges for the call (if charges apply). For EUCD calls to System 85 which enter a queue before being answered, answer supervision must be returned (at the latest) either just before an agent actually answers or just before the first recorded announcement (if provided), whichever comes first. This is the ***preferred operation,*** and is provided automatically by R2 V2 (beginning with Issue 1.6).

For earlier issues of R2 V2, the preferred operation is provided on a system-wide basis when Field 12 of Procedure 275, Word 4 is set to "1". When this field is left with the default entry "0", System 85 returns answer supervision for EUCD calls at the time each call first enters queue. Since billing is for a longer period of time, this operation would result in higher toll charges for the customer (when 800 Service is used) and/or for each calling party.

Answer Supervision and Abandon Call Search

As discussed, the switch returns answer supervision to the serving CO either at the time an incoming call first enters the split's queue or just prior to connecting the call to the first recorded announcement. However, after answer supervision has been returned, "ghost calls" can occur if the calling party abandons the call.

A ghost call occurs whenever an agent answers a call ***after*** the calling party hangs up, but ***before*** the CO returns the disconnect signal to the System 85. Primarily depending on the type of CO, the delay in returning disconnect is on the order of 2 to 25 seconds. If a ghost call were to occur, an ineffective call would be distributed to an agent. Also, as if the agent had received an actual EUCD call, the agent would be inappropriately credited with an EUCD call and bypassed for distribution of subsequent EUCD calls.

To minimize this problem, abandon call search can be assigned to the switch (Procedure 275, Word 4). When abandon call search is assigned, the switch checks incoming CO, 800 Service, and FX trunks just before ringing an idle agent. If the trunk is found to be on-hook at the CO, the switch releases the trunk. If the trunk is active, the switch distributes the call to the idle agent.

> **NOTE:** It is recommended, whenever processor occupancy allows, that abandon call search be assigned to the switch.

Split Supervision

Supervision of an EUCD split is normally provided by a split supervisor. The split supervisor may be an experianced agent who performs agent training duties, serves as a consultant to a group if agents, and regulates the split's operation. In addition to the usual agent capabilities, the split supervisor is able to:

- Observe agent performance using agent override
- Add or remove agents to/from the split
- Verify the split's recorded announcement
- Activate or deactivate Call Forwarding for the split
- Activate or deactivate overload Balancing for the split
- Turn of the reload warning lamp after a tape reload by the switch.

Also, one local attendant can be designated as the system supervisor. This attendant console cannot perform agent duties, but may perform the following tasks:

- Activate or deactivate Call Forwarding for a split
- Turn off the reload warning lamp after a switch tape reload.

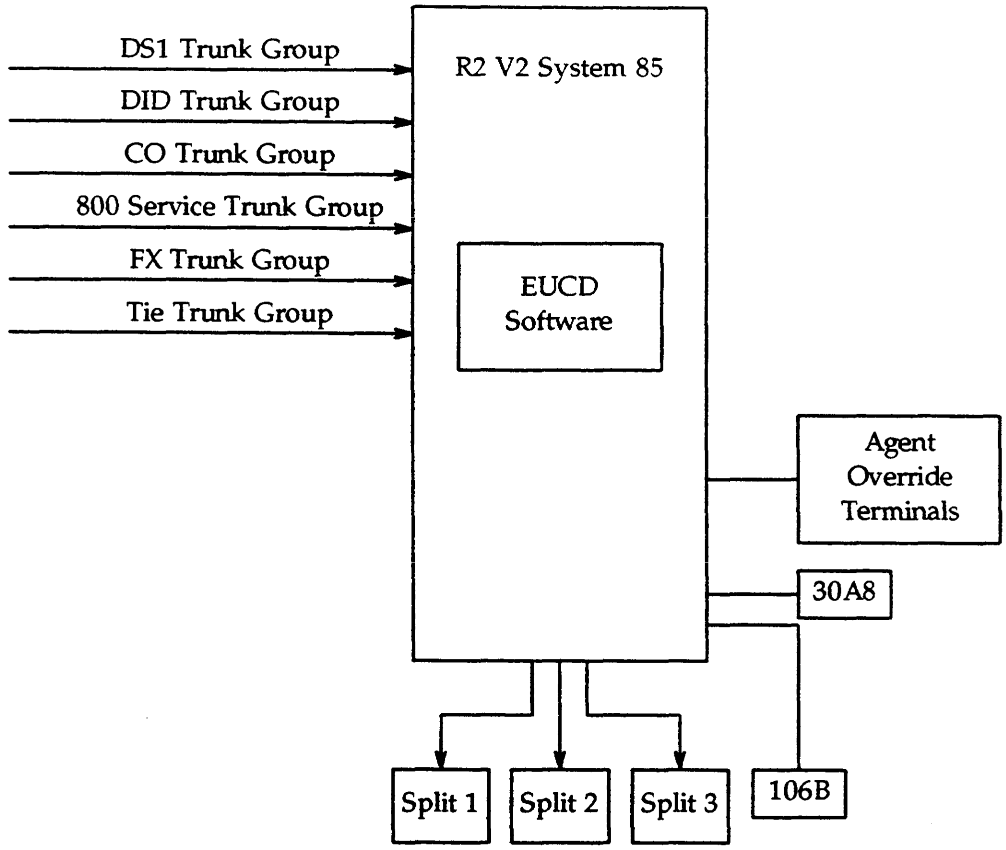


Figure 54-2. Simplified Pictoral of an EUCD Arrangement

Intraflow and Interflow

Intraflow, redirection of EUCD calls to a local destination and interflow, redirection of EUCD calls to a distant network node, are available to increase call-distribution flexibility. Call forwarding is used as the software mechanism for intraflow (local redirection), while overload balancing is used as the mechanism for interflow (distant redirection). Both ***intraflow*** and ***interflow*** can operate two ways: ***threshold*** redirection and ***unconditional*** redirection (redirection of ***all*** calls). Threshold redirection is used to divert EUCD calls during a heavy call load. Unconditional redirection is used to divert all calls when an EUCD split is inactive.

NOTE: When interflowed EUCD calls use the AAR or ARS feature to route these calls outside the local switch, the FRL of the split's supervisor is used to determine whether these calls can use the available network facilities.

"Chaining" of Destinations for Intraflow—All

When Intraflow-All is activated, the split can have as many as three local destinations (such as, an extension, the attendant group, or another EUCD split) for unconditional redirection of EUCD calls. These destinations are arranged in a priority scheme by order of activation. If the first destination is unavailable, the switch checks the second and third destinations (while the call remains in the first split's queue). Of these three destinations, only the last destination should either be another EUCD split or the attendant group (that is, be accessed by a queue).

NOTE: When a destination before the last destination has a queue, the call unconditionally enters the queue, and subsequent destinations are never checked.

If an unconditionally intraflowed call enters an EUCD queue, and this queue also has Intraflow—All active, the call will recursively intraflow again. This time, the call intraflows to one of the new split's destinations.

If an unconditionally intraflowed call enters an EUCD queue, and this queue has Intraflow **—** ***Threshold*** active, the call will not intraflow again. Instead, this call is marked as "Intraflow—Threshold Not Allowed", and must be answered by an agent in this split.

"Chaining" of Destinations for Intraflow—Threshold

There can also be three local destinations (for example, an extension, the attendant group, or another EUCD split) for threshold redirection of EUCD calls. These destinations are arrainged in a priority scheme by order of activation. If the first destination is unavailable, the switch checks the second and third destinations (while the EUCD call remains in the first split's queue). AU three of these destinations can be EUCD splits (that is, accessed by a queue). When the inflow parameter at a receiving split is met, the switch executes the intraflow operation and marks the call as "Intraflow—Threshold Not Allowed". These calls are not allowed to intraflow again. EUCD calls are not allowed to conditionally intraflow throughout the switch in a recursive manner.

NOTE: Intraflow—Threshold operates to attenuate heavy call loads while the split has ***active*** EUCD agents to answer the calls. So, when the parameters for the first intraflow operation are carefully chosen, the need to redirect the same EUCD call again is minimal.

If a conditionally intraflowed call enters an EUCD queue, and this queue has Intraflow—***All*** active, the call will recursively intraflow again. This time, the call intraflows to one of the new split's destinations.

Parameters for Intraflow Threshold Redirection

Threshold (overflow) redirection operates when the number of calls that are waiting in a queue is greater than or equal to the preset overflow level. A split's overflow level can be ste to any value between 1 and 99. However, if the destination for redirection is an EUCD split, the call to be diverted will only be accepted into queue if the number of calls in queue is less than or equal to the destination split's inflow level and there is ***at least one staffed agent*** in the destination split. A split's inflow level can be set to any value between 0 and 98. When overflow redirection does occur (to one of the destinations that is checked at 2-second intervals), the switch diverts the first call in queue.

NOTE 1: When an inflow level is set to "0", a call will not be accepted unless there is ***at least one available agent*** in the destination split.

NOTE 2: The same field that assigns the overflow level (Procedure 026, Word 1, Field 5) also determines the threshold amount of queued calls that causes the split's queue warning lamp (on the 30A8 panel) to light.

For example, split A has an overflow level of 21 and split B has an inflow level of 5. Presently, split A is quite busy. Every agent in split A is busy on a call and the queue contains 20 waiting calls. The next incoming call to split A will attempt to enter split As queue. If split B has five or less calls in queue, the incoming call will enter split A's queue and the first call in split A's queue will redirect to split B. If split B has six or more calls in queue, redirection can occur to an alternate destination (if assigned). If every destination for redirection of EUCD calls is busy, the incoming call enters the original queue as the 21st call.

NOTE: Assigning the overflow level for split A to "1" would have the effect of emptying the queue. This operation for diverting EUCD calls would be applied to the ***first call*** attempting to enter the queue.

Interflow With EUCD

Each split can have one destination outside the local switch for diversion of EUCD calls. This destination is provided via Overload Balancing. Overload Balancing is always automatically implemented as the last priority. The Overload Balancing destination can be used alone or after the list of local destinations. As such, there can be as many as four possible destinations.

Threshold interflow diversion is always provided using the Overload Balancing—Overflow access code (Encode 85), whereas unconditional interflow redirection is always provided using the Overload Balancing—All access code (Encode 84).

Using AAR/ARS With Overload Balancing

The AAR and ARS features are used in conjunction with Overload Balancing to divert EUCD calls outside the switch. Any 7-digit, 10-digit, or international telephone number can be specified as an Overload Balancing destination as long as an AAR/ARS pattern has been translated for the destination. (Normally, Overload Balancing is used to divert EUCD calls over an ETN trunk group to an associated extension number of an ACD/EUCD split residing in a different ETN node.)

As the interflow software sends Overload Balancing calls to the AAR/ARS features for routing, these Overload Balancing calls can be allowed to route over ***any*** preference (trunk group) within the routing pattern. (The preferences actually allowed are determined by the split supervisor's FRL.)

If every accessible preference within the pattern is busy, Overload Balancing returns the interflowed call to the head of the queue.

Overload Balancing ignores the Authorization Code routing parameter as interflow calls are being routed by the AAR/ARS software. Overload Balancing will circumvent this parameter allowing AAR/ARS to route interflowed calls without calling-party intervention.

Activating Overload Balancing

The digit strings for Overload Balancing destinations can take one of these forms:

- 1-Digit AAR Access Code + RNX (Location Code) + XXXX (Extension Number)
- 1- to 4-Digit ARS Access Code + RRX (Office Code) + XXXX
- 1- to 4-Digit ARS Access Code + NPA (Area Code) + RNX (Office Code) + XXXX
- 1- to 4-Digit ARS Access Code + International Telephone Number.

The digit strings for Overload Balancing destinations ***cannot*** be

- An extension number within the DCS network
- 1- to 4-Digit Trunk-Group Access Code + Destination Telephone Number.

As the split supervisor activates Overload Balancing the switch thoroughly validates the digits of the destination telephone number. As two examples, if the digits are outside the appropriate numbering plan, or if an AAR/ARS pattern has not already been translated to route interflow calls to this destination, the switch retuns intercept tone to deny the activation.

The split supervisor can divert Overload Balancing calls to a ***default*** destination. After establishing the default destination, the split supervisor need only dial "#" during the Overload Balancing activation to divert calls to this telephone number.

Refer to the "User Operations" section of this EUCD feature description to review the step-by-step Overload Balancing operations for the split supervisor.

Overload Balancing at the Receiving ETN Node

If the AAR feature routes an interflow call to an ACD/EUCD split residing in a different ETN node, the inflow threshold of the receiving split is ***not*** checked. These interflowed calls are unconditionally accepted. Also, the original priority level of the EUCD call is not passed with the interflowed call, and the recorded announcement of the originally called split is not played for the calling party (unless it was already played at the sending switch).

If an interflow call routes over an ETN trunk group (Trunk Types: 41, 42, 43, 44, 45, 46, and 47), the receiving switch provides the answering agent with 3-burst zip tone to indicate an interflowed call. (The receiving switch infers that incoming EUCD calls from over an ETN tie-trunk group were interflowed from another ETN switch.)

A city-of-origin announcement can be assigned to an incoming tie-trunk group at the receiving switch. If this announcement is combined with the 3-burst zip tone for an incoming ETN trunk group, the zip tone and the announcement will identify incoming calls from over the trunk group as interflow calls with a specific source.

Interflow calls routed over ETN trunk groups can be given priority queuing at the receiving switch by activating Overload Balancing to divert the interflow calls to the ***priority*** associated extension number at the sending switch.

Also, when automatic tie-trunk types are used, priority queuing can be assigned to these incoming tie-trunk groups at the receiving switch to give these interflowed calls preferential treatment within the ACD/EUCD queue.

Split Membership

The split supervisor is able to regulate membership in the supervisor's own split by adding and removing agents to/from the split. Furthermore, agents can be transferred from one split to another, provided the two split supervisors coordinate the transfer. Before adding or removing an agent to/from a split, the agent must be placed in the unstaffed mode.

Agent Override

An agent override function is available which allows a local terminal user to enter an agent's call. Agent override is useful as a tool to observe the call handling performance of the agents. However, the agent must be actively engaged in a call. Agent override is activated by a dial access code. An optional warning tone is available and is provided by the use of a different dial access code.

Using either agent override dial access code, an audible 2-way connection is always provided by the switch.

EUCD From the Calling Party's Perspective

Recorded Announcements for Caller

When there is a delay in answering the call, either one or two optional recorded announcements inform the calling party of the delay before the call completes. The first recorded announcement can provide a unique message for each EUCD split. The second recorded announcement is system-wide and assures the caller that the call has not failed.

The timing interval proceding the first recorded announcement is set from 2 to 30 seconds (by 2-second increments). The calling party is always connected to the beginning of an announcement. Therefore, the timing between placing the call in queue and actually hearing the announcement can be as much as the timing interval chosen (from 2 to 30 seconds) plus the duration of the announcement.

The second recorded announcement is also available for each split that uses a first recorded announcement. The timing interval between the end of the first announcement and the beginning of the second announcement is set from 2 to 30 seconds (by 2-second increments). The calling party is always connected to the beginning of the announcement. Therefore, the actual timing between announcements can be as much as the timing interval chosen (from 2 to 30 seconds) plus the duration of the second announcement.

If an agent becomes available at any time before, during, or after a recorded announcement, the switch removes the announcement (if necessary) and sets the call-distribution process in motion.

NOTE: The System 85 only provides a delay announcement for incoming EUCD calls when there is at least one staffed agent in the associated EUCD split. The calling party continues to hear ringback. Otherwise, the delay announcement would encourage calling parties to wait when no agents are available to answer their calls.

Music-on-Hold

Music-on-Hold is also optional and may be provided to callers who are waiting in queue. A music signal provides continuous audible feedback indicating that the connection is still in effect.

When Music-on-Hold is provided for EUCD queues, music is also provided (on a per-switch basis) for the Hold, Conference—Three Party, and Transfer features.

Tabular Representation of Available Choices

The following table shows the calling party interfaces provided by EUCD. Generally, each EUCD split can use a different option. However, the Music-on-Hold option is assigned on a system-wide basis. If this option is assigned, incoming calls to every split receive music. When the option is not assigned, incoming calls to every split do not receive music.

A split cannot be provided with the system-wide second recorded announcement unless the split is also provided with a first recorded announcement. An easy way to circumvent this problem, if desired, is to provide a first recorded announcement with similar wording to the system-wide announcement.

TABLE 54-A. Available Choices for Customer Interface

Option	Calling Party Hears*
No Announcement	Ringback
First Recorded Announcement (No music provided)	Ringback, Announcement, Silence
First and Second Recorded Announcements (No music provided)	Ringback, First Announcement, Silence, Second Announcement, Silence
First Recorded Announcement (Music provided)	Ringback, Announcement, Music
First and Second Recorded Announcement (Music provided)	Ringback, Fist Announcement, Music, Second Announcement, Music

* When an agent becomes available for the call, the switch will immediately discontinue ringback, either announcement, or the choice of silence or music in order to distribute the call.

Zip Tone

Agents in an EUCD split can receive zip tone before connecting to an EUCD call. A single burst of zip tone designates a call dialed directly to the agent's split (one burst of zip tone is only provided for agents with automatic answering). Two bursts of zip tone designate a call redirected from a local system split. Three bursts of zip tone designate a call that routed to the split via a trunk group in the network (that is, by overload balancing).

NOTE: Zip tone is not provided for non-EUCD calls (calls that do not pass through a split's queue before terminating to an idle agent's voice terminal). Instead, non-EUCD calls terminate to the first idle appearance on an agent's voice terminal with ringing.

Queue-of-Origin and City-of-Origin Announcements

After hearing zip tone, the agent may receive either a queue-of-origin or city-of-origin announcement. The calling party cannot hear either the zip tones or the announcement.

When implemented, the queue-of-origin announcement is heard when intraflow redirects an EUCD call to an agent in another local split. When implemented, the city-of-origin announcement is heard when interflow redirects an EUCD call to an agent at a distant node of the DCS or ETN network.

The queue-of-origin and city-of-origin announcements are 1.5 seconds long, run continuously, and are connected to an agent on a barge-in basis. Pressing an optional REPEAT button repeats the city-of-origin announcement.

NOTE: When recording "continuous" announcements (city- and queue-of-origin), the message (such as "Detroit") should be repeated for the ***full duration*** of the announcement set's announcement. As an example, the 13A announcement set provides 24-second announcements. "Detroit" should be repeated ("Detroit, Detroit, Detroit, ...") for the full 24 seconds. In this way, agents will more readily hear the announcement, and the number of "reset intends" between machine playbacks is reduced.

Display Capabilities for Agents

The functionality provided by the Display—Voice Terminal feature can be applied to uses for EUCD agents. These visual capabilities can replace and / or supplement the traditional queue-of-origin and city-of-origin announcements.

NOTE: The CALLMASTER, 7406D, 7407D, or 7405D voice terminal would provide the necessary display functionality for an EUCD agent.

The display capabilities that are available for use by EUCD agents are as follows:

- Queue-of-Origin Display

  When an EUCD call is redirected to a local EUCD split, the answering agent receives a visual display identifying the originally called split. The queue-of-origin display is obtained by assigning an appropriate name (such as, Sales Dept.) to the queue directory number or the split supervisor's individual extension number for the originally called split.

- City-of-Origin Display

  When an EUCD call is redirected to a distant EUCD split, the answering agent receives a visual display identifying the city of the originally called split. The city-of-origin display is obtained by assigning an appropriate name (such as, Chicago) to the incoming trunk group at the distant node.

- Calling Number Display

  When a call from within the DCS network is directed to an agent's individual extension number, the agent receives a visual display identifying the calling party or department. This is useful when an agent needs to correspond with other individuals or departments within the company.

- Dialed Number Display

  This function displays the dialed number when an agent places a call. This display helps to ensure accuracy in dialing.

- Elapsed Time Function

  This function displays the amount of time elapsing since the ELAPSED TIME button was pressed. The Elapsed Time function operates during calls and between calls. Therefore, an agent can use this clock as a tool for pacing the agent's handling of calls and after call work.

- Calculator Function on 7407D

  Calculator functionality is provided with the 7407D voice terminal that allows an agent to conveniently perform any necessary arithmetic computations. The calculator on the 7407D can operate during calls and between calls. Therefore, an agent can use the calculator both as a tool for quoting prices to customers and as an aid for after call work.

DNIS (Dialed Number Identification Service) Without Call Vectoring

In the initial availability of DNIS, EUCD agents equipped with a display voice terminal (such as, CALLMASTER, 7405D, 7406D, or 7407D) receive visual displays that specify the dialed number for calls terminating to the agent's voice terminal. The Call Forwarding—Follow Me feature or the Call Coverage feature (using the "All" criterion) is used to provide the redirection needed to deliver this functionality.

In traditional EUCD arrangements, groups of agents are organized into "splits" (functional groups of answering positions). Under this approach, an agent is trained to answer calls for one specific purpose in an efficient and professional manner. However, EUCD managers are recognizing the need to relax this concept of limiting each split to one call-answering task.

The alternative is to provide splits where each group of agents is proficient with several types of calls. The desired gain is to provide adequate service for the several call types with fewer agents and with less administrative intervention by the EUCD manager. Using this approach, the changing staffing needs of the several call types are averaged in time, and enough agents are staffed to provide adequate service for the prevailing average load. Where five agents might be needed in each of three smaller splits (15 agent total) to handle three types of calls, only 11 or 12 agents might be needed in the single (more general) split.

This idea of averaging the call-handling load is sound for certain applications, but the goal of improved agent efficiency is more readily achieved with the DNIS capability. The DNIS function of the EUCD feature allows each answering agent to know the purpose of each incoming call as the call terminates to the agent's voice terminal. As a result, the natural efficiencies of the single split / single call type arrangement are not compromised. With the calling number display provided by DNIS, agents are aware of each call's purpose and can answer each incoming call with the appropriate greeting. Agents need not invest time merely to determine the purpose of calls.

The following table shows sample displays that an EUCD agent might receive.

TABLE 54-B. DNIS Display Information

Type of Call	Display
Inside call	`a=R JONES to CLAIMS                  f`
Outside call	`a=OUTSIDE CALL to SALES              f`

Call Forwarding—Follow Me Configuration

Using this configuration, a "dummy" extension (not a dummy split) is assigned in Procedure 000 (without assigning an equipment location) for each call purpose within a split. These extensions are, in turn, forwarded to the split's associated extension number. A calling party dials the number of the dummy extension, and the call is forwarded to the split's queue.

> **NOTE:** Since Call Forwarding can be activated for the extension from an attendant console, an actual voice terminal and line circuit are not required to complete the configuration. After these Call Forwarding relationships are established from the console, a "Run Tape" operation should be performed (using the MAAP or SMT) to make these relationships permanent.

Call Coverage (All) Configuration

Using this configuration, a "dummy" extension (not a dummy split) is assigned in Procedure 000 (without assigning an equipment location) for each call purpose within a split. Also, these extensions are assigned to the appropriate coverage group in Procedure 000, Word 2. These extensions are, in turn, redirected using Cover All (Procedure 011, Word 1) to the split's associated extension number. A calling party dials the number of a dummy extension, and the call is redirected to the split's queue.

NOTE: Since the Call Coverage configuration is an administered relationship, an actual voice terminal and line circuit are not required to complete the configuration. However, as a normal aspect of the Call Coverage feature, internal callers to the EUCD split will receive coverage tone (followed by the system-wide caller response interval) before the call enters the split's queue.

Agent Convenience

Multibutton voice terminals are recommended for use with EUCD. The feature buttons provide convenience and flexibility for the agents. The agent's mode of operation can be altered without interruption to callers, and the agent is freed from remembering numerous dial access codes. Furthermore, the status lamps on the terminal provide a visual reminder of the current agent mode.

NOTE: Refer to Table 54-C for a cross-reference of applicable feature buttons and dial access encodes.

Automatic answering is available and recommended to provide hands-free operation for the EUCD agents. With automatic answering assigned to a terminal, incoming calls automatically y connect to the terminal without ringing. The call automatically disconnects from the agent's terminal when the calling party disconnects. In the AUTO-IN mode, the agent is available for the next call immediately after disconnecting from the previous call.

NOTE: When automatic answering is used, it is preferable to provide headsets for the agents. However, automatic answering can be used in conjunction with voice terminal handsets.

Automatic answering combined with handset usage may result in unexpected and somewhat inconvenient agent operation.

The primary effect is that going on-hook (hanging up the handset) places an agent's position in the unstaffed mode, and the agent must manually staff the position to receive another EUCD call. In order for handset usage to be effective, an agent must wait to answer calls with the handset continuously held to the agent's ear.

The secondary effect is that on straight line sets (where handset operation is usually tried) there is no RELEASE button available. Agents must wait (approximately 6 seconds) for the automatic disconnect process to finish before returning to an available state. (If the agent chooses to go on-hook instead of waiting, the voice terminal returns to the unstaffed mode.)

The buttons that are available for use by EUCD agents are as follows:

- APPEARANCE BUTTONS (usually, one or two)

  Agents use these buttons to place and answer calls. Two status lamps (red and green) are adjacent to each appearance button. The red lamp lights when an agent presses an appearance button. The green lamp flashes to alert an agent to an incoming call. Since incoming EUCD calls always terminate to the first appearance, only one appearance is necessary to perform basic agent duties. However, two appearances provide an expanded functionality for agents. With the second appearance, an agent can more readily receive calls to the agent's individual extension number (e.g., from another department), transfer calls (e.g., to another particular agent), and place calls. A third appearance can be provided when an agent has a special need to place calls.

  To provide an appropriate functionality for the agents, the recommended assignment of the appearance buttons is as follows.

  1. Terminating and originating capabilities
  2. Terminating and originating capabilities
  3. Originating (only) capabilities.

- ASSIST

  The ASSIST button is an Abbreviated Dialing button with the split supervisors individual extension number as the stored number. An agent uses this button to request help from the split supervisor. Pressing this button places a call from the agent to the split supervisor.

- AUTO-IN

  This button is pressed when an agent wants to receive calls in the automatic mode. In the automatic mode, an agent is available to receive an EUCD call immediately after disconnecting from the previous call. Activating this mode (pressing the AUTO-IN button) automatically releases all other modes.

- AUX-WORK

  This button allows an agent to discontinue receiving EUCD calls. Often, this mode is active during breaks from the work schedule (e.g., lunch, coffee breaks, etc.). Pressing the AUX-WORK button automatically releases all other modes.

- HOLD

  This button allows an agent to place an active call on hold. When a call is placed on hold, the EUCD feature will not distribute new calls to that agent until the held call is released or transferred.

- LWC

  This button activate the Leave Word Calling feature. The EUCD agents can quickly and easily leave standard messages with the Leave Word Calling feature. Pressing the LWC button leaves a "return call" message for a called party within the DCS network.

- MANUAL-IN

  Pressing this button allows an agent to receive a single EUCD call. In the Manual-In mode, an agent is considered unavailable to receive another EUCD call upon disconnecting from a call. In this way, the agent can finish call-related paper work or do follow-up work before accepting another call.

- MUTE

  Pressing this button on the CALLMASTER voice terminal turns off the EUCD agent's voice transmission and sidetone in the headset. (The EUCD agent can still hear the calling party.) Another press of the MUTE button returns the agent's voice transmission and sidetone to the connection. Also, disconnecti.ng the headset deactivates the muting function.

  Pressing the RELEASE button does not deactivate the muting function.

- RELEASE

  Pressing this button releases any type of call in progress at an answering position.

  > NOTE The RELEASE button should not be assigned to an agent's (or observer's) voice terminal unless automatic answering (Procedure 052, Word 1) ***is also assigned*** to the voice terminal.

  Well trained agents routinely press the RELEASE button just after the closing salutation to the calling party. If the RELEASE button is not pressed, the duration of the automatic disconnect process is on the order of 6 seconds. However, using the RELEASE button, the disconnect process completes in less than 1/10 of 1 second.

  The time savings accrued from regular use of the RELEASE button provides several advantages:

  — Reduced charges for toll trunk facilities

  — Quicker distribution of the next call to an agent in the AUTO-IN mode

  — More reliable 106B lamp display of the current agent state (that is, agent active on EUCD call, or agent engaged in after call work) for an agent in the MANUAL-IN mode.

- REPEAT

  Pressing this button repeats the optional city-of-origin announcement for an agent.

- STAFFED

  This button is pressed by an EUCD agent to indicate to the switch that the agent's voice terminal is in the occupied mode. Another press of the STAFFED button would return the agent's position to the unoccupied mode. When the STAFFED button is pressed and the position becomes occupied, the position is placed in the AUX-WORK mode; pressing the AUTO-IN or the MANUAL-IN button would allow an agent to receive an EUCD call.

For EUCD agents using voice terminals equipped with headsets, plugging in the headset automatically places an agent in the occupied mode. Removing the headset automatically places the agent in the unoccupied mode.

buttons that are available for use by split supervisors are as follows:

- ADD AGENT

  The ADD AGENT button is an Abbreviated Dialing button with the Member Add access code as the stored number. When the split supervisor presses this button, the Member Add access code is automatically dialed.

- AGENT OVERRIDE

  The AGENT OVERRIDE button is an abbreviated Dialing button with an Agent Override access code (with or without warning tone) as the stored number. Using this button, the split supervisor or an observer can enter an agent's call to observe the agent's work performance.

- DELETE AGENT

  The DELETE AGENT button is an Abbreviated Dialing button with the Member Delete access code as the stored number. When the split supervisor presses this button, the Member Delete access code is automatically dialed.

- INTERFLO ALL

  The INTERFLO ALL button is an Abbreviated Dialing button with the Overload Balance All access code as the stored number. When the split supervisor presses this button, the access code for unconditional distant forwarding of EUCD calls is automatically dialed.

- INTERFLO THRESHLD

  The INTERFLO THRESHLD button is an Abbreviated Dialing button with the Overload Overflow access code as the stored number. When the split supervisor presses this button, the access code for overflow distant forwarding of EUCD calls is automatically dialed.

- INTRAFLO ALL

  The INTRAFLO ALL button can be a Call Forwarding—Follow Me feature button. The split supervisor uses this button to activate unconditional local forwarding of EUCD calls.

This button can also be an Abbreviated Dialing button with the Call Forwarding—Follow Me access code as the stored number. When the split supervisor presses this button, the access code for unconditional local forwarding of EUCD calls is automatically dialed.

- INTRAFLO THRESHLD

  The INTRAFLO THRESHLD button can be a Call Forwarding—Busy and Don't Answer feature button. The split supervisor uses this button to activate overflow local forwarding of EUCD calls.

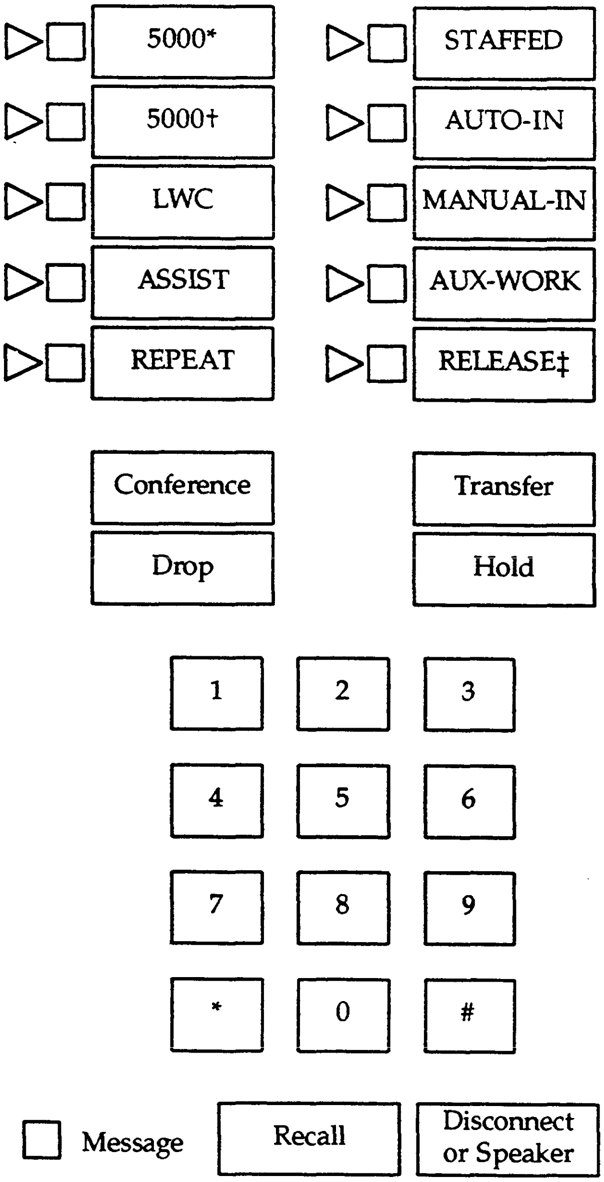
  This button can also be an Abbreviated Dialing button with the Call Forwarding—Busy and Don't Answer access code as the stored number. When the split supervisor presses this button, the access code for overflow local forwarding of EUCD calls is automatically dialed.

- VERIFY ANNCT

  The VERIFY ANNCT button is an Abbreviated Dialing button with the Announcement Verify access code as the stored number. When the split supervisor presses this button, the access code to verify the split's recorded announcement is automatically dialed.

The recommended button layouts for EUCD voice terminals are shown in Figures 54-3 through 54-11. These figures are provided to clarify the feature description and to assist in implementing the agents' sets.

Figures 54-3, 54-4, 54-5, and 54-6 show the recommended button configurations for convenient handling of EUCD calls. Figure 54-3 shows the recommended layout for a 16-button voice terminal. Figure 54-4 shows the layout for a CALLMASTER Voice Terminal. Figure 54-5 shows the layout for a 7407D Integrated Display Terminal. Figure 54-6 shows the layout for a 7406D With Display. All of these voice terminals contain two appearance buttons for convenient answering of incoming calls.



* Terminating and originating capabilities are recommended for the first appearance of an agent's voice terminal.

† Terminating and originating capabilities are recommended for the second appearance of an agent's voice terminal.

‡ Whenever the RELEASE button is assigned to a voice terminal, automatic answering should also be assigned to the terminal.

Figure 54-3. Button Configuration for EUCD Agent (16-Button Set)

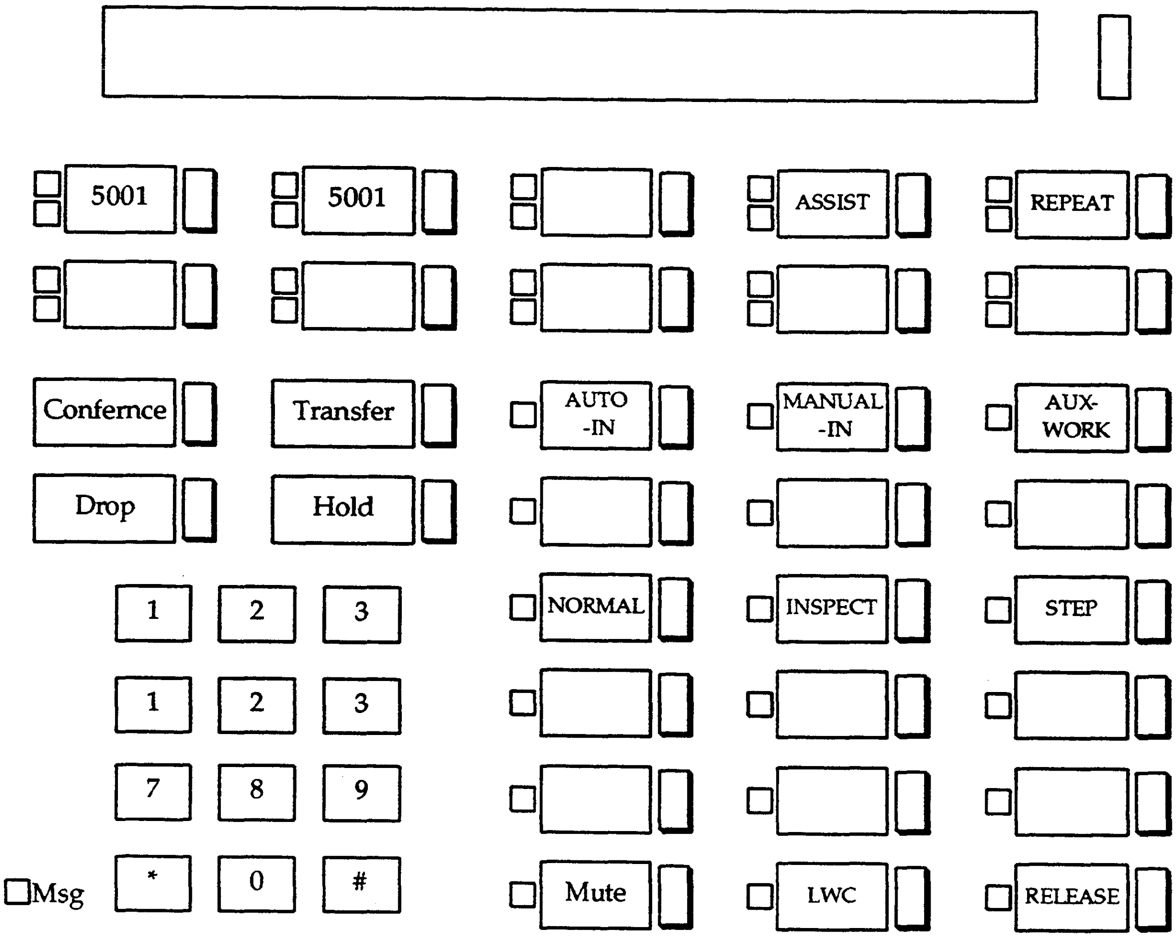


Figure 54-4. Button Configuration for EUCD Agent (CALLMASTER Voice Terminal)

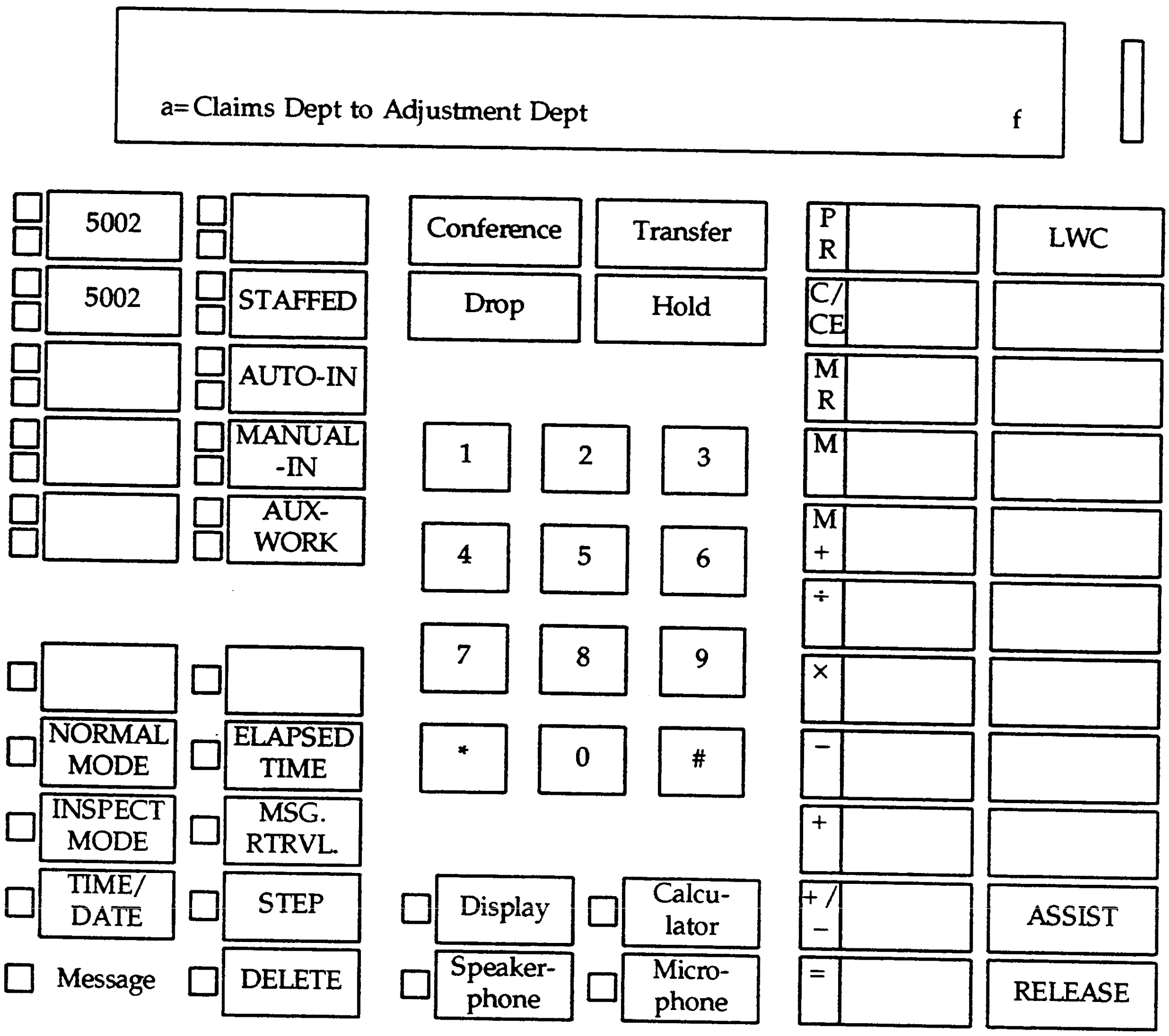


Figure 54-5. Button Configuration for EUCD Agent (7407D Integrated Display Terminal)

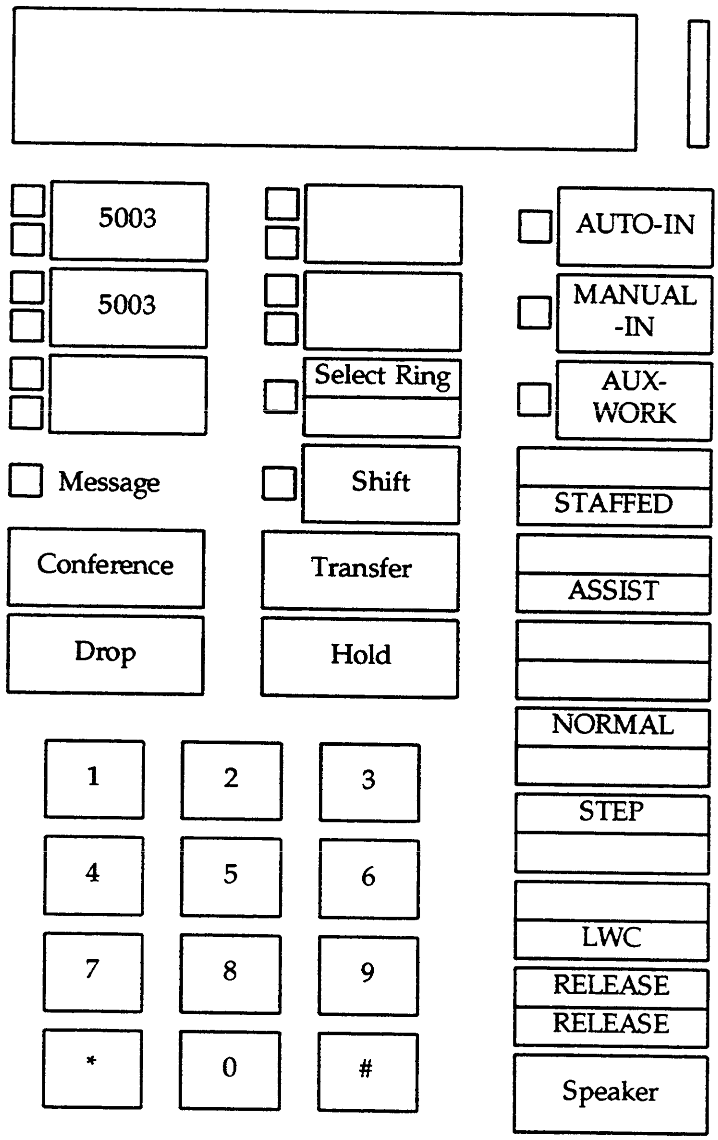
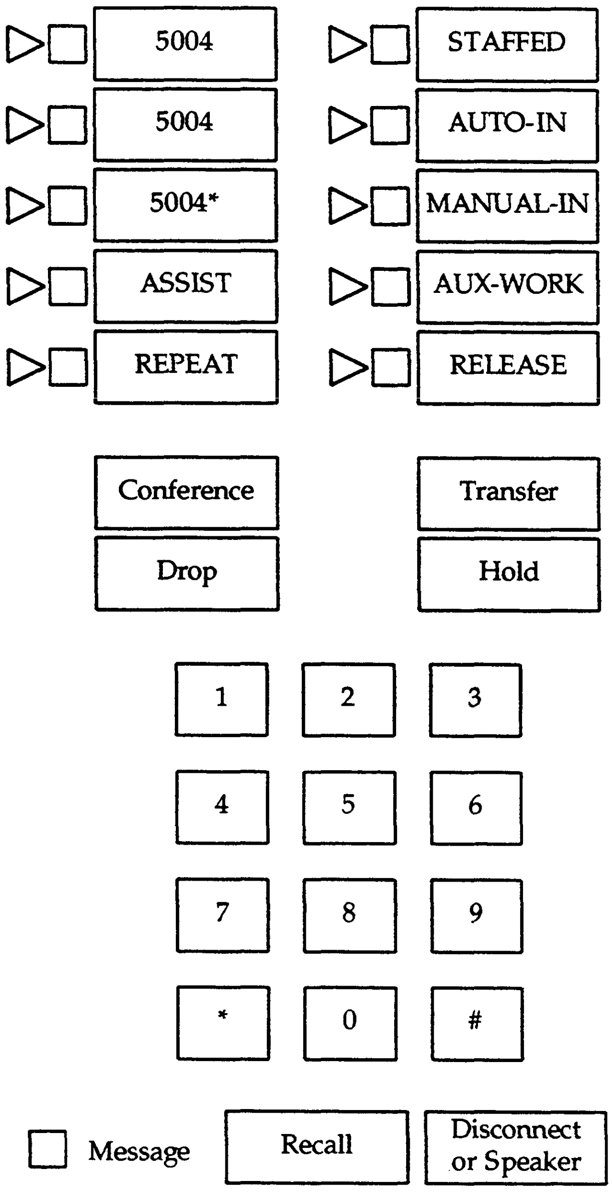


Figure 54-6. Configuration for EUCD Agent (7406D With Display)

Figure 54-7 shows the recommended button configuration for an agent who needs to receive terminal-to-terminal calls. The recommended layout is for a 16-button voice terminal. If additional buttons are desired, a larger voice terminal would be necessary. This voice terminal contains three appearance buttons. The first and second appearances provide basic EUCD functionality, while the third appearance allows direct call completion to the agent's individual extension number.

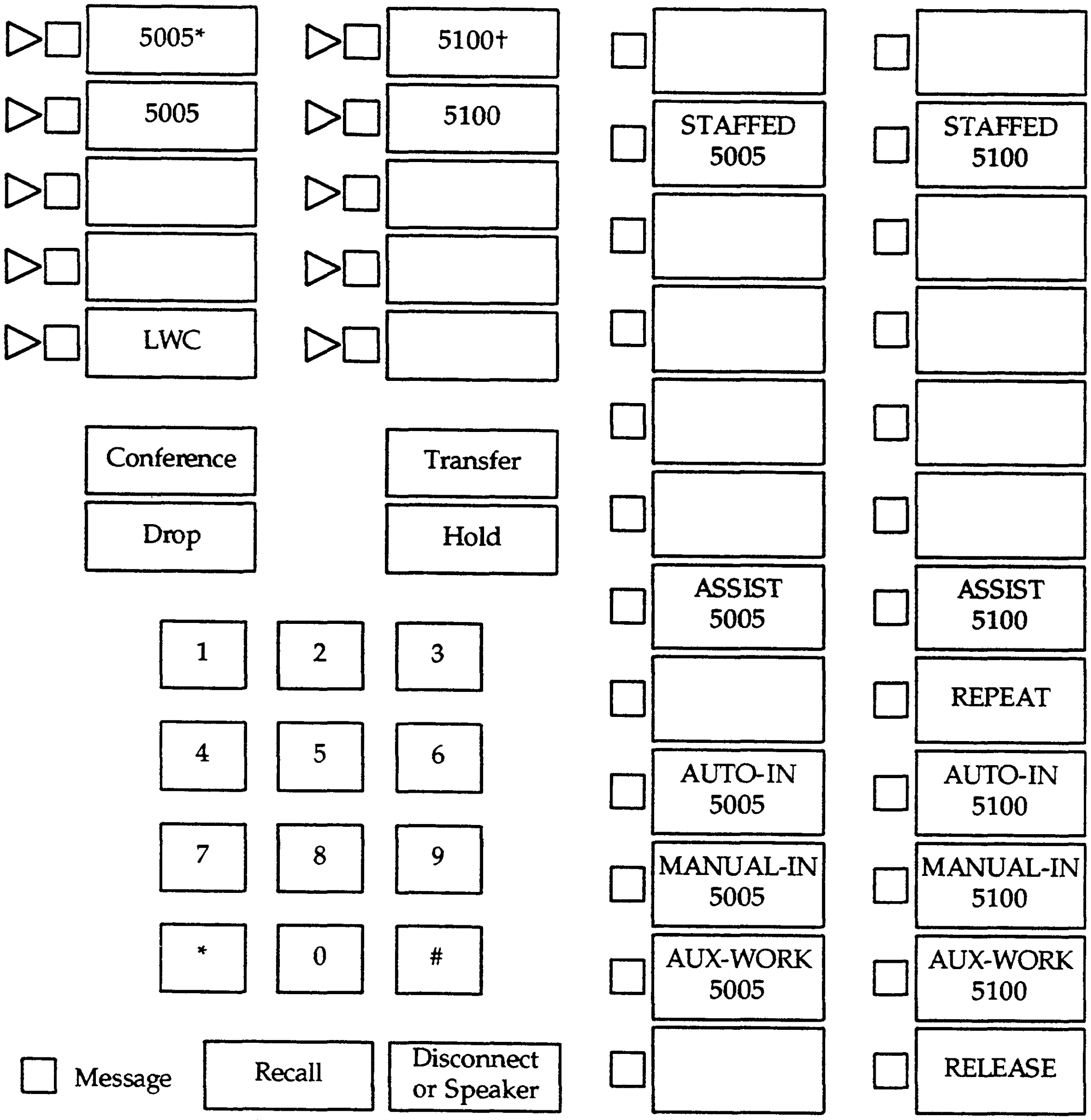


* Originating capabilities (only) are recommended for the third appearance of an agent's voice terminal.

Figure 54-7. Agent Who Also Receives Direct Calls (16-Button Set)

Figure 54-8 shows the recommended button configuration for an agent who handles EUCD calls for two splits. Since many of the buttons are duplicated, a larger set is required for this purpose.

The blank buttons can be assigned as desired by the customer.



* Automatic answering can only be assigned to an appearance of ***one extension number*** on an EUCD agent's voice terminal.

† A System 85 EUCD agent normally works in one split at a time. The two splits operate independently. If an agent is simultaneously staffed in both splits, the switch will deliver EUCD calls from both queues at the same time.

Figure 54-8. Agent Handling EUCD Calls for Two Splits (40-Button Set)

Figure 54-9 shows the recommended button configuration for a split supervisor without agent responsibilities. The recommended layout is for a 16-button voice terminal. This voice terminal contains three appearance buttons, but the third is optional.

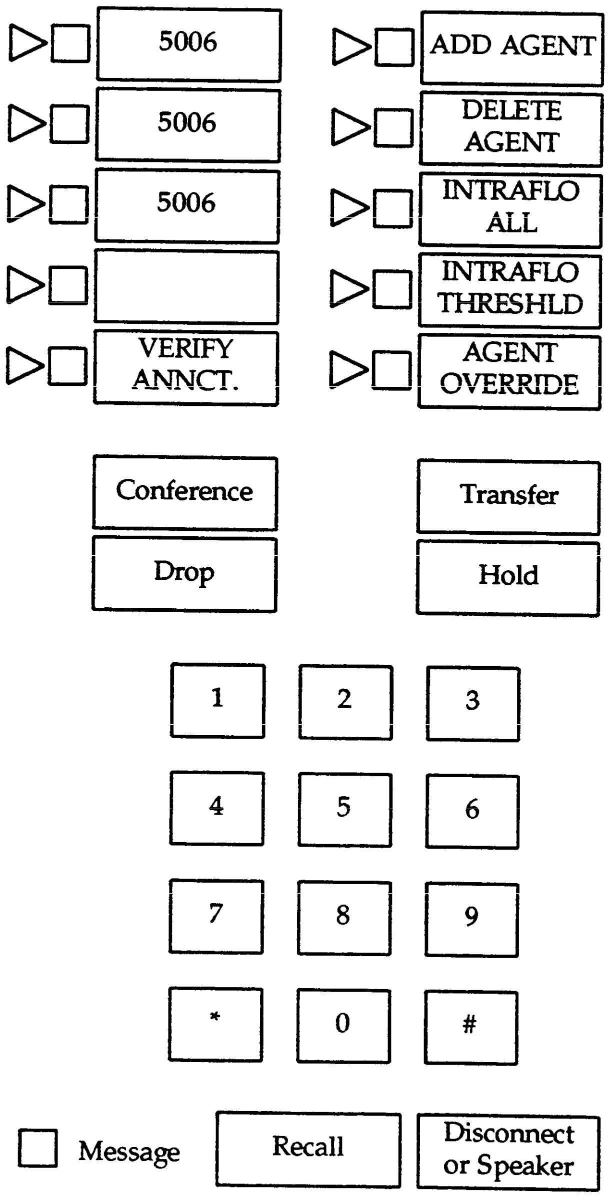


Figure 54-9. Split Supervisor, Without Agent Responsibilities (16-Button Set)

Figures 54-10 and 54-11 show the recommended button configurations for a split supervisor with agent responsibilities. Since these sets contain the buttons needed by an agent and the supervisor, a larger set is required for this purpose.

The blank buttons can be assigned as desired by the customer.

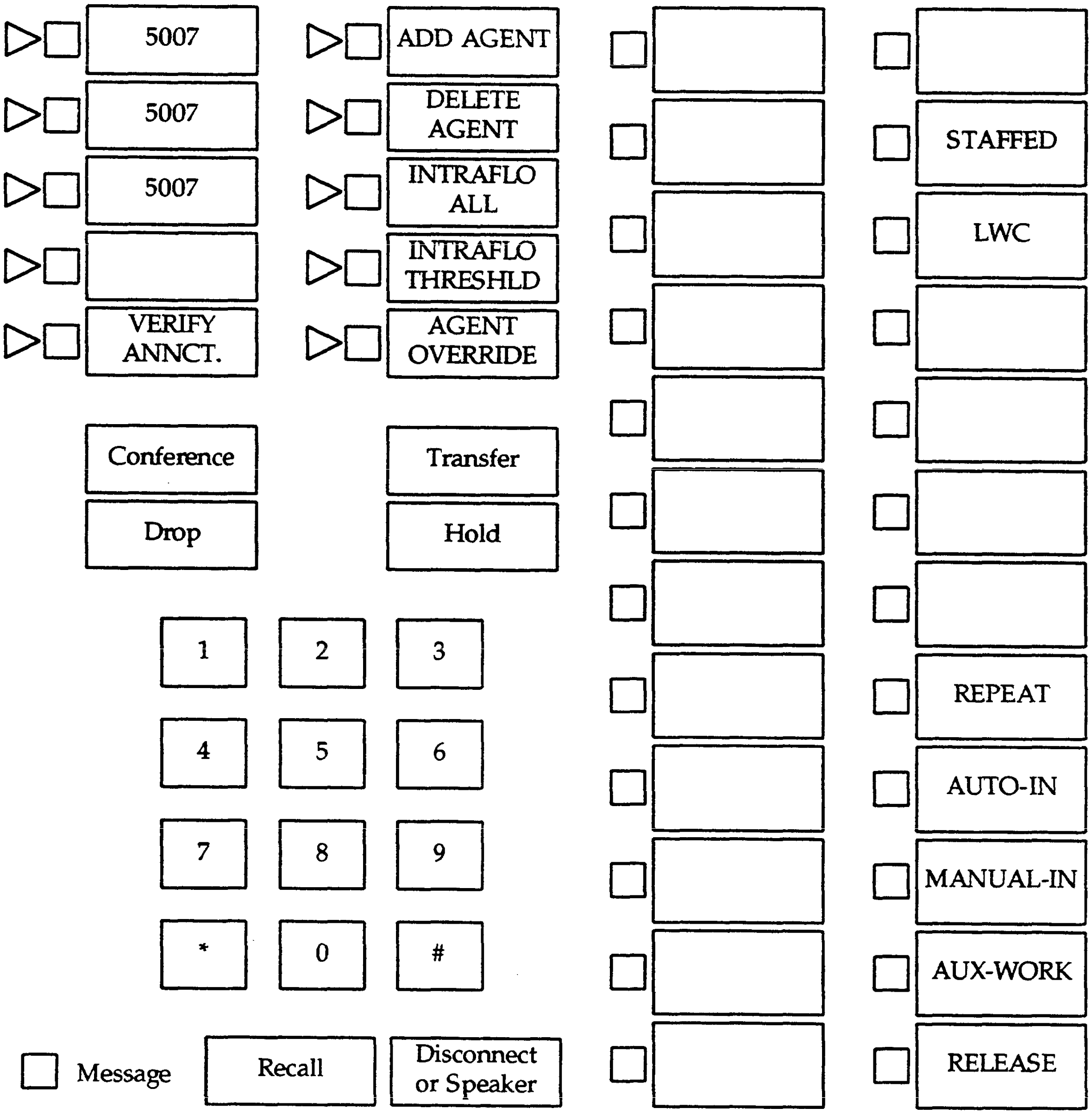


Figure 54-10. Split Supervisor With Agent Responsibilities (40-Button Set)

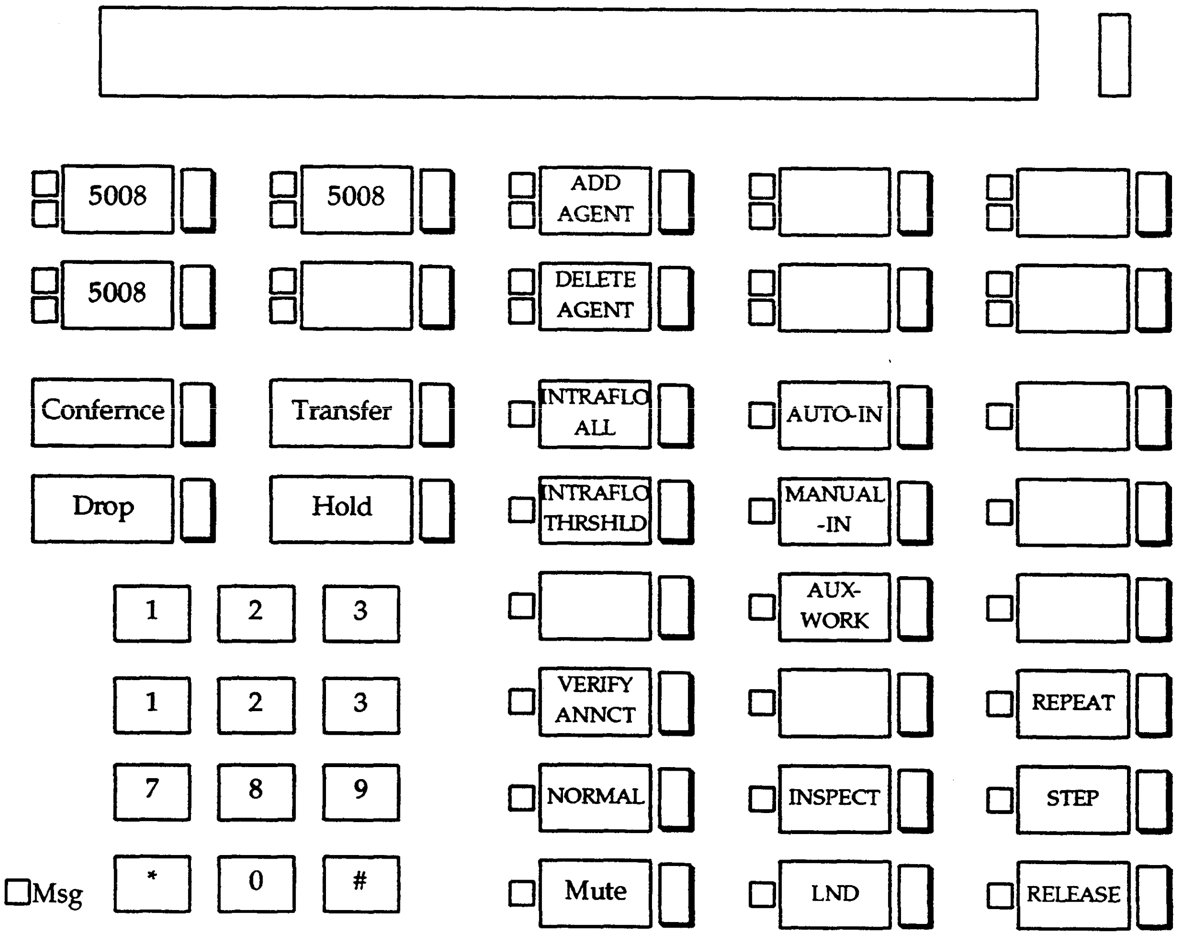


Figure 54-11. Split Supervisor—With Agent Responsibilities (CALLMASTER Voice Terminal)

User Operations

The following are the user operating procedures for this feature.

To Activate Call Forwarding for EUCD Calls

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Call Forwarding—Activate access code (either unconditional

   or

   Press the **[INTRAFLO ALL]** or **[INTRAFLO THRESHLD]** button. [Dial tone]
3. Dial the number of the forwarding destination. [Confirmation tone]

4. Press the **[RELEASE]** button,

or

Go on-hook. (This activation can be performed as many as three times. The order of activation sets the order of priority for forwarding.)

The system supervisor (using the designated attendant console) should:

1. Press an idle loop button.
2. Press the **[START]** button. [Dial tone]
3. Dial the Call Forwarding—Activate access code (either unconditional or overflow). [Dial tone]
4. Dial the split supervisor's extension number (to identify the split). [Dial tone]
5. Dial the number of the forwarding destination. [Confirmation tone]
6. Press the **[RELEASE]** button. (This activation can be performed as many as three times. The order of activation sets the order of priority for forwarding.)

To Establish a Default Destination for Overload Balancing

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Overload Balancing—Default access code. [Dial tone]
3. Dial the AAR/ARS access code. [Dial tone]
4. Dial the default 7-digit number of the private-network destination or the default 7- to 17-digit DDD number of the public-network destination. [Silence]
5. Dial **[#]** [Confirmation tone]
6. Press the **[RELEASE]** button or go on-hook.

To Activate Overload Balancing to the Default Destination

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Overload Balancing—Activate access code (either unconditional or overflow),

   or

   Press the **[INTERFLO ALL]** or **[INTERFLO THRESHLD]** button. [Dial tone]
3. Dial **[#]** [Confirmation tone]
4. Press the **[RELEASE]** button or go on-hook. (This destination is ***automatically inferred*** as the last priority destination.)

To Activate Overload Balancing to a Special Destination

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Overload Balancing—Activate access code (either unconditional or overflow),

   or

   Press the **[INTERFLO ALL]** or **[INTERFLO THRESHLD]** button. [Dial tone]
3. Dial the AAR/ARS access code. [Dial tone]
4. Dial the 7-digit number of the private-network destination or the 7- to 17-digit DDD number of the public-network destination. [Confirmation tone]
5. Press the **[RELEASE]** button or go on-hook. (This destination is automatically inferred as the last priority destination.)

To Add an Agent to a Split

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Member Add access code,

   or

   Press the **[ADD AGENT]** button. [Dial tone]
3. Dial the extension number of the agent to be added. [Confirmation tone]
4. Press the **[RELEASE]** button,

   or

   Go on-hook. (The added agent is in the unstaffed mode.)

To Remove an Agent From a Split

The split supervisor should:

1. Be sure that the agent is in the unstaffed mode.
2. Press an idle appearance button. [Dial tone]
3. Press the **[RELEASE]** button or go on-hook.

To Activate Agent Override With Warning Tone

An observer should:

1. Be sure that the agent is active on a call.
2. Press an idle appearance button. [Dial tone]
3. Dial the agent override (warning tone) access code,

   or

   Press the **[AGENT OVERRIDE]** (with warning tone) button. [Dial tone]
4. Dial the extension number of the agent to be observed.
5. After observation, press the **[RELEASE]** button.

To Activate Agent Override Without Warning Tone

An observer should:

1. Be sum that the agent is active on a call.
2. Press an idle appearance button. [Dial tone]
3. Dial the agent override (no tone) access code,

   or

   Press the **[AGENT OVERRIDE]** (without warning tone) button. [Dial tone]
4. Dial the extension number of the agent to be observed.
5. After observation, press the **[RELEASE]** button.

To Verify a Split's First Recorded Announcement

The split supervisor should:

1. Press an idle appearance button. [Dial tone]
2. Dial the Announcement Verify access code,

   or

   Press the **[VERIFY ANNT]** button.
3. Listen to the announcement.
4. Press the **[RELEASE]** button or go on-hook.

Call to an EUCD Split

An attendant should:

1. Press the **[ANSWER]** button. [Talking connection between the attendant and the calling party]
2. Press the **[START]** button. [The switch returns dial tone, and places the calling party in soft hold.]
3. Dial an associated extension number for the desired split. [Ringback tone]
4. Press the **[RELEASE]** button ***within 4 seconds.*** [The switch places the calling party in the split's queue.]

NOTE: If the attendant does not release within 4 seconds, the call is treated as a direct attendant call. Attendant calls to an EUCD split do not enter the split's queue. Instead, System 85 scans the split for an available agent. If an available agent is not found, there are two possible switch responses. First, if Attendant Call Waiting is assigned to the switch, the call waits on the split supervisors voice terminal. Second, without Attendant Call Waiting, the switch returns busy tone to the attendant.

To Extend a CAS Call to an EUCD Split at a Branch Location

A CAS attendant should:

1. Press the **[ANSWER]** button. [Talking connection between the attendant and the calling party]
2. Press the **[START]** button. [The switch returns dial tone and places the calling party on hold.]
3. Dial an associated extension number for the desired split at the branch. [Ringback tone]
4. Press the **[RLT RELEASE]** button ***within 4 seconds.*** [The switch places the calling party in the split's queue.]

NOTE: If the CAS attendant does not release within 4 seconds, the call is treated as a direct attendant call. Attendant calls to an EUCD split do not enter the split's queue. Instead, System 85 scans the split for an available agent, and if found, completes the call to that agent. If an available agent is not found, there are two possible switch responses. First, if Attendant Call Waiting is assigned to the branch location, the call waits on the split supervisor's voice terminal. Second, without Attendant Call Waiting, the switch returns busy tone to the CAS attendant.

To Turn Off the Reload Warning Lamp

The system supervisor (using the designated attendant console) should:

1. Press an idle loop button.
2. Press the **[START]** button. [Dial tone]
3. Dial the Reload Warning Lamp access code. [The switch returns confirmation tone, and the lamp turns off.]
4. Press the **[RELEASE]** button.

A split supervisor should:

1. Go off-hook on an idle appearance. [Dial tone]
2. Dial the Reload Warning Lamp access code. [The switch returns confirmation tone, and the lamp turns off.]
3. Press the **[RELEASE]** button or go on-hook.

Considerations

Switch Capacities

The switch is capable of handling 512 answering positions arranged in as many as 30 splits. The splits' sizes are always built in blocks of 16 positions. Therefore, a split may contain 16, 32, 48, etc., potential answering positions. An acceptable configuration using all 512 positions spread across 16 splits would be 4 splits of 64 positions, 4 splits of 32 positions, and 8 splits of 16 positions. A processor occupancy evaluation can be performed to determine the impact of the desired configuration.

Message Center service and the AUDIX feature use EUCD splits. The use of EUCD splits by AUDIX and Message Center decreases the number of EUCD split and agent positions available for traditional agent operations (such as, order taking).

An EUCD split serves as the gateway to each AUDIX system. An AUDIX adjunct can only have as many as 32 voice ports. Therefore, these AUDIX gateway splits should only be configured to contain 16 or 32 (usually, 32) members.

As many as 30 queue status lamps (that is, four 30A8 SSI panels) can be provided with the switch to display the status of EUCD queues.

Each 106B display unit can monitor the status of up to 20 agents. As many 106Bs as needed can be provided. For example, if a switch contained 512 agents (the maximum for EUCD on System 85), 26 display units could be provided to monitor the calling activity of every agent.

NOTE: An agent can only be monitored on one 106B display unit at a time.

Any number of queued callers can listen to the same delay recorded announcement at the same time.

Legal Considerations

Laws and union contractual agreements governing the use of agent override differ in different locations, and are subject to change. It is the responsibility of the customer's switch administrator to understand and comply with the applicable regulations.

Delay Recorded Announcements

The System 85 only provides a delay announcement for incoming EUCD calls when there is at least one staffed agent in the associated EUCD split. The calling party continues to hear ringback. Otherwise, the delay announcement would encourage calling parties to wait when no agents are available to answer their calls.

Intraflow and Interflow

Operation of Intraflow and Interflow The call forwarding function of the EUCD feature (intraflow) is used to redirect EUCD calls to local destinations. The overload balancing function (interflow) is used to redirect EUCD calls to distant network nodes. Any attempt to use the Call Forwarding feature to redirect EUCD calls to remote destinations is denied.

There can be three local destinations for redirection of EUCD calls. These destinations are arranged in a priority scheme. If the first destination is unavailable, the switch checks the second and third destinations. Assign the destinations by repeatedly activating either the Call Forwarding—Follow Me (unconditional forwarding) or the Call Forwarding—Busy and Don't Answer (overflow forwarding) feature. Cancel the destinations in reverse order by repeatedly deactivating Call Forwardng.

Additionally, there can be one internode destination for redirection of EUCD calls. As mentioned, this destination is always provided via overload balancing. The internode destination is always automatically implemented as the last priority. The internode destination can be used alone or at the end of the list of local destinations. As such, there can be as many as four possible destinations.

Internode Forwarding Not Allowed: The call forwarding function of the EUCD feature can only be used to redirect EUCD calls to local destinations. The overload balancing function is used to redirect EUCD calls to distant network nodes. Any attempt to use the Call Forwarding feature to redirect EUCD calls to remote destinations is denied.

Power Interruptions

After a power interruption, an automatic reloading occurs of switch translations from the tape into memory. The system reload warning lamp lights. To turn off the reload warning lamp, the system supervisor should press an idle loop button, press the START button, dial the Reload Warning Lamp access code, and press the RELEASE button.

After a tape reload, all split members are unavailable for split calls. Agents with plugged in headsets are placed in the AUX-WORK mode. Agents without plugged in headsets are placed in the UNSTAFFED mode and must press the STAFFED button to return to the AUX-WORK mode. From the AUX-WORK mode, an agent may press the AUTO-IN or MANUAL-IN button to receive split calls.

Agent Override

Agent override allows a voice terminal user to enter an EUCD agent's calls. However, the agent must have an established call in progress.

To observe successive calls using agent override, the observer must reenter the connection after each call is established.

While an observer (using agent override) is connected to an agent's active call, features such as Conference—Three Party, Transfer, Call Waiting, and Hold are denied for use by the agent.

System 85 does not allow agent override to be activated toward a split supervisor's voice terminal.

DNIS (Dialed Number Identification Service)

The DNIS function attaches names to EUCD calls that use ***dial-repeating*** type routing to math an EUCD split. The switch receives the dialed ***digits*** and routes the calls to dummy ***extensions.*** In turn, these dummy extensions redirect (forward or cover) ***every*** call to the split's queue.

For DNIS, dummy splits ***should not*** be used instead of dummy extensions. Dummy splits are not needed for DNIS. When automatic routing is used to direct calls to an EUCD split, these trunk groups can be directly assigned to the "real" split and given ***unique*** names. In fact, using a dummy split causes an unexpected problem. Since calls are redirected from the dummy split to the real split, the real split cannot use the Intraflow—Threshold function. (Every call arriving to the real split has already redirected once.)

CALLMASTER Voice Terminals and EUCD

This is the recommended voice terminal for EUCD agents. This digital multiappearance voice terminal was primarily designed for use in the ACD environment, but is also useful for EUCD agents. The special attributes of this terminal for EUCD agents include:

- Built-in alphanumeric display
- Two direct Starset-headset connectors (no adapters needed)
- Raised feature buttons for improved tactile response
- Horizontal button layout for easier button access
- Status lamps for every button give complete visual feedback

- Fixed mute button
- Moderate price
- Lower-cost migration to ACD.

White Noise With CALLMASTER Voice Terminal

Connected to R2 V2 switches, the CALLMASTER voice terminal, with a directly connected headset, produces white noise at a low volume. This noise is the most noticeable when the EUCD queue is empty and an agent is waiting for another call.

Single-Appearance Terminals and EUCD

Although EUCD agents are allowed to use single-appearance voice terminals, there are some persuasive reasons why they shouldn't. The following is a summary of the shortcomings confronted when single-appearance voice terminals are used in the EUCD environment

- No status lamps to indicate the current agent state
- No method to change the current agent state (without disrupting the active call)
- No convenient way to use automatic answering to receive calls
- NO RELEASE button to quickly disconnect a finished call
- No ASSIST button to quickly obtiain assistance from the split supervisor
- No display modules or built-in display units (for DNIS, city-of-origin, or queue-of-Origin displays)
- Higher-cost migration to ACD (where multiappearance voice terminals become more useful).

7401 D Terminals and EUCD

Although EUCD agents are allowed to use 7401 D voice terminals, there are some persuasive reasons why they shouldn't. The following is a summary of the shortcomings confronted when 7401D voice terminals are used in the EUCD environment

- No automatic answering for a 7401D voice terminal
- NO status lamps to indicate the current agent state
- Only two administrable line appearances
- Only seven administrable feature buttons
- Two button presses (instead of one) to activate a feature or to change the agent state
- No headset adapter accepted (a Starmate headset, or equivalent, must be used)
- No display modules or built-in display units (for DNIS, city-of-origin, or queue-of-origin displays).

Hard and Soft Processor Swaps

Stable EUCD calls will endure a hard processor swap.

If an EUCD call is being distributed to an idel agent when a hard swap occurs, the call will fail.

EUCD queues are stored in a status portion of switch memory. Therefore, if an EUCD call is queued to a split when a hard swap occurs, the call is never routed to an idle agent. The queue is cleared.

If an observer is using agent override to monitor an agent when a hard swap occurs, this connection will endure a hard swap.

Call Forwarding and Overload Balancing relationships are stored in a translation portion of switch memory. Therefore, if a split supervisor activates intraflow or interflow and then a hard swap occurs, these relationships will endure the hard swap.

The EUCD feature operates normally during a soft processor swap.

TABLE 54-C. EUCD Function Access

Function Name	Application	Dial Access Encode	Feature Button Name	Accessed BY (Note 1)
Call Forwarding—Follow Me	Unconditional local forwarding of split's calls (Intraflow—All).	1	INTRAFLO ALL	1, 2
Call forwarding—BY/DA	Overflow local forwarding of split's calls (Intraflow—Threshold).	2	INTRAFLO THRESHLD	1, 2
Call Forwarding—Cancel	Deactivates either split forwarding condition.	3		1, 2
Hold	Places call in progress on hold.	4	HOLD	2, 3, 4
Auto-In Mode	To receive calls in the automatic mode.	70	AUTO-IN	2,3
Aux-Work Mode	To get out of receive EUCD calls mode.	71	AUX-WORK	2, 3
Manual-In Mode Occupied Mode	Enables agent position to receive a single EUCD call. Indicates to switch that agent's position is occupied.	72 73	MANUAL-IN STAFFED	2, 3 2, 3
Member Add	Adds members to an EUCD split.	74		2, 5
Member Delete	Removes members from an EUCD split.	75		2, 5
See notes at end of table.				

TABLE 54-C. EUCD Function Access (Contd)

Function Name	Application	Dial Access Encode	Feature Button Name	Accessed By (Note 1)
Announcement Verify	Allows verification of split's recorded announcement.	76		2
Agent Override (No Tone)	Allows entry into an agent's call in progress without tone alert.	77		2, 4
Agent Override (Warning Tone)	Allows entry into an agent's call in progress with tone alert.	78		2, 4
Reload Warning Lamp	Turns off reload warning lamp after a tape load on the switch.	79		1, 2
Overload Balancing —All	Unconditional distant forwarding of split's calls.	84		1, 2
Overload Balancing —Overflow	Overflow distant forwarding of split's calls.	85		1, 2
Overload Balancing —Default	Establishes default destination for overload balancing.	86		2
Overload Balancing —Cancel	Deactivates either overload balancing condition.	87		1, 2
Release Call	Releases any type of call in progress.	—	RELEASE	2, 3, -4
Repeat Announcement	Repeats city-of-origin message to agents.	—	REPEAT	2, 3
Split Supervisor Assistance	Allows an agent to request help from the split supervisor.	—	ASSIST (NOTE 2)	3

NOTE 1: 1 = System Supervisor (using the designated attendant console)
2 = Split Supervisor
3 = EUCD Agent
4 = Observer
5 = Switch Administrator (using an SMT).

NOTE 2: The ASSIST button is an Abbreviated Dialing button with the split supervisors individual extension number as the stored number.

NOTE 3: Where an encode is shown in the third column, Abbreviate Dialing buttons (using the access code as the stored number) can always be assigned to multiappearance terminals.

Interactions With Other Features

The following System 85 features affect or are affected by the operation of this feature.

Attendant Call Waiting

Attendant calls to a local EUCD split do not enter the split's queue. The switch scans the split to find an idle terminal to complete the call. If no idle line is found in the split, the call waits on the split supervisors terminal. When an attendant places a call to an EUCD individual terminal and that terminal is busy, the call waits on the busy individual terminal if Attendant Call Waiting is provided.

Attendant Direct Extension Selection With Busy Lamp Field

An attendant can use the appropriate DXS (Direct Extension Selection) buttons to place or extend calls to the associated extension number of an EUCD split. However, since a split's queue is never really "busy", the BLF (Busy Lamp Field) lamps adjacent to these DXS buttons are never lit.

Operation

The Attendant Release Loop Operation feature does not apply to all calls that an attendant extends to an EUCD split. Once an attendant-extended call enters the EUCD queue, this call is not timed, and no reminder will be given to the attendant.

Automatic Callback

Automatic Callback can be activated toward the individual extension number of an EUCD agent. However, when this is done toward an agent in a busy split (for example, calls are waiting in queue), a race condition occurs. The switch simultaneously begins the Automatic Callback sequence and the distribution of the next queued call when the agent goes on-hook. Since the EUCD call-distribution algorithms are considerably faster, the agent will usually (if not always) receive the EUCD call first.

Automatic Callback can also be activated toward an associated extension number of an EUCD split. When this is done, the callback sequence is performed against the split supervisor's voice terminal. Again, if the split is busy and if the split supervisor is answering EUCD calls, the same race condition occurs with the same results.

> **NOTE:** Automatic Callback is usually activated toward an agent to facilitate calling from within the organization and is always more effective when activated toward single-appearance terminals. If EUCD agents (using single-appearance terminals) need to receive intracompany calls, the race condition can be avoided by using the Priority Calling feature. However, the better solution would be to provide agents with multiappearance voice terminals equipped with either two or three appearances. In this way, an extra appearance is available for routine communication with other departments.

Busy Verification of Lines

An attendant can use the Busy Verification of Lines feature to check the busy/idle condition of the terminals in an EUCD split. The agent's terminals can be checked whether or not the terminals are available for split calls. If Busy Verification of Lines is activated toward an EUCD associated extension number, only the split supervisor's line is verified.

Call Coverage

An EUCD split (including a Message Center split or an AUDIX system) can be assigned as the final point in a coverage path. However the split number, rather than an associated extension number, is used to designate the coverage point. Therefore, all coverage calls which are redirected to an EUCD split are placed in the nonpriority queue.

On an attendant-extended call to a principal with coverage assigned to an EUCD split (including a Message Center split), the attendant receives cover redirect feedback tone before the call is redirected to the EUCD split. During the 4-second period of silence following the tone (the caller response interval), the attendant should release the call to allow the call to be queued. Also, the attendant can release the call prior to the tone.

CDR (Call Detail Recording)

The CDR feature records the trunk-group dial access code as the calling number on an incoming EUCD call. The extension number of the answering agent, rather than the split's published number, is recorded as the dialed number.

Call Forwarding—Busy and Don't Answer

The Call Forwarding—Busy and Don't Answer feature doesn't forward calls to an associated extension number of an EUCD (including AUDIX) split. When this is attempted, the switch returns intercept tone.

When the split supervisor activates this feature, calls are forwarded to a local destination when an overflow condition occurs The Don't Answer portion of this feature does not apply.

The split supervisor cannot use this feature to forward calls for the supervsor's individual extension number. Activation of this feature only forwards calls which are directed to the split's queue.

There can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destinations were an attendant or an EUCD split (without the inflow level specified for the split), the remaining priority destinations would not be checked. Instead, a forwarded call would unconditionally enter the attendant's or the split's queue.

The forwarding destinations are assigned by repeated activation of the Call Forwarding—Busy and Don't Answer feature. Priority is determined by the order of activation. The destinations are canceled in reverse order by repeated deactivation of the feature.

When an EUCD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split.

Call Forwarding— Don't Answer

An EUCD split supervisor cannot activate Call Forwarding—Don't Answer to forward the supervisor's calls. When an EUCD split supervisor activates Call Forwarding—Don't Answer, the split's calls are forwarded to a local destination in an overflow condition (as if Call Forwarding—Busy and Don't Answer were instead assigned to the supervisors class of service).

For EUCD calls, there can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destination is an attendant or an EUCD split (without the inflow level specified for the split), the remaining priority destinations would not be checked. Instead, a forwarded call would unconditionally enter the attendant's or the split's queue.

When an EUCD call is forwarded to another local split, the caller bean the first delay recorded announcement correponding to the initially dialed split.

Call Forwarding—Follow Me

The Call Forwarding—Follow Me feature can be used to forward all calls to an associated extension number of an EUCD (including AUDIX) split. When this done, forwarded calls enter the split's queue.

The split supervisor cannot use this feature to forward calls to the supervises individual extension number. Activation of this feature only forwards calls which are directed to the split's queue.

There can be three forwarding destinations arranged in a priority scheme. If the first priority destination is unavailable, the second and third destinations are checked. However, if the first or second priority destination is an attendant or an EUCD split, the remaining priority destinations are never checked. Instead, a forwarded call would unconditionally enter the attendant's or split's queue.

The destinations are assigned by repeated activation of the Call Forwarding—Follow Me feature. Priority is determined by the order of activation. The destinations are canceled in reverse order by repeated deactivation of the feature.

When an EUCD call is forwarded to another local split, the caller hears the first delay recorded announcement corresponding to the initially dialed split

Centralized Attendant Service

A CAS (Centralized Attendant Service) attendant is allowed to extend a CAS call to an EUCD split at a branch location. When this is done, the CAS attendant should release the extended call by pressing the RLT RELEASE button within four seconds. This operation allows the extended call to enter the split's queue.

Conference—Three Party

When an EUCD agent adds another agent to an EUCD call, the resulting conference is not considered a work-related activity for the second agent. The 106B status indicator shows the second agent as engaged in non-EUCD activity.

While an observer (using agent override) is connected to an EUCD agent's call, the Conference—Three Party feature is denied for use by the agent.

DCS (Distributed Communications System)

In a DCS environment, direct attendant-calls and attendant-extended calls to an EUCD split in another node are queued. However, for attendant-extended calls, the attendant-does not receive confirmation tone to indicate that the queue has been entered.

ETN (Electronic Tandem Network)

When Overload Balancing is used to interflow EUCD calls to an ACD/EUCD split within a different ETN node and these calls reach the receiving switch over an ETN trunk group, the receiving switch applies distinctive 3-burst zip tone to these calls as they are distributed to an idle ACD/EUCD agent.

FRL (Facilities Restriction Level)

When interflowed EUCD calls use AAR or ARS feature to route these calls outside the local switch, the FRL of the split's supervisor is used to determine whether these calls can use the available network facilities.

Hunting

Any individual extension number of an EUCD agent may be included in a Hunting sequence. Hunting functions normally in this situation. However, an EUCD split associated extension number cannot be assigned to a Hunting sequence. When this is attempted, an administration error will occur.

Message Center Service

One message center split can use intraflow to divert excess calls to another Message Center split. However, when this is done, it is recommended that both splits be assigned to the same AP. If the MCS splits are assigned to different APs, the diverted calls will still enter the backup split's queue, and these calls will be distributed to available MCS agents. But, the Coverage Screen will not be displayed for the answering agent in the backup split.

Override

When Override is activated toward the individual extension number of a busy EUCD agent, the override call intrudes into the agent's active call. When Overnde is activated toward the individual extension number of an idle EUCD agent, the override call alerts the agent with 3-burst ringing.

When Override is activated toward an associated extension number of an EUCD split and the split supervisor is busy, the call intrudes into the split supervisor's active call. When the split supervisor is idle, the call alerts the supervisor with 3-burst ringing. (The Override call does not enter the split's queue.)

> **NOTE:** If the called agent (or split supervisor) is using a multiappearance terminal, an override call (in preference to intruding into the active call) will terminate to an idle appearance (if available) with 3-burst ringing. When no idle appearances are available, the override call will intrude into the active call.

Priority Calling

When Priority Calling is activated toward a single-appearance terminal in an EUCD split, the call waits on that terminal. When Priority Calling is activated toward an associated extension number of an EUCD split, the call always waits on the controlling terminal. (The priority call does not enter the split's queue.)

> **NOTE:** If the called agent (or split supervisor) is using a multiappearance terminal, a priority call will terminate to an idle appearance (if available) with 3-burst ringing. If no idle appearances are available, the switch returns busy tone to the calling party.

Timed Reminder

The Timed Reminder feature does not apply to calls that an attendant extends to an EUCD split. Once an attendant-extended call enters the EUCD queue, this call is not timed and no reminder will be given to the attendant.

Transfer

When an EUCD agent transfers an EUCD call to another agent, the transferred call is not considered a work-related activity for the second agent. The 106B status indicator shows the second agent as engaged in non-EUCD activity.

While an observer (using agent override) is connected to an EUCD agent's call, the Transfer feature is denied for use by the agent.

Restricting Feature Use

The MAAP and SMT can apply Termination Restriction to the individual extension number of an EUCD agent. This has the effect of preventing direct calls from terminating at the agent's terminal. However, calls to the EUCD split terminate normally at the agent's position.

An attendant can apply Controlled Termination Restriction to the individual extension number of an EUCD agent. This has the effect of preventing direct calls from terminating at the agent's terminal. However, calls to the EUCD split terminate normally at the agent's position.

Hardware Requirements

The EUCD feature requires the following additional or special hardware.

To provide the recorded announcements:

- 13A announcement system(s) (eight channels per 13A), A 36A voice coupler, with a 2012D power transformer, for each 13A announcement trunk,

  or

- KS-65270, L12 digital announcer(s) (single-channel announcement set) to provide the recorded announcements. One line circuit of an SN228B or SN229 (eight circuits per pack) must be provided for each KS-65270 to support remote announcement recording.

  or

- KS-65272 4-channel digital announcer to provide recorded announcements. One line circuit of an SN228B, SN229, or TN742 (eight circuits per pack), or one circuit of a TN746 (16 circuits per pack) must be provided for each KS-65272 to support remote announcement recording.
- Space in an auxiliary cabinet to house the announcement set(s)
- An auxiliary trunk circuit of an SN231 circuit pack for each announcement trunk (four circuits per SN231)

To provide the queue warning lamp control option:

- 30A8 system status indicator panel to display queue warning status for eight EUCD splits
- SN241 contact interface (eight circuits per circuit pack).

To provide lamp monitoring of EUCD agents:

- 106B display unit to display the status of 20 EUCD agents
- Two circuits of an SN224 circuit pack to control each display SN224).

To provide convenience for agents, supervisors, and observers:

Headsets

— 3122 Starset II PLANTRONICS* headsets for hands-free operation of the agents

NOTE: Several agent models are available.

* Trademark of Plantronics, Inc.

— 3122 Starset Supra (supervisory model) headsets for convenient operation of split supervisors and observers

NOTE: Several supervisory models are available.

NOTE: The Starset headsets are not compatible with the 7047D voice terminal. (However, the Starmate headsets can be used with this voice terminal.)

— 31712 headset adapters to connect headsets to voice terminals.

NOTE: The headset adapters are not needed to connect the Starset headsets to the CALLMASTER voice terminal.

To provide display capabilities for agents:

- CALLMASTER, 7406D With Display, 7407D, or 7405D (with D401A display module) voice terminals.

Feature Administration

Assignment of the EUCD feature is on a per-system, per-trunk group, and a per-extension class of service basis. The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature. The EUCD feature can also be administered using the Manager IV.

The following are the applicable MAAP and SMT procedures.

MAAP AND SMT PROCEDURES ENCHANCED UNIFORM CALL DISTRIBUTION			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to the agents' extension numbers	Yes
001	1	Assigns an associated extension number to a split supervisor's individual extension number.	Yes
010	1	Assigns EUCD membership to an agent's class of service. Also, use this procedure to assign agent override to an observer's class of service.	Yes
026	1	Administers the split size, the queue warning and overflow levels, the ICI message number, the queuing trunk group, the lamp control circuit for the queue warning indicators, type of hunting to an EUCD split, and the type of EUCD split. Each split is identified by a split number (a number between 1 and 30).	Yes
026	2	Assigns members to a split in the order in which hunting is to take place. The first entry is recognized as the split supervisor.	Yes
026	3	Assigns the EUCD system supervisor.	Yes
027	1	Assigns the first delay recorded announcement and the timing intervals preceding the first and second delay recorded announcements to an EUCD split.	Yes
027	2	Assigns the second delay recorded announcement to the switch.	Yes
027	3	Assigns the queue-of-origin and city-of-origin announcements to an EUCD split.	Yes
052	1	Assigns automatic answering (line type) to a multiappearance terminal.	Yes
054	1	Administers the feature buttons and the RELEASE button for the automatic answering mode. Refer to Table 54-C for a cross-reference of applicable feature buttons and dial access encodes.	Yes
060	1	Assigns a split member to a position on the 106B display.	Yes
060	2	Associates the halves of the 106B display.	Yes

MAAP AND SMT PROCEDURES ENCHANCED UNIFORM CALL DISTRIBUTION (Contd)			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
100	1	Assigns the queuing trunk type, an announcement trunk type, or the contact interface trunk type to a trunk group. The applicable encodes include: 6 Special queue 65 SN241 contact interface 90 EUCD first announcement 91 EUCD second announcement 92 EUCD origin announcement.	No
115	1	Assigns trunk-group termination to an EUCD split and priority queuing to the trunk group.	No
150	1	Assigns the equipment locations of auxiliary trunks and incoming EUCD trunks to their trunk-group numbers.	No
155	1	Administers the contact interface circuit for the queue warning indicators.	No
204	1	Designates the desired alphanumeric display for incoming calls to a split's special queue trunk group that reach an attendant.	No
275	4	Assigns abandon call search and answer supervision for EUCD to the system class of service.	Yes
350	1	Assigns the first digit of the feature dial access codes (if required).	No
350	2	Assigns the EUCD feature dial access codes. Refer to Table 54-C for a cross-reference of the applicable dial access codes and feature buttons.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — ENHANCED UNIFORM CALL DISTRIBUTION	
PATH NAME	**PURPOSE**
terminal-change terminal display unit	Displays or prints the split members being monitored by a 106B display unit
terminal-change class-of-service attributes	Assigns EUCD membership to a voice terminal class of service and assigns agent override to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change group call-distribution attributes	Administers the characteristics of the EUCD split.
terminal-change group call-distribution members	Adds or removes agents to/from an EUCD split. Also, use this screen to assign automatic answering to the agent's extension. This screen is also used to assign the split's priority associated extension number.
terminal-change group call-distribution trunk-groups	Assigns trunk group termination to an EUCD split and priority queuing to the trunk group.
terminal-change system parameters (select the Call-Distribution option)	Declares the attendant-console number of the system supervisor and assigns abandon call search and answer supervision to the system class of service.
terminal-change terminal buttons	Assigns the feature buttons for the EUCD feature. Refer to Table 54-C for a cross-reference of applicable feature buttons and dial access encodes.

Expert Agent Selection

Description

The EAS (Expert Agent Selection) feature enhances Call Vectoring and ACD (Automatic Call Distribution) capabilities by distributing selected ACD calls to subsets of ACD agents who are members of larger splits. These agent subsets are based on the agents' call-handling skills, which could be based on agent training, type of product or service, foreign-language skills, or other expertise.

An incoming call acquires skills as it progresses through vector processing. ACD agents activate skills by dialing an access code plus as many as four skill codes. Both ACD calls and ACD agents may have one or more skills active. A call that has one or more skills active can only be delivered to an available agent if the call and the agent have at least one skill in common.

The ACD feature distributes incoming calls to as many as 60 ACD splits (agent groups). Each split contains one or more agents, who can handle any of the various types of calls that are directed to that split. This arrangement works well for industries that have little variation in the types of calls they receive. There is, however, another type of ACD user who handles calls for more than 60 different produces or services. Some products or services may have a call volume that justifies a dedicated pool of agents, while others do not. Furthermore, products or services for which the call volume is small may require agents with specific skills and training. The EAS feature addresses the needs of this type of ACD user by increasing the number of agent groups and by providing a skill-based agent structure within a split (skill group).

Feature History and Development

This feature was first available with DEFINITY Generic 2.2.

Required Features

The following and DEFINITY Generic 2 features must be assigned to the system.

- Automatic Call Distribution

  Expert Agent Selection can only be used with ACD splits that are assigned the MIA (Most Idle Agent) call distribution (hunting) algorithm.

- Call Vectoring

  Incoming calls acquire skills by way of vector processing.

Skill Groups

EAS increases the number of agent groups by expanding each ACD split into a set of 10 skills called a skill group. That is, when EAS is active, each ACD split becomes a skill group.

Split		Skill Group
1	→	10-19
2	→	20-29
	●	
	●	
	●	
60	→	600-609

The EAS feature effectively increases the maximum number of agent groups from 60 (the maximum number of ACD splits) to 600.

Call Skills and Agent Skills

A skill is a 2-digit or 3-digit number in the range 10-609. An incoming call acquires skills as it progresses through vector processing. This process is represented by the 3 large rectangles above the circle in Figure 55-1. Vectors contain commands that activate a VDN skill preference (1 - 3) or a specific skill (10 - 609). An incoming call can have as many as three skills active.

ACD agents activate call-handling skills by dialing an access code followed by as many as four skill codes. This process is represented by the large rectangle below the circle in Figure 55–1. A skill code is a 1-digit number in the range 1 - 9 and it corresponds to the last (right-most) digit of a skill. For example, to activate skills 21, 22, and 23, an agent (assigned to skill group 20 - 29) dials the Agent Skill Entry access code followed by the skill codes 1, 2, and 3.

The system automatically assigns a default skill to an agent when the agent logs in. The default skill is the skill whose last (right-most) digit is zero (10, 20, 30, etc.). Every skill group has a default skill that is always in effect, whether an agent has activated skill codes . or not. Agents who have only the default skill active can only receive calls that have the default skill active. However, because the default skill is active for every agent, a call that has the default skill active can be delivered to any agent.

The ACD scanning task, represented by the circle in Figure 55-1, scans the incoming call queue and the agent queue associated with each ACD split (assigned the MIA distribution algorithm) trying to match the skills a call has acquired to an agent's call-handling skills.

Both ACD calls and ACD agents may have one or more skills active. A call that has one or more skills active can only be delivered to an available agent if the call and the agent have at least one skill in common. A skill can be unique to each skill group or a common meaning can be defined for a given skill across two or more skill groups.

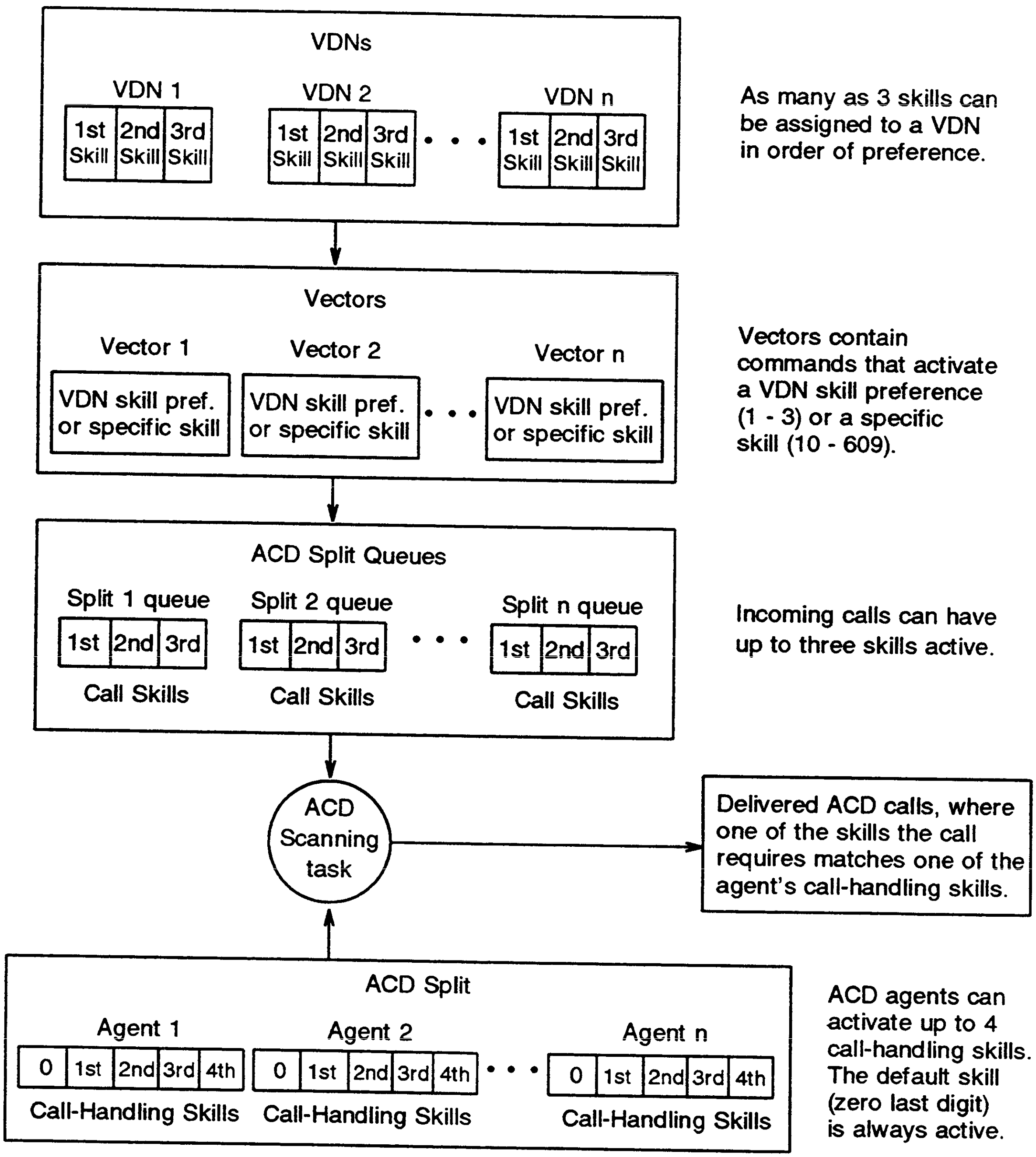


Figure 55-1. Expert Agent Selection Block Diagram

Activating Call Skills

There are two ways to activate call skills, the referenced method and the direct method. With the referenced method, skill preferences are assigned to a VDN and vector commands activate the skills. As many as 3 skills can be assigned to a VDN in order of preference. The first skill assigned to a VDN, called the first VDN skill preference, specifies the skill that is required to handle a call to the VDN. The second and third VDN skill preferences are optional or backup skills. With the direct method, vector commands also activate skills, but skill preferences do not have to be assigned to VDNs.

Referenced Method

Although these two ways of activating call skills are similar, there is an important difference. When the referenced method is used to activate call skills, a vector command can activate different skills for different VDNs. This is possible because a vector command activates a VDN skill preference (1 - 3) and different VDNs can be assigned different skill preferences. For example, if VDN 2000 is assigned 13 as its first skill preference and VDN 3000 is assigned 25 as its first skill preference, a vector command that activates the first VDN skill preference for VDNs 2000 and 3000 will queue calls for VDN 2000 to skill 13 and calls for VDN 3000 to skill 25.

If a vector command references a skill preference that is not assigned to the VDN, vector processing will skip the command. That is, if a VDN has one skill preference administered and the associated vector has a command that references the second or third skill preference, the command will be skipped.

Direct Method

When the direct method is used to activate call skills, vector commands activate the same skills for all calls processed by a particular vector, regardless of the VDN the caller dialed. This is because the vector commands activate a specific skill (10 - 609) rather than referencing a skill preference that could have different values for different VDNs.

The referenced or direct method of activating call skills can be used exclusively or a combination of the two methods can be used. Examples of these call-skill activation methods are presented later in this description.

Vector Commands

The following vector commands can activate or test a VDN skill preference (1 - 3) or a specific skill (10 - 609):

- Queue to main skill,
- Check backup skill,
- Go to step, and
- Go to vector.

The "queue to main skill" command unconditionally queues calls to a skill group at the specified skill and priority. The "check backup skill," "go to step," and "go to vector" commands are conditional and are only invoked if the specified condition is met. The

EAS feature adds conditional testing based on skills. Before the EAS feature was available, a "check backup split" command could be used to queue a call to a backup split if, for example, more than three agents are available. With the EAS feature, a "check backup skill" command can be used to queue calls to a backup skill if, for example, more than three agents with a particular skill are available.

A call can acquire more than one skill if the vector that processes the call has more than one "queue to main skill" command. To accomplish this, the set of "queue to main skill" commands must specify the same priority and skills within the same skill group. For example, to queue calls at low priority and activate skills 11, 12, and 13, three "queue to main skill" commands are required. All three commands specify low priority, but each command specifies a different skill in the same skill group. When vector processing encounters the first "queue to main skill" command, it queues the call at low priority and activates the specified skill. As subsequent "queue to main skill" commands are encountered, the call retains its place in queue and the specified skills are added to the list of active skills. (Before DEFINITY Generic 2.2, the call would be removed from queue and requeued if a second "queue to main split" command were encountered, even if the command specified the same split number.)

As many as three skills can be active at the same time. If three skills are active and vector processing encounters another "queue to main skill" command that specifies the same priority and a different skill in the same skill group, the command is ignored.

Subsequent "queue to main skill" commands that specify a skill in a different skill group remove the call from queue and queue it to the new skill group. Subsequent "queue to main skill" commands that specify the same skill but a different priority remove the call from queue and requeue the call to the same skill at the new priority.

A subsequent "queue to main skill" command that specifies the default skill (the right-most digit is zero) adds the default skill to the list of active skills. Adding the default skill to the list of active skills queues the call to the entire skill group. A call that has the default skill active can be delivered to any available agent in the skill group (because the default skill is active for all agents). If a call is queued to the default skill and a subsequent "queue to main skill" command specifies a nondefault skill, all skills except the nondefault skill are canceled.

Skills are prioritized in the order in which they are activated. The prioritized list of skills takes precedence over the MLA distribution algorithm. For example, if a call has two skills active and an agent with the call's second skill has been available longer than an agent with the call's first skill, the call will be delivered to the agent with the call's first skill, even though the agent with the call's second skill has been available longer.

Call Management System

The Release 3.1 CMS (Call Management System) supports the EAS feature with VDN and vector administration, real-time displays, and historical printed reports. Earlier releases of CMS do not support the EAS feature.

Expert Agent Selection Examples

The following examples show how the EAS feature could be used to distribute calls to agents working for a Tourist Information Service. Example 1 shows how calls to the New England area are handled when only one skill preference is assigned to each VDN. Examples 2 and 3 add a second and third VDN skill preference. Example 4 replaces the third VDN skill preference with a specific skill.

Example 1

As Figure 55–2 illustrates, the company publishes one "900" number for each state in the New England area. Each 900 number terminates to a VDN that is assigned a single, unique skill. All of the VDNs for the New England area are associated with vector 1 which activates each state's unique skill. Skill group 10-19 has 12 agents and each agent is trained to handle calls for a specific state. For example, if a call terminates to VDN 1100 (Maine), it can only be handled by agent 1 or agent 2. If agents 1 and 2 are not available, the call waits in queue until one of these agents becomes available.

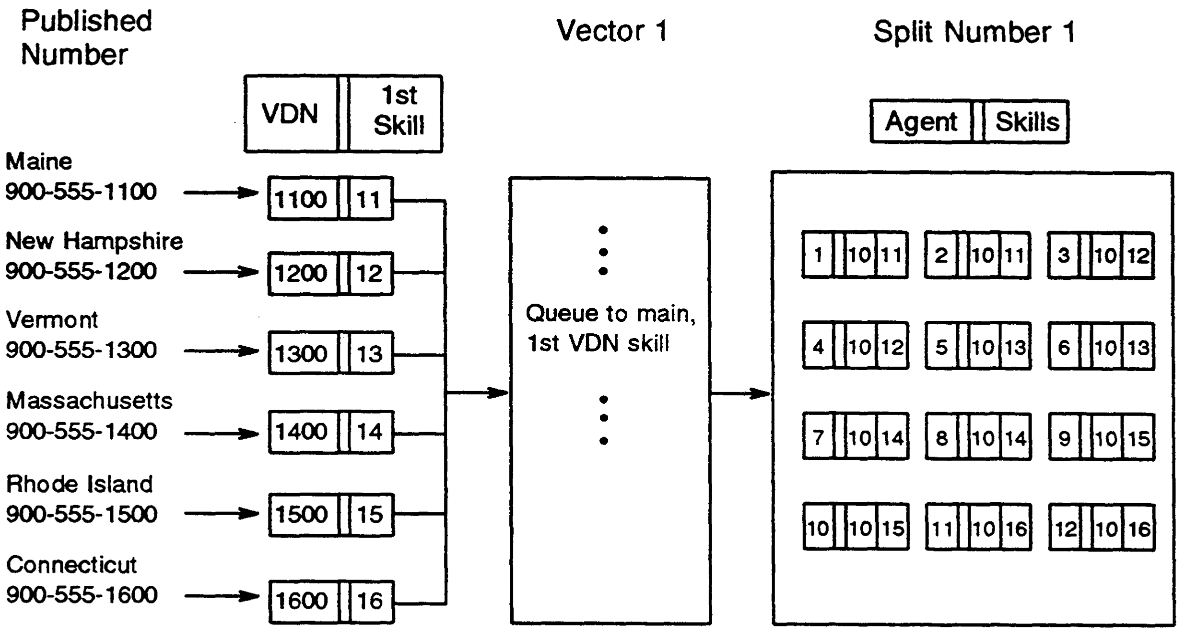


Figure 55-2. Expert Agent Selection - Example 1

Using a similar arrangement for the other 44 states, the Tourist Information Service requires only 9 skill groups to provide 50 information services (one service for each state) with 50 distinct groups of agents.

Example 2

As time passes, agents are trained to handle calls for more than one state. Because agents can activate a maximum of four skills, when one or more agents can handle calls for all six states in a particular area, a new skill must be defined. As shown in Figure 55-3, skill 17 accomplishes this purpose. A second skill preference specifying skill 17 has been added to each VDN and a second "queue to main skill" command that activates the second VDN skill preference has been added to vector 1.

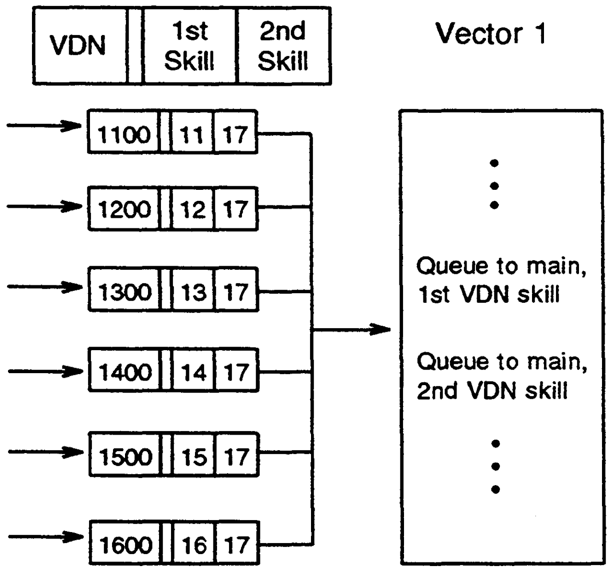


Figure 55-3. Expert Agent Selection - Example 2

With this arrangement, agents who can handle calls for four or fewer states activate the corresponding skills. Agents who can handle calls for any state in the New England area activate skill 17 only. Vector 1 first activates each state's unique skill (the first skill preference assigned to each VDN), then without removing calls from queue, the second VDN skill preference is added. The result is that any call for information about the New England area can be handled by either an available agent trained to handle calls for one or more states or an available agent with skill 17 (all states in the area).

Although it is important to match an agent's call-handling skills to the skills a call has acquired, the longer a call waits in queue the less important this becomes. At some point, answering the call is more important than matching skills, even if the agent is not thoroughly trained to handle the call.

As shown in Figure 55-4, skill 10 (the default skill for skill group 10- 19) accomplishes this purpose. Skill 10 has been added to each VDN as the third skill preference. In addition to another "queue to main skill" command, a "wait" command has been added to vector 1.

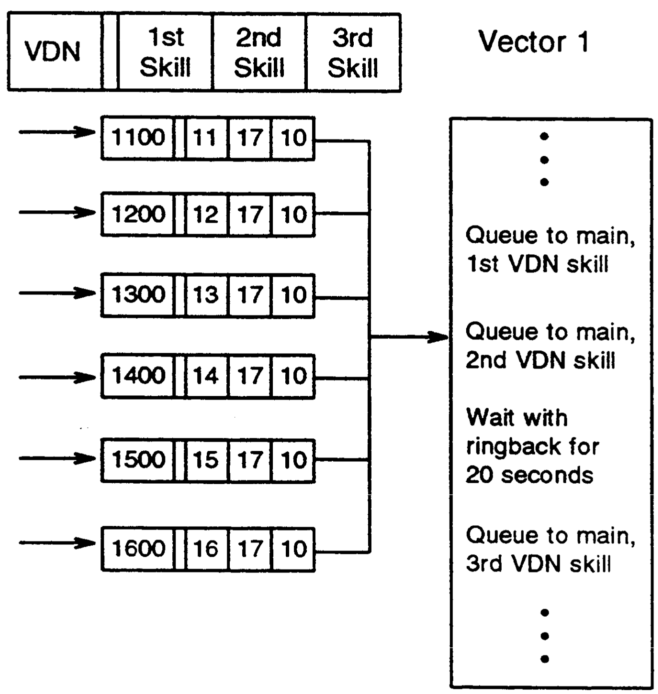


Figure 55-4. Expert Agent Selection - Example 3

As in Example 2, agents who can handle calls for four or fewer states activate the corresponding skills, agents who can handle calls for any state in the New England area activate skill 17 only. The system automatically activates the default skill (10).

Vector 1 first activates each state's unique skill, immediately adds the second VDN skill preference, and then waits 20 seconds. If a call has not been answered after 20 seconds, the third VDN skill preference is added to the list of active skills. The third VDN skill preference activates the default skill, which queues the call to the entire skill group. The result of this arrangement is that the longer a call waits in queue, the larger the pool of agents grows.

Example 4

Figure 55-5 shows an example of a vector that uses the referenced and direct methods of activating call skills. This configuration is similar to Example 3, except the third "queue to main" command activates a specific skill (19) instead of a VDN skill preference.

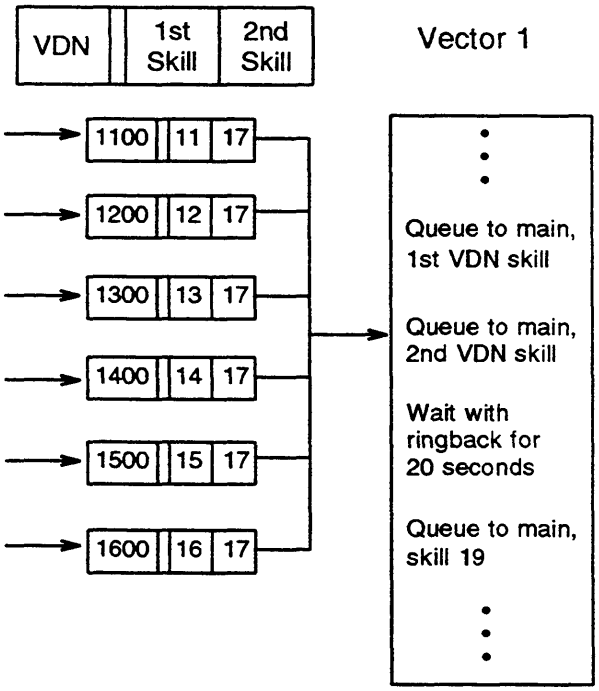


Figure 55-5. Expert Agent Selection - Example 4

Agents who can handle calls for four or fewer states activate the corresponding skills. Agents who can handle calls for any state in the New England area activate skill 17 only. Agents who have some training and can reasonably handle a call for any state in the area activate skill 19.

Vector 1 first activates each state's unique skill, immediately adds the second VDN skill preference, and then waits 20 seconds. If a call has not been answered after 20 seconds, skill 19 is added to the list of active skills. Like Example 3, the result of this arrangement is that the longer a call waits in queue, the larger the pool of agents grows.

Considerations

Call-Handling Skills

To activate call-handling skills, an ACD agent dials the Agent Skill Entry access code followed by as many four skill codes. After dialing the last skill code, an agent can wait for timeout (100 seconds) or dial the end-of-dialing digit (#). If an agent is using a rotary telephone or some other equipment from which a # cannot be dialed, the agent must wait for timeout (ten seconds after entering the last skill code).

The digit 0 and the * are ignored as skill codes. For example, the following entries

3700#

or

37**#

are interpreted as skills 3 and 7. Also, entering the same agent skill more than once is allowed, but the duplicate entries are ignored. For example, the following entries

3344#

or

3434#

are interpreted as skills 3 and 4.

The order in which skill codes are entered does not effect call delivery. In other words, entering skill code 3 first and skill code 7 second does not mean that an agent is more likely to receive more skill-3 calls than skill-7 calls.

Deactivating the EAS Feature

All vector-command references to VDN skill preferences must be removed before the EAS feature can be deactivated.

User Operations

To Activate Call-Handling Skills

To activate call-handling skills, an ACD agent must be either staffed or logged into CMS (Call Management System). Any active call-handling skills are cleared when an agent unstaffs or logs out. Call-handling skills can be changed (by activating different skills) without logging out

1. Go off-hook on an idle appearance. [Dial tone]
2. Dial the Agent Skill Entry access code. [Dial tone]

3. Dial up to four skill codes followed by # (the end-of-dialing digit) or wait 10 seconds for timeout. [Confirmation tone]

To Cancel Call-Handling Skills

1. Go off-hook on an idle appearance. [Dial tone]
2. Dial the Agent Skill Entry access code. [Dial tone]
3. Dial # (the end-of-dialing digit). [Confirmation tone]

   or

   Wait 10 seconds for timeout. [Confirmation tone]

   NOTE: While logged in, agents retain the default skill.

Interactions With Other Features

Abbreviated Dialing

Abbreviated Dialing can be used to dial the Agent Skill Entry access code.

Automatic Call Distribution

The EAS feature can only be used with ACD splits that are assigned the MIA (Most Idle Agent) call distribution (hunting) algorithm. If a call that has skills active is delivered to an ACD split that is assigned director circular hunting, the skills are ignored.

Hardware Requirements

None.

Feature Administration

Assignment of the EAS feature is on a per-system basis.

On DEFINITY Generic 2, this feature is administered using DEFINITY Manager II.

This feature can also be administered using CSM (Centralized System Management).

Before the EAS feature can be administered, the Automatic Call Distribution and Call Vectoring features must be administered. Refer to the appropriate feature descriptions for information about administering these features.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES EXPERT AGENT SELECTION			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
030	2	Beginning with DEFINITY Generic 2.2, Procedure 030, Word 2 is used to add a new vector or to change, copy, or remove an existing vector. (Prior to DEFINITY Generic 2.2, Procedure 030, Word 3 was used to add, change, copy, or remove vectors.)	Yes
030	3	Assigns VDN skill preferences (1 - 3) or specific skills (10 - 609) to vector commands.	Yes
031	2	Assigns up to 3 skill preferences to a VDN.	Yes
276	1	Assigns Expert Agent Selection to the feature group class of service.	No
350	2	Assigns the Agent Skill Entry dial access code. The applicable encode is: 107 Agent Skill Entry	No

Extension Number Portability

Description

Extension Number Portability allows a user in a private network to move from one node to another (from switch to switch) without changing extension numbers. Although not required for DCS (Distributed Communications System), portability is ideal in a DCS as it provides added transparency. Portability can also be used in an ETN (Electronic Tandem Network) or with an AUTOVON (Automatic Voice Network)* gateway arrangement. Not only does the user's extension number remain the same, but the office code number (the first 3-digits of the 7-digit public or private network address) also remains unchanged.

Feature History And Development

This feature was introduced for 5-digit extensions in System 85, Release 2, Version 3. In R2 V4, Issue 1.3 4-digit ENP was added.

Physical Requirements—The Portability Subnetwork

An extension number can be "ported" to any node within a group of switches known as a portability subnetwork. In Figure 56-1, a portability subnetwork is formed under the following constraints:

- Each switch can be a System 85 Release 2, Version 3 or 4, or a Generic 2, and be part of an unrestricted 5-Digit Dialing plan.

  or

  Each switch can be a System 85 Release 2, Version 4, Issue 1.3 or a Generic 2, and be part of an unrestricted 4-Digit Dialing plan.

  System 85 R2 V3 introduced unrestricted 5-digit dialing and System 85 R2 V4, Issue 1.3 introduced unrestricted 4-digit dialing. Unrestricted 4- or 5-digit dialing is essential for true extension number protability. The differences between unrestricted 4- or 5-digit dialing and ***"prefix"*** 5-digit dialing (available on R2 V2 System 85 and earlier, DIMENSION System Feature Package 8 [Issue 3], and System 75 switches) are described later under ***4- and 5-Digit Dialing.***

- The switches must be connected by tie trunks in an ETN arrangement.

  Each switch must be able to route a call to any node in the portability subnetwork. Tie trunks provide connectivity for this type of call routing.

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* Portability does not apply to the AUTOVON as such. It can however, be used within a network of System 85s and Generic 2s that has AUTOVON access through the Precedence Calling feature.

- Each switch must use location code routing (AAR [Automatic Alternate Routing] or WCR [World Class Routing] features).

  This feature provides automatic routing for calls between nodes in a private

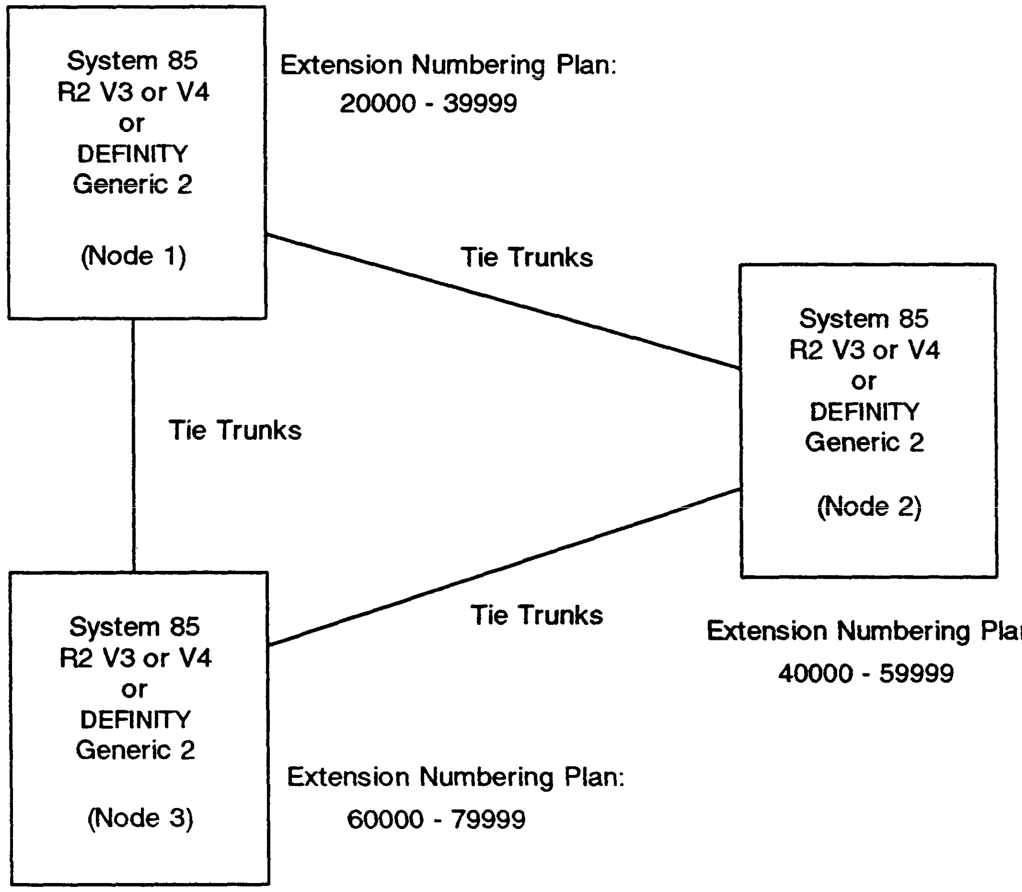

Figure 56-1. Portability Subnetwork Characteristics With 5-Digit Extensions

Compatibility With Other Switches

Any System 85 can be upgraded in order to support a 4- or 5-digit dialing plan and Extension Number Portability. This may require new hardware as well as new software and a new dialing plan.

Extension Number Steering Alternative

There will be cases where it is not practical to upgrade a switch and incorporate a new dialing plan. In the case of a DIMENSION System or a System 75, Extension Number Portability is not provided. For these situations, ***Extension Number Steering*** can be used. Extension Number Steering does not provide the capacity of Extension Number Portability and is harder to implement and administer. For special cases, its use may be justified.

Figure 56–2 shows a 4-digit DIMENSION System and two System 85 R2 V3 switches in an ETN arraingement. A 5-Digit Dialing subnetwork and a portability subnetwork are set up using only System 85s (there is no node assignment for the DIMENSION System). A user can move from Node 1 of the portability subnetwork to the DIMENSION System. To make this move transparent to users at Node 1 and Node 2, the relocated user's extension number on Node 1 is assigned to Extension Number Steering. When Node 1 receives the digits 34500, it sends them out to the DIMENSION System for call termination. System 85, DIMENSION Systems and Generic 2.1 switches can add or delete digits from those received using 7 to 10 digit conversion or subnetwork trunking. Generic 22 switches can perform digit modification in a similar way but with greater capability. At the DIMENSION System switch, the extension number 34500 is changed to a 4-digit extension number. In the example in Figure 56-2, the user is assigned to extension number 4500 so that Node 1 can delete the initial digit 3 and send the remaining 4500.

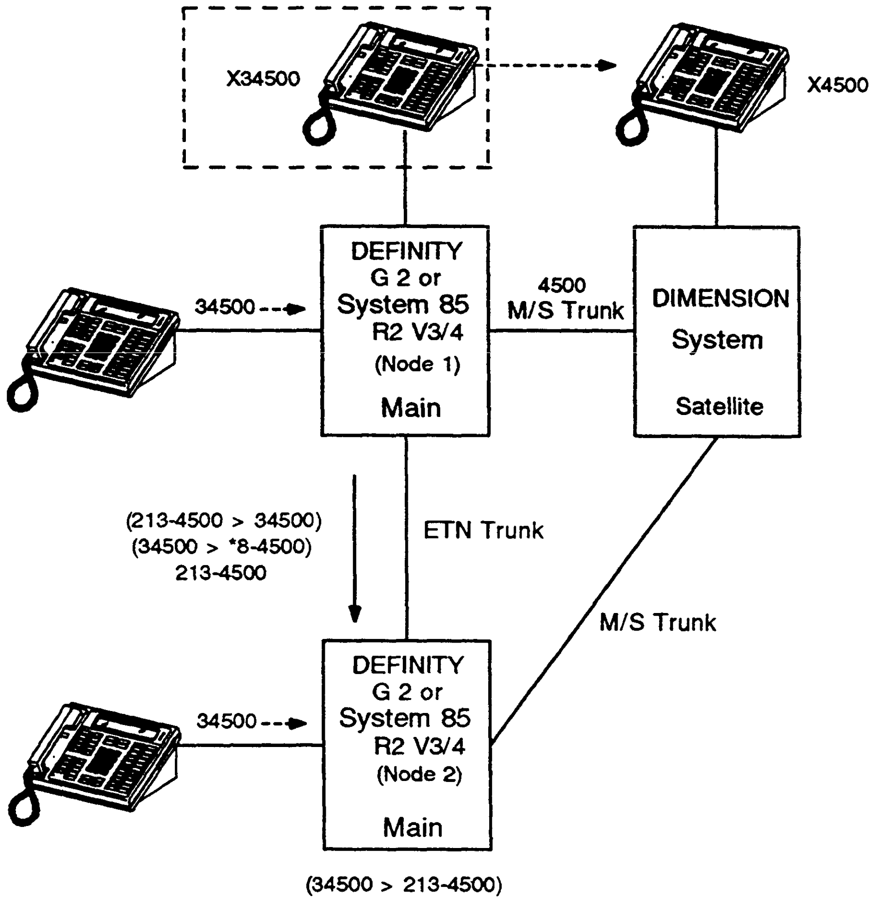

Figure 56-2. Extension Number Steering

The moved user must be made aware of the 4-digit dialing plan on the DIMENSION switch, and users on the DIMENSION switch should be made aware of the moved user's new extension number (4500). If a DIMENSION user calls the old extension number (34500), the call will route to Node 1 and then back to the DIMENSION switch. While this will work, it ties up costly tie-trunks and switch facilities unnecessarily.

4- and 5-Digit Dialing Plans

Prefix 5-Digit Dialing

In R2 V2, System 85 offered a 5-digit extension number dialing plan. This early version of 5-digit dialing prefixed a fifth digit onto a 4-digit extension number. The fifth (leading) digit is called a prefix and is used by the switch to determine call routing. This enables usres to reach a specific extension on any node by dialing a 5-digit extension number. Prefix 5-digit dialing uses ***extension numlwr steering*** (also known as multidigit steering). Any number of digits, up to the full five, can be used to route a call to a specific extension on a specific node.

In larger configurations, it is often necessary to route on the first two digits. It is desirable to keep the number of digits used for extension number steering purposes small (one or two). Using more digits for steering makes the overall numbering plan more complex and difficult to manage and coordinate between switches.

Unrestricted 5-Digit Dialing

Beginning with Release 2, Version 3 of System 85, unrestricted 5-digit dialing, became available. With unrestricted 5-digit dialing the full five digits are used as a true extension number. With portability, any 5-digit extension number can reside on any node in the portability subnetwork. A single node could contain the extension numbers 45678 and 87654. Within this arrangement, extension number steering is not used for portability. When a 5-digit extension number is dialed, it is examined digit by digit to determine the switch in the portability subnetwork where it resides.

This method can be used by any private network that wants a simplified dialing plan (dialing a 5-digit extension number rather than dialing a feature access code [AAR or WCR] plus a 7-digit private-network number). A 5-digit dialing plan does not require the DCS or Extension Number Portability features.

Unrestricted 4-Digit Dialing

Beginning with Release 2, Version 4, Issue 2.0 of System 85, unresticted 4-digit dialing became available. In this form, only four digits are used as a true extension number. With portability, any 4-digit extension number can reside at any node in the portability subnetwork. The same node could contain the extension numbers 4567 and 7654. Again, extension number steering is not used for portability. When a 4-digit extension number is dialed, it is examined digit by digit to determine where in the subnetwork it resides.

This method can also be used by a small private network that wants a simplified dialing plan (dialing a 4-digit extension number rather than dialing a feature access code [AAR or WCR] plus a 7-digit private network number). Unrestricted 4-digit dialing does not require the DCS, or Extension Number Portability features.

Architectural Requirements—The Numbering Plan

A portability subnetwork is designed as a single system. Otherwise, calls from outside the subnetwork would be difficult if not impossible to route properly. Switches outside of the subnetwork have no way of knowing where a particular extension is located within the subnetwork. The subnetwork is treated like a single switch. Because each node in the subnetwork is capable of routing a call to any terminal in the subnetwork, outside calls can be routed to the nearest node, and this node can perform the final routing.

Network Location Code Assignments

In a typical ETN, each switch is assigned a location code called the "home RNX" or "home location code." A call using the home location code is terminated locally. Other location codes are called "remote location codes," and indicate that the call must be extended to another node, which is identified by the location code. With location code routing, a single location code can (hypothetically) handle up to 10,000 extension numbers. Switches with more than 10,000 extensions (5-digit dialing required) are assigned multiple home location codes, one to handle each block of 10,000.

In a portability subnetwork, all nodes share a common set of home location codes. The final destination node for a call cannot be identified by simply reading the location code associated with a call. One call to 320-1234 may go to Node 1 while another call to 320-1230 may go to Node 2. Figure 56-3 shows how this problem is resolved. The switches shown as Nodes 1, 2, and 3 form a portability subnetwork. The other switches are outside the portability subnetwork but are part of the 4- or 5-digit dialing ETN.

Each switch in the ETN shown in Figure 56-3 contains a table with every assigned location code showing whether these are home or remote location codes. The switches outside the portability subnetwork (no node number shown) each have a unique home location code. To these switches, all other location codes are remote. On the switch at the left, a call using location code 420 is routed to a local terminal. Calls using any other location code are routed out to the next switch (Node 2) in the ETN. Since the 420 switch is tied directly to only Node 2 of the ETN, all network calls not for location code 420, are routed to Node 2. This is normal ETN routing. Once a call enters the portability subnetwork, normal ETN routing changes.

Notice that each location code used within the portability subnetwork is assigned as a home location code at each switch with the subnetwork. That is, all switches within the portability subnetwork share the same set of home location codes. This is done to allow any network number (XXX-XXXX) assigned within the portability subnetwork to home on any node in the subnetwork. When a home location code is received, the call is routed to the appropriate node based on the extension number. When a remote location code is received, the call is routed out of the subnetwork to the appropriate ETN node.

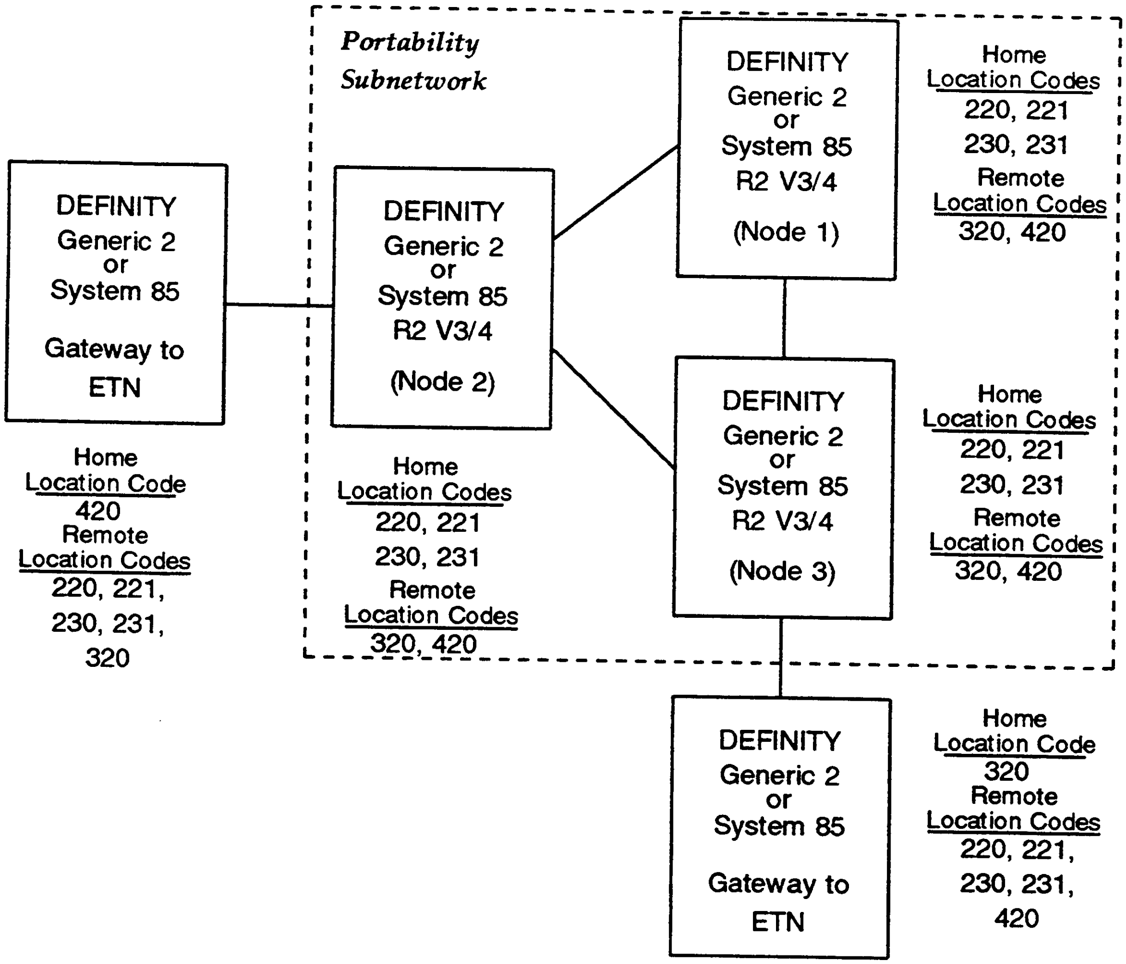

Figure 56-3. Network RNX (Location Code) Assignments

Subnetwork Extension Number Assignments

Within a portability subnetwork, each extension number is separately assigned on each switch. This allows each node to properly route any call to any extension within the subnetwork. Looking back to Figure 56-1, the extension numbering shown at each node is incomplete. The extension numbers shown are the extension numbers initially assigned to terminals at each node. Each node is also assigned all remote subnetwork extension numbers. The remote extension numbers are not associated with local terminals. Rather, they point to the node in the subnetwork where that extension number is located (Procedure 254, Word 2). When a subnetwork extension number is received, the digits indicate one of two types of routing

- The extension number digits point to a local terminal. The call is terminated at the local node.
- The extension number digits point to a node. The digits are converted to a 7-digit private-network number and the call is routed to the appropriate node for completion.

Ported Extension Numbers

When an extension number is ported to a new node following administration must be accomplished.

- At the Gaining Node:

  1. Procedure 354, Word 2 is used to change the ported extension number's node number assignment.
  2. The ported extension number is added to the gaining node using Procedure 000, Word 1.

- At the Losing Node:

  1. The ported extension number is removed using Procedure 000, Word 1.
  2. The node number of the ported extension number is changed to the gaining node using Procedure 354, Word 2. Then,

- At All Remaining Nodes:

  1. Since the same extension number has a node assignment at all remaining nodes, that assignment is changed to the gaining node using Procedure 354, Word 2).

NOTE: When this administration order is followed, the likelihood of circular routing during the transition is minimized.

Portability and Other Network Arrangements

Figure 56-4 shows an example of how a portability subnetwork is formed within an existing network. In this example, the network uses the following System 85/Generic 2 characteristics:

- All switches are part of an ETN. This reduces long distance costs and simplifies dialing between dispersed locations. Network calls can be placed using 7-digit private-network numbers (XXX-XXXX) rather than 10-digit public-network numbers.
- The three switches located in Texas are in a DCS. This reduces dialing requirements between these switches and provides some feature transparency.
- The operations in Dallas and Fort Worth are closely related. Moves to and from these two cities are frequent, so these switches are formed into a portability subnetwork. These switches must be System 85, Release 2, Version 3 or later, or DEFINITY Generic 2. The dialing plan for the DCS must use unrestricted 4- or 5-digit dialing.

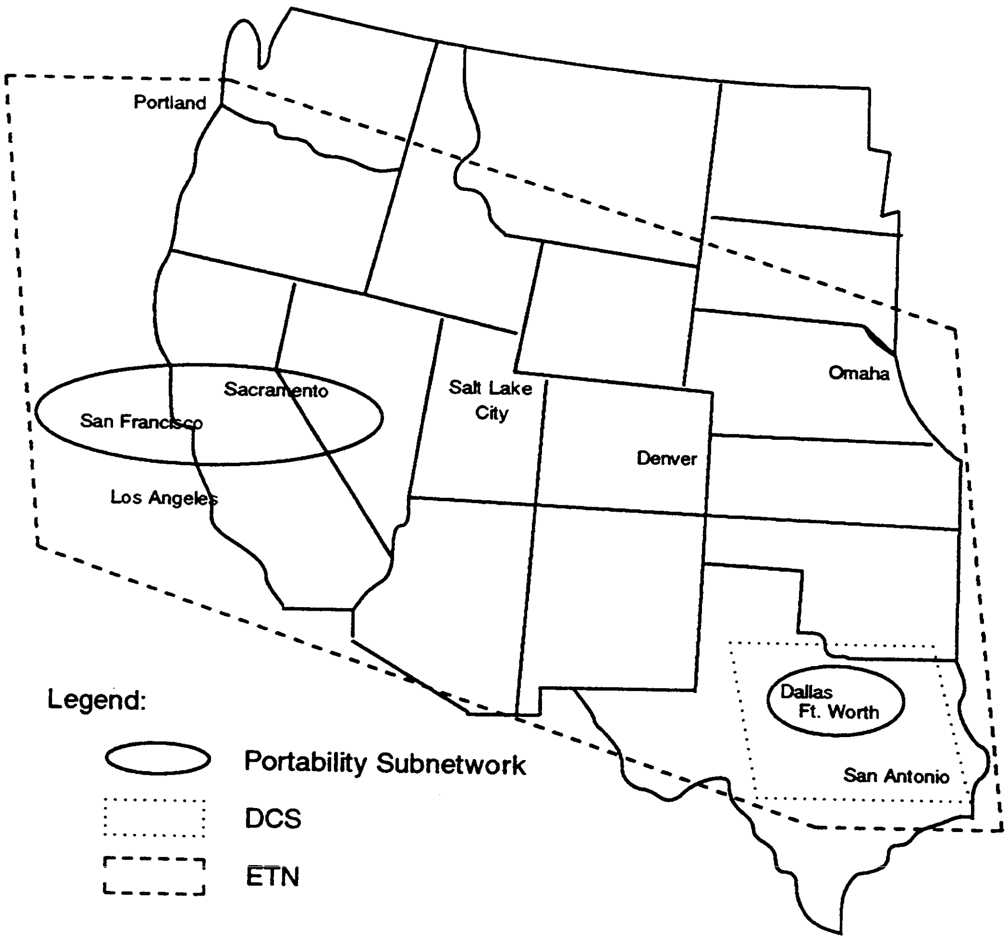

Figure 56-4. Mixing System 85 Networking Arrangements

In the example in Figure 56-4, a second portability subnetwork is set up between San Francisco and Sacramento, California. While Extension Number Portability allows users to move from San Francisco to Sacramento and retain their old extension number, ETN number, etc., these switches are not part of a DCS. Feature transparency is not provided as it is in the arrangement in Texas. While the DCS and Extension Number Portability features are fully compatible, they are not the same. These features can be used together on the same network, they can be used in partial conjunction with one another as in the Texas network in Figure 56-4, or they can be completely independent of one another.

User Operating Procedures

The Extension Number Portability feature does not require any new user operating procedures if 4- or 5-digit dialing is already in use. If not, users will have to be reminded that they need only dial a 4- or 5-digit extension number to reach users within the subnetwork. Users in a newly established portability arrangement may notice slightly longer call setup times when calling an extension on another switch. This is due to the trunk siezure and setup times. This can be minimized by using ISDN—PRI trunking between switches.

Considerations

Although this feature has the distinct advantage of being able to move extension numbers along with users, several switching aspects must be considered when evaluating its applications.

4- or 5-Digit Dialing

A coordinated 4- or 5-Digit Dialing plan must be common to all switches within the portability subnetwork.

Porting Extensions Across Toll Boundaries

Porting extension numbers across toll boundaries may not be advisable. When this is done, a call placed from the public-network in the new location must be placed as a ***toll*** call to reach the ported extension now located in the local calling area.

As an example, suppose that (as shown in Figure 56-4), an employee transfers from San Francisco to Sacramento, and the employee's extension number is ported. When this is done, the employee's spouse (who has also moved to Sacramento) must dial a San Francisco telephone number to reach the employee at work. Of course, this call could also be placed through the local attendant in Sacramento without incurring toll charges.

Effects on Tie Trunk Usage

Extension Number Portability requires each node (switch) in a portability subnetwork to be able to route calls out over tie trunks to any of the other nodes. For certain types of calls, this means that additional trunking facilities may be required to complete the call. If portability is not used, when users move their number is changed. In this case, calls from the public or private network (using the new number) route directly to the user's new location. Routing on the new number may be more efficient than through portability as switches outside the portability subnetwork have no way of determining on which switch a ported extension resides.

- An outside ***private-network*** switch routes the call to the nearest portability subnetwork node and lets that node handle the final routing.
- An outside ***public-network*** switch always routes to the number's original location within the portability subnetwork.

Since the extension may reside on another node, an additional trunk may be required to complete the call. This situation is shown in Figure 56–5.

The solid line shows the routing of an incoming DDD call using the moved user's changed telephone number (no portability). The dashed line represents a call to the same user; only this time, the user kept their old telephone number through portability. The nodes of a portability subnetwork may require additional trunk circuits to maintain the desired balance of facilities established by system engineering.

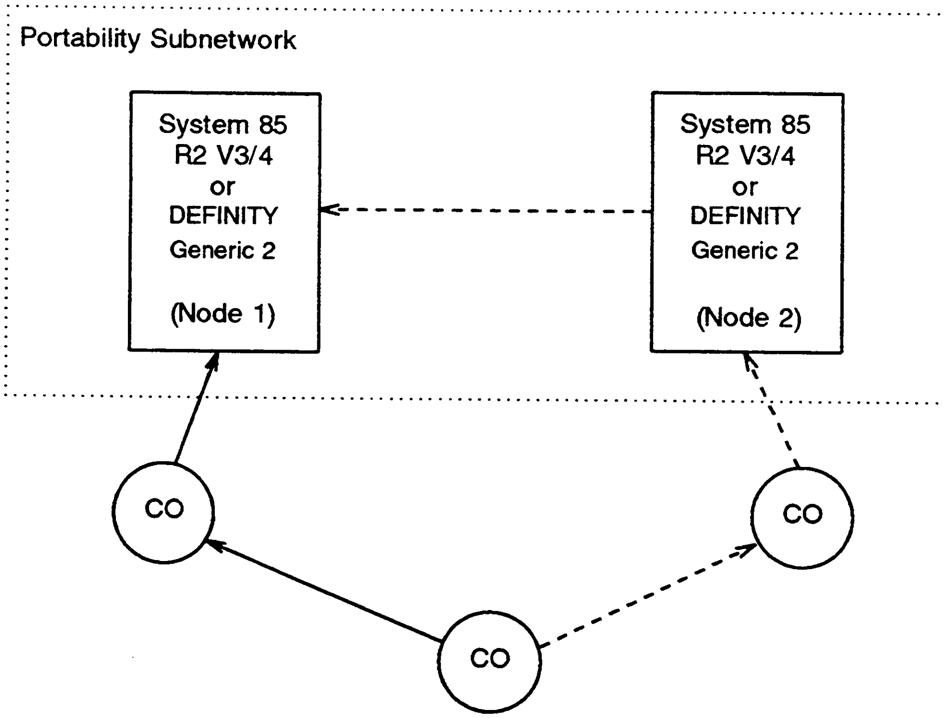

Figure 56-5. Call Routing With A Portability Subnetwork

Effects on Voice Terminal Features

When moving to another node, the user must consider the effects the move will have on feature operation. Some features cannot be used between nodes (for example, Attendant Busy Lamp Field). Other features can be used between nodes as long as the nodes are part of a DCS (such as, Call Forwarding). Still other features must be abandoned at the old node but can be reestablished at the new node (for example, Call Coverage). These aspects are further described in "Interactions" section.

Effects on Switch Administration

Administration of the switches and data that must be maintained at each node in the subnetwork is the main concern. To provide Extension Number Portability, each node in the subnetwork must maintain extension number assignments for all nodes. These assignments must be kept up-to-date. Whenever an extension number is moved to a new node, each node in the subnetwork must be informed of the change. See the "Feature Administration" section of this chapter for a detailed explanation.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of the Extension Number Portability feature.

Attendant Direct Extension Selection With Busy Lamp Field

The Attendant DXS/BLF feature does not support unrestricted 5-digit dialing. For switches with unrestricted 5-digit dialing the Attendant DXS buttons and BLF lamps are inoperative.

Attendant Display

When 5-Digit Dialing is used, the attendant console should have an 8-character alphanumeric display. The 4-character alphanumeric displays show only the first (leading) four digits of a 5-digit extension number. This is not sufficient to identify the extension.

AUDIX

When an extension is ported from one node in a portability subnetwork to another and a centralized AUDIX serves both switches, the centralized AUDIX system must be informed of the extension's move.

AAR (Automatic Alternate Routing)

On System 85 and DEFINITY Generic 2.1 switches, the AAR feature must be used to route calls inside a portability subnetwork. Instead of the usual AAR assignment of an RN(X) to a pattern, the ENP feature associates a pattern with a node number. Extension Number Portability requires network routing which may result in additional call-setup time (over local calls).

ACD (Automatic Call Distribution)

Extension numbers residing at other nodes cannot be part of an ACD split at the local node.

AIOD (Automatic Identification of Outward Dialing)

The hardware associated with AIOD does not support 5-digit dialing. Only the four least significant digits of a 5-digit extension number are identified. (Deleting the first digit may be satisfactory if only one first digit is used on the switch.)

Bridged Call

Every image of an extension number must be removed from the old node when the extension number is moved to a new node. Bridging from one node to another is not allowed.

Call Coverage

If an extension is assigned Call Coverage or is a point in a coverage path, these assignments must be removed before the extension can be moved to another node. The extension number can then be assigned to another coverage group at the new node, if desired.

CDR (Call Detail Recording)

Each switch in a portability subnetwork is assigned a node number. A data item in the CDR is provided which contains this node number so that, if necessary, subsequent accumulation of records from switches carry an identifying field from which a multinode call can be constructed. This field contains the value of the node number assigned in Procedure 275, Word 3.

Call Forwarding—Busy and Don't Answer

Call Forwarding—Busy and Don't Answer is not a transparent DCS feature on System 85. Calls cannot be forwarded to other DCS nodes using this feature. Consequently, an extension number that has been ported to another node cannot be a forwarded-to extension from the old node.

Call Forwarding—Don't Answer

Call Forwarding—Busy Don't Answer is not a transparent DCS feature on System 85. Calls cannot be forwarded to other nodes using this feature. Consequently, an extension number that has been ported to another node cannot be a forwarded-to extension from the old node.

Call Forwarding—Follow Me

Call Forwarding—Follow Me is a transparent DCS feature. However, calls to a terminal with forwarding active to an extension number that was local when forwarding was established, but was subsequently ported, do not route properly. To correct this problem, the forwarding must be canceled and then reapplied.

Call Pickup

When an extension number is moved to a new node, it must be removed from its old pickup group and added to a new one, if desired.

Calling Number Display to Station

The hardware associated with this feature does not support 5-digit extension numbers. Digital terminals with the 40-character display must be used to display 5-digit extension numbers.

Conference—Attendant Five Party

Because the attendant may not know which extension numbers are local and which ones have been ported (ported extensions connect to the local switch via a tie trunk), all conferences should be held on the console until completion. If the attendant releases a conference when only trunks are involved, the switch drops the conference.

Conference—Attendant Six Party

Only the attendant console and local voice terminal circuits can provide the disconnect supervision required to release a conference circuit once the conference is finished. Since the attendant may be unaware of which extension numbers are local and which ones have been ported (ported extensions connect to the local switch via a tie trunk), all conferences should be held on the console until completion. If the attendant releases a conference when only trunks are involved, the switch drops the conference automatically to prevent the conference circuit from locking up.

DCS (Distributed Communications System)

A DCS does not provide full feature transparency. As indicated in other parts of this section on interactions, several features cannot be used between nodes. This must be pointed out because it is not readily apparent where any particular extension number resides. A user may try to activate some feature toward an extension number thinking that the extension number resides on the local switch because it looks similar to their own. In most cases, it probably will. But if the extension number has been ported to another node, some types of feature activations toward this extension number are denied. Although the user must be removed from these features at the old node, they can be added to these same features at the new node, if provided.

Extension Number Steering

This feature can provide a kind of "portability" for switches that are not part of the ENP subnetwork, but this capability is separate from the Extension Number Portability feature.

Hunting

An extension number must be removed from a hunt sequence before it can be moved to another node.

Intercom

An extension number must be removed from all intercoms before it can be moved to another node.

Last Number Dialed

The Last Number Dialed feature stems and redials 4- or 5-digit extension numbers dialed to voice terminals in a portability subnetwork.

Message Waiting—Automatic

The operation of Automatic Message Waiting is not changed; but with the expanded numbering plan beginning with R2 V3 (possible 100,000 extension numbers), "it takes longer to update the status of each AMW lamp.

Personal CO (Central Office) Lines

A user's Personal CO Line must be reassigned using a line to the CO serving the new node. This may require a number change for that Personal CO Line.

Tenant Services

Because AAR is not a partitioned feature, an extension partition in a partitioned System 85 or Generic 2 can only serve as a satellite location in a portability subnetwork. Therefore, Extension Number Portability is allowed as long as the ported number is consistent with the numbering plan in the partitioned switch. For example, the ported extension number is not allowed to duplicate an extension number in another partition. Also, the extension numbers in every extension partition of the partitioned switch and in the associated portability subnetwork must contain the same number of digits. That is, both the partioned switch and the portability subnetwork must use a compatible 4- or 5-digit dialing plan.

Unattended Console Service—Preselected Call Routing

Unattended Console Service—Preselectd Call Routing is not part of DCS transparency. The designated extension number cannot be moved to a new node and still function as the night station for the old node.

WCR (World Class Routing)

The World Class Routing feature is fully compatible with the Extension Number Portability feature when standard network is active.

Generic 2.2 switches use up to seven routing networks to provide private network call routing. The same networks are used for the portability subnetwork. For locally originated calls, extension numbers are evaluated by the internal digit analysis software. If an extension number is recognized as ported, it is passed to WCR network 0 for digit modification and then routing. For incoming trunk calls, the process is the same except that the number is in a network format and initial analysis is accomplished by network digit analysis for the private network.

Hardware Requirements

No special hardware is required for this feature. However, the following hardware considerations may be applicable.

For Traditional Modules:

Extension Number Portability may increase the call traffic between nodes in the portability subnetwork and, therefore, additional tie trunks (SN233) maybe required.

For Universal Modules:

Extension Number Portability may increase the call traffic between nodes in the portability subnetwork and, therefore, additional tie trunks (TN760C) may be required.

Feature Administration

On System 85 switches, Extension Number Portability is administered using the MAAP (Maintenance and Administration Panel). Certain administration can also be performed by the customer using the SMT (System Management Terminal) or using the AP (Applications Processor) System Management applications.

On DEFINITY Communications System Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

In a portability subnetwork, initial and subsequent switch administration must be carefully planned and coordinated. Whenever an extension is moved between nodes in a portability subnetwork, some administration is required at every node in the subnetwork.

Initial Administration for the Portability Subnetwork

The following administration must be performed at each switch in the portability subnetwork.

For All Switches in the Portability Subnetwork

Procedure 204, Word 1 — Console Messages and Listed Directoy Numbers

Every LDN (Listed Directory Number) must be administered on every node in the portability subnetwork. This procedure is used to administer local LDNs. Remote LDNs are administered as extension numbers that reside at remote nodes (see Procedure 354, word 1).

Procedure 275, Word 3 — System COS - Miscellaneous — Multimachine nodes:

- Switch Type: Enter Type 2 (Portability) or Type 3 (DCS and Portability).
- Local Switch Number: Enter a unique node number for this switch.

Procedure 276, Word 1 — Feature Group Class of Service:

- Standard Network: Enter 1 to activate standard network (required) for each switch.
- Enter 1 to activate multipremises software (required) for each switch.

following administration is required to support 4- or 5-digit dialing.

Procedure 350, Word 1 — Dialing Plan - First Digit

Beginning with System 85 R2 V3, any digit 0 through 9 can be assigned as the first dialed digit of a group of extension numbers. These assignments must be common to every node in the portability subnetwork.

- First Dialed Digit: Assign the digits 0 through 9 as the first dialed digit for extension numbers. The same set (first digits) must be the same for extension numbers on all nodes of the portability subnetwork.

- Number of digits: Enter "4" or "5."
- Call type: Enter "1."

Procedure 354, Word 1 — Extension Number Groups

Each extension number in the portability subnetwork must be assigned to a node number at each node in the subnetwork. Extension numbers are initially assigned in groups. Extension number groups assigned to terminals that reside at other nodes must be assigned a node number.

- First Extension Number of the group (for example, 7000 or 70000)
- Last Extension Number of the group (for example, 7999 or 79999)
- Node Number (1 to 999).*

Procedure 354, Word 2 — Extension Destination

Each ported extension must be associated with the appropriate node number at each node in the portability subnetwork.

- Extension or Steering Code (for example, 71234)
- Use. Enter "3."
- Node number (field 7). Enter the gaining (ported-to) node number.

For System 85 and DEFINITY Generic 2.1 Switches

Procedure 285, Word 1 — System COS - Network — Network Uniform Numbering Plan:

- Number of digits in location code Enter "3."
- Number of extension digits: Enter "4."

Procedure 321, Word 4 — Automatic Alternate Routing-Routing

Each RNX (location code) used in the portability subnetwork must be assigned as a home location code at each ENP node. Remote location codes are assigned for non-ENP nodes within the DCS, ETN, or AUTOVON arrangement.

For ENP Nodes

- Location Code (RNX): Enter location code (RNX) number.
- Node Number: Enter dash (-) for home location codes.
- Pattern Number: Enter 641 for home location codes.
- First Digit: Used only for home location codes with 5-digit dialing. The digit prefixed to the last four digits of the uniform number to form a 5-digit extension number.

* For portability subnetworks that are part of a DCS, it is recommenended but not required that the portability subnetwork use the DCS node numbers.

For Non-ENP Nodes:

- Location Code (RNX): Enter location code (RNX) number.
- Enter node number for location codes that are remote to the portability subnetwork but part of the DCS, ETN, or AUTOVON arrangement.
- Enter AAR pattern number used to route calls for location codes remote to the portability subnetwork.

Procedure 322, Word 4 — Extension to Home Location Code Translation

The first or first and second digits of a group of extension numbers are associated with a location code. This procedure is used to convert 4- or 5-digit subnetwork extension numbers into 7-digit uniform numbers for routing between subnetwork nodes. The association between a first (or first and second) digit of a group of extension numbers and a location code should be consistent with the assignment made in Procedure 321, Word 4. That is, the same first digit should be associated with the same location code in both procedures (321, Word 4 and 322, Word 1).

For DEFIIWW Generic 2.2 Switches

Procedure 354, Word 4 — Node Number to VNI Mapping

Each node number in the portability subnetwork must be associated with its corresponding WCR VNI.

- Node Number. Enter the assigned number (range 1 through 999). One for each node in the portability subnetwork.
- Virtual Nodepoint Identifier. Enter the corresponding VNI (range 1 through 1023).*

Once the VNI is assigned, the ported extension number is passed to WCR network 0 for further processing.

Procedure 101, Word 3 — Trunk Group Characteristics — Prefixing

One of the WCR routing networks (networks 2 through 7) is used to support the portability subnetwork. This network processes all incoming calls on tie trunks from other nodes of the portability subnetwork. These incoming calls are directed to the supporting WCR network through trunk group prefixing. When the call reaches the local switch over one of these tie trunks, the assigned prefix is added to the beginning of the address digits. When this prefix is the WCR dial access code for the supporting network, it insures that the incoming call is processed correctly for a subnetwork call.

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* A value of "0" can be assigned for a VNI, however, this is a non-routing VNI and would result in intercept. VNI 0 cannot be used for a valid destination.

Field 1 Trunk Group: Enter the trunk group number for the portability subnetwork tie trunk group.

Field 2 Type of Address: Enter a dash (-) for non-ISND—PRI trunk groups. For ISDN—PRI trunk groups (signaling type 20), enter the address type code for the form of address expected on incoming calls as follows:

0 = Unknown (any address form may be possible)

1 = International address format

2 = National (NANP 10-digit) address format

3 = Not defined

4 = Subscriber address format (such as, private network UDP, 7-digit)

5 = Not defined

6 = Abbreviated (special) address format

7 = Not defined.

Fields 3 through 6, Digits 1 through 4: Enter the network dial access code for the WCR network used to support Extension Number Portability.

Field 7, Signaling Type: Enter the appropriate signaling type for the trunk group.

Procedure 314, Word 1 — Network Digit Analysis and Dialing Plan

Home Location Codes:

As with earlier switches, each location code used in the portability subnetwork must be defined as a "home location code" on each node in the subnetwork. For Generic 2.2 switches, this is done in the WCR private network that supports the portability subnetwork, using Procedure 314, Word 1, and constitutes the dialing plan for that network.

Home Location Code:

- ***Setting Up the Dialing Plan in the Portability Network (Incoming Calls)***

In the WCR network used for extension number portability, each home location code must be defined by a string identifier. This may require defining the complete set of location codes or, if the first, or first and second digits of all the location codes are common, it may be possible to use a 1- or 2-digit string identifier. This must be done on each switch in the portability subnetwork.

Fields 1 through 3, Digits: Enter the digits of the home location codes as necessary.

Field 7 Segment: Enter "1."

Field 8 Last Segment: Enter "1."

Field 9 String Length: Enter "7." Generally, a WCR network address will have seven digits. However, if additional digits are needed, indicate the total number of digits in a network address.

Field 10 String Type: Enter "6."

Field 11 Action: Enter "1."

Field 12 Action Object: Enter the appropriate digit modification index number from Procedure 320, Word 1.

Field 13 Action Attribute: Enter "0."

Field 14 Network Number: Enter the network number (range 2 through 7) for the WCR network used for the portability subnetwork.

The restart action will pass the received digits to digit modification using the digit modification index number entered in field 12, Action Object.

Dialing Plan:

- ***Setting Up the Dialing Plan in Network 0:***

In network 0, each extension number group must be defined by a string identifier. This corresponds to the extension number group assignments made in Procedure 354, Word 1. Usually one or two digits will be sufficient to identify each block of extension numbers. This is done at system setup time.

Fields 1 (and possibly 2) Digits: Enter the digits necessary to specifiy the extension number range (group). Usually this will require only the first digit.

Field 7 Segment: Enter "1."

Field 8 Last Segment: Enter "1."

Field 9 String Length: Enter "4" or "5" as appropriate.

Field 10 String Type: Enter "6."

Field 11 Action: Enter "1."

Field 12 Action Object Enter the appropriate digit modification index number from Procedure 320, Word 1.

Field 13 Action Attribute: Enter the network number (range 2 through 7) for the WCR network used for UDP. (Although an entry is required, this field is ignored for Extension Number Portability calls.)

Field 14 Network Number: Enter "0."

The restart action passes the dialed digits to digit modification using the digit modification index number entered in field 12, Action Object, and then routes the call using the VNI specified in Procedure 354, Word 4.

Procedure 320, Word 1 — Network Digit Modification

There are two cases where digit modification is used with Extension Number Portability. One is for home location codes and one is for ported extensions.

Home Location Codes (Incoming Calls)

Once an incoming tie trunk call has been identified as belonging to a home location code, digit modification performs the digit deletion and / or insertion required for extension number addressing. For incoming tie trunk calls, digit modification is used to correct the network address (usually by deleting the location code) to an extension number for the home location code. Procedure 320 provides the instructions for this digit deletion.

1. Digit Modification Index Number. This number is used by the switch to locate a particular set of digit modification instructions. The number from this field is entered in field 12 of Procedure 314, Word 1.
2. Digits to Delete: For assignment of a "home location code" this will usually be "3" to convert the network address to an extension number.
3. Segment Number: For assignment of a "home location code" this will usually be:

   "0" for 4-digit extension numbers indicating no digits to insert

   or

   "1" for 5-digit extension numbers.
4. INSERTION DIGIT SEGMENTS ONE THRU FOUR

   For assignment of a "home location code," fields 4 through 11 are not usually used for 4-digit extension numbers. For 5-digit extension numbers the prefix (leading) digit of the extension number must be added in field 4.

For incoming calls (tie trunk), when digit modification is completed, the call is restarted in network 0 which passes it to the internal dial plan for processing. WCR network 0 does not perform digit analysis on these incoming calls.

Dialing Plan (Outgong Calls)

Once an extension number has been identified as a ported extension, digit modification performs the digit deletion and / or insertion required for network addressing. For outgoing calls in extension number format, digit modification inserts the digits necessary (usually the location code) to form a WCR network address. Procedure 320 provides the instructions for this digit insertion.

1. Digit Modification Index Number. This number is used by the switch to locate a particular set of digit modification instructions. The number from this field is entered in field 12 of Procedure 314, Word 1.
2. Digits to Delete: For outgoing calls to ported extensions this will usually be "0." However, this may be "1," if the first digit of the extension number does not match the first digit of the location code.
3. Segment Number: For a ported extension number, this will usually be "1."
4. INSERTION DIGIT SEGMENTS ONE THRU FOUR

   For a ported extension number, usually only fields 4, 5, and 6 are used to insert the location code (for construction of the network address). Enter the network location code for the ported extension number (new node).

For ported extension numbers, when digit modification is complete, the call is passed to generalized route selection. Reanalysis in the network specified in field 13, Procedure 314, Word 1 is not required because the VNI was assigned previously in Procedure 354,

ADMINISTRATION PROCEDURES — EXTENSION NUMBER PORTABILITY

PROCEDURE	WORD	PURPOSE	SMT
204	1	Assigns local Listed Directory Numbers.	No
275	3	Assigns the node number and type of switch.	Yes
276	1	Assigns feature groups to the switch, including standard Network and Multipremisis (required for Extension Number Portability).	No
285	1	For System 85 and Generic 2.1 switches, assigns the number of digits in the location codes and extension numbers of a network. (Must be "3" and "4," respectively, for both 4- and 5-Digit Dialing). For Generic 2.2 switches, assigns system class of service characteristics including WCR route selection method.	Yes*
314	1	On Generic 2.2 switches, assigns network dialing plan characteristics, including string identification used with digit analysis and VNI designation, used for call routing. On System 85 and Generic 2.1 switches, this procedure is not used for Extension Number Portability.	N/A
315	1-2	On Generic 2.2 switches, used to display the results of network digit analysis assigned in Procedure 314. Assists in network analysis and debugging, including the portability subnetwork.	N/A
320	1	On System 85 and Generic 2.1 switches, this procedure is not used for Extension Number Portability. On Generic 2.2 switches, this procedure provides digit modification instructions for numbers being re-routed to another network (such as the portability subnetwork).	N/A
321	4	On System 85 and Generic 2.1 switches, assigns the AAR routing patterns to a location code and first digit. Not used with Generic 2.2	Yes
322	1	For System 85 and Generic 2.1 switches, associates the first or first and second digits of a group of extension numbers with a location code. For Generic 2.2 switches, defines outgoing ISDN-PRI parameters for a routing preferance.	No

* Display only procedure for the SMT.

(Continued)

ADMINISTRATION PROCEDURES EXTENSION NUMBER PORTABILITY (Continued)			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
350	1	Assigns a call type (extension number or trunk or feature DAC) and number of digits (4 or 5 for extension numbers) to each first dialed digit.	No
354	1	Assigns groups of extension numbers to the local node and to remote nodes.	No
354	2	Associates an extension number (or steering code) with a location code, DAC, or node number. This procedure is used to change the node assignment for an extension number that has been ported to a new node. It is also used to move extension numbers to a node outside of the portability subnetwork.	No
354	4	For Generic 2.2 switches, associates the portability subnetwork node numbers with the corresponding WCR VNIs.	N/A

Facilities Restriction Level

Description

The FRL (Facilities Restriction Level) provides access control and protection from unauthorized use for switch and network facilities. Each originating facilty (voice and data terminals, incoming trunk, remote access trunks, and attendant positions) is assigned an FRL. Each trunk group and network routing preference (AAR [Automatic Alternate Routing], ARS [Automatic Route Selection], or WCR [World Class Routing] feature) is also assigned an FRL. Since the same trunk group can be assigned to more than one routing pattern, the FRL may also be different in different routing patterns. To illustrate, a public-network trunk group may have an FRL of five in an AAR pattern and an FRL of two in an ARS pattern.

When a user places a call, the switch selects a routing pattern based on the dialed digits. The switch compares the FRL assigned to the calling facility ***(the default FRL)*** to the FRL of the first preferance in the pattern. A call can access a trunk only if the FRL of the preference is lower than or equal to the FRL of the call. If the call has a high enough FRL to access the first preference, the switch checks for an available trunk If all trunks are busy, the switch checks each alternate preference to which the call has access.

Feature History and Development

The FRL feature was first available in Release 1. The Authorization Codes feature, a companion feature to the FRL feature, was enhanced in Release 2, Version 3. Otherwise, there have been no changes to this feature since its introduction.

Authorization Codes

If a call cannot be routed using the default FRL and a higher FRL would allow access to more trunk groups, the switch prompts the caller for an authorization code. An authorization code allows a caller to override the FRL assigned to the terminal being used. For instance, an executive may wish to override the authorization code assigned to another person's terminal or one available to the general public. The Authorization Code is assigned its own FRL, and if higher than the default FRL, the authorization code FRL is used to allow the search for an outgoing trunk facility to continue.

Network Access Flag

A second characteristic of an Authorization code is the network access flag assigned to each authorization code in Procedure 282, Word 1, Field 3. This flag indicates whether or not the associated authorization can be used from an off-net location. That is, if the network access flag is set at "0," that authorization code cannot be used from a an off-net station (such as with the Remote Access feature). If the network access flag is set to "1," then the authorization code can be used from off-net locations. See the Authorization Codes feature for further details.

Hierarchy of Levels

There are eight levels of facility restriction (0 through 7). Zero is the lowest level of access, and seven is the highest. Trunk groups could be assigned FRLs as follows:

FRL	Calling Privileges
0	Local (intraswitch)
1	Private Network
2	FRL #1 + local public network
3	FRL #2 +FX
4	FRL #3 + WATS, Band 1
5	FRL #4 + WATS, Bands 1 and 2
6	FRL #5 + WATS, (All Bands)
7	FRL #6 + DDD

Normally, most private-network trunk groups will have low FRLs. Premium trunk groups including public-network trunk groups with toll calling access will usually have higher FRLs. Using this scheme, most employees can place private-network calls while only employees with access to high enough FRLs can access the public network. Assigning high FRLs to expensive public-network (FX and WATS) and sensitive private-network trunk groups, limits access to these facilities. Incoming trunk groups (tie trunks and remote access trunks) can be set up so that the switch always requests an authorization code.

Alternate FRLs

Alternate FRLs are used to change access controls for periods of other than normal activity (such as, evenings or weekends). Alternate FRLs are based on primary FRLs, and when invoked, change the FRL for all facility or authorization codes assigned the primary. To activate or deactivate the alternate FRLs, the attendant simply presses a button on the attendant console. Alternate FRLs can be higher, lower, or the same as the primary FRLs they replace. Less restrictive alternate FRLs can, for example, permit access to public-network trunk facilities on the weekend or after hours when toll rates are lower. Consequently, users normally denied public-network calling privileges can place these calls. More restrictive alternate FRLs can be turned on after business hours to prevent unauthorized personnel from making toll calls. Note: Invoking alternate FRLs changes the FRL for call origination only. It does not change the FRL requirement assigned to outgoing trunk groups or AAR, ARS, and WCR routing preferences.

FRL Raising

The FRL raising function applies only to queued calls that were placed using one of the networking features, AAR, ARS, or WCR. Just before the "last try," when the time-in-queue limit elapses for the queued call, the switch can raise the call's FRL to help provide an allowable trunk for the call. FRL Raising is assigned on a per-system basis in Procedure 330, Word 1, field 4.

FRL Raising first compares the timed-out call's current FRL with the assigned ***Threshold FRL*** (Field 3). If the call's current FRL is equal to or greater than the Threshold FRL, the call is qualified for FRL Raising. At this time, the switch considers substituting the assigned Raised FRL (Field 4) for the timed-out call's current FRL. Substitution is made if the Raised FRL will be higher than the current FRL.

TCM (Traveling Class Mark)

When a routing network (AAR, ARS, or WCR) call routes to another network switch (ETN) by way of a tie trunk, a TCM digit maybe added at the end of the dialed number. Network switches use this digit in place of the FRL of the tie trunk to determine an FRL for the call. The TCM is only sent from one switch to another over tie trunks administered as "network tandem" (for the AAR or ARS feature) or with field 3 of Procedure 103, Word 1, set to either "1" or "2" (for the WCR feature). The TCM is based on the FRL of the call at the originating switch and may correspond to the default FRL of the originating facility or a raised FRL either from FRL Raising or an Authorization Code, if required to complete the outgoing connection. Note: A second TCM may also accompany the intertandem call. If so, the second TCM is a ***conditional routing count*** TCM and has nothing to do with FRL.

User Operations

The basic FRL feature is automatic, and its operation is transparent to the user.

To Activate Alternate FRLs:

Press the **[ALTERNATE FRL]** button. [ALTERNATE FRL lamp lights.]

To Deactivate Alternate FRLs:

Press the **[ALTERNATE FRL]** button. [ALTERNATE FRL lamp goes out.]

To Use an Authorization Code FRL:

1. Place the call in the normal manner. [Call-progress tone]

   (If an authorization code is needed, recall tone is returned followed by second dial tone.)

2. Enter your assigned Authorization Code. [Call-progress tone from distant switch]

Considerations

Hard and Soft Processor Swaps

FRLs, Alternate FRLs, and Authorization Code FRLs are stored in a translation portion of switch memory. Therefore, these assigned FRLs will endure a hard processor swap.

Alternate FRL activations are stored in a status portion of switch memory. Therefore, if an attendant activates Alternate FRLs and then a hard processor swap occurs, Alternate FRLs will not be active after the swap is finished.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of the feature.

AAR (Automatic Alternate Routing)

The FRL feature is used with the AAR feature to control access to trunk facilities in network routing patterns. Each preference within each routing pattern is assigned an FRL. If the FRL of a call is not high enough to access an idle trunk in a preference, that preference is skipped in the search for an accessible preference.

ACD (Automatic Call Distribution)

When interflowed ACD calls use a network routing feature to route these calls outside the local switch, the FRL of the split's supervisor is used to determine whether these calls can use the available network facilities.

ARS (Automatic Route Selection)

The FRL feature is used with the ARS feature to control access to trunk facilities in network routing patterns. Each preference within each routing pattern is assigned an FRL. If the FRL of a call is not high enough to access an idle trunk in a preference, that preference is skipped in the search for an accessible preference.

Bridged Call

An FRL is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When an FRL is assigned to a ***shared extension,*** the calling privileges granted by the FRL apply to every image of that extension.

CDR (Call Detail Recording)

The call record stores the FRL that is used for call routing.

Call Vectoring

In Procedure 010, Word 3, an FRL can be assigned to VDNs for use with the Route-To command. The FRL of a VDN is used to determine whether the call is allowed to route over available network facilities.

DCA (Data Communications Access)

Facilities Restriction Levels provide a means of restricting terminal access to trunk groups via the AAR (Automatic Alternate Routing) or WCR (World Class Routing) features. The

DCA trunk group(s) is placed in a separate network routing pattern to which an FRL is assigned. When the network routing feature (either AAR or WCR) is used to call the DCA number, the user's FRL is compared to the FRL of the network routing pattern before access to the DCA port is granted. The attendant can also use the alternate-FRLs attribute to alter these restrictions when appropriate.

EUCD (Enhanced Uniform Call Distribution)

When interflowed EUCD calls use a network routing feature to route these calls outside the local switch, the FRL of the split's supervisor is used to determine whether these calls can use the available network facilities.

Host Computer Access

Each Host Computer Access trunk group can be assigned an FRL to restrict certain users from access to the host computer. An attendant or the switch administrator (user of Applications Processor System Management applications) can invoke an alternate set of FRLs when desired.

Look-Ahead Interflow

The Look-Ahead Interflow feature uses the FRL assigned to the VDN that the calling party originally dialed to divert incoming calls outside the R2 V4 System 85 or DEFINITY Generic 2.

If the VDN associated with the sending switch's vector does not have a high enough FRL to access the first preference in the network routing pattern determined by the destination digits of the "route to" step, the Look-Ahead Interflow software considers the "route to" step as having an invalid destination. (If the invalid "route to" step is the final effective step in the vector, the step is treated as a "stop" step. Otherwise, the sending switch continues vector processing with the next sequential step.)

The network routing of a Look-Ahead Interflow "route to" step can be denied when Alternate FRLs are activated. After an attendant at the sending switch activates Alternate FRLs, the FRL of the VDN associated with the sending switch's vector changes to the assigned (in Procedure 286, Word 1) Alternate FRL. If the Alternate FRL is not high enough to access the first preference in the network routing pattern determined by the destination digits of the "route to" step, the "route to" step is considered to have an invalid destination.

Queuing

The Queuing feature uses FRL to determine a call's ability to access a trunk group. A call will not queue on a trunk group it cannot access. Also, the Queuing feature exercises the FRL raising function for a last chance effort to complete a call in queue. The FRL raising function is first tested against a Threshold FRL set in the Queuing feature that is used to determine a calls eligibility for FRL raising.

Remote Access

An FRL is assigned to remote access trunks as a call originating facility. This FRL is then used as the FRL for incoming remote access calls unless an authorization code is used to control use of the Remote Access feature. If an authorization code is used for access to the Remote Access feature, and the network access flag of the authorizaion code allows its use from off-net locations, then the FRL of the authorization code is used for the remote access call.

WCR (World Class Routing)

The FRL feature is used with the World Class Routing feature to control access to trunk facilities in network routing patterns. Each preference within each routing pattern is assigned an FRL. If the FRL of a call is not suflficient to access an idle trunk in a preference, that preference is skipped in the search for an accessible preference.

Hardware Requirements

None.

Feature Administration

Assignment of the Facilities Restriction Level is on a per-extension class of service and per-trunk group basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal), TCM (Terminal Change Management), or FM (Facilities Management) features.

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — FACILITIES RESTRICTION LEVEL			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to an extension number.	Yes
010	3	Assigns an FRL to a extension class of service.	Yes
103	1	Administers network trunk group parameters including Authorization code requirements Default FRL for the trunk group Incoming tie trunk access to network TCM characteristics.	Yes
200	1	Assigns an FRL to an attendant console.	No
203	1	Assigns the Alternate FRL button to an attendant console. The applicable encode is as follows: 19 Alternate FRL.	No
283	1	Displays the extension numbers, trunk groups, or authorization codes associated with a specific FRL.	Yes
286	1	Administers alternate FRLs.	Yes
309	1	For System 85 and Generic 2.1 switches, assigns the FRL to an ARS routing preference.	Yes
318	1	For DEFINITY Generic 2.2 switches, assigns the FRL to a WCR routing preference.	N/A
321	1	For System 85, and Generic 2.1 switches, assigns the FRL to an AAR routing preference.	Yes
330	1	Assigns FRL raising and the associated Threshoold FRL.	Yes

The following are the applicable TCM path names used with the AP 16.

FACILITIES RESTRICTION LEVEL	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Use this screen to display and change the FRL associated with a extension class of service.
terminal-change extensions facility-restriction	Displays or prints a report of the extensions with an assigned FRL.

The following are the applicable FM path names used with the AP 16. A printed report of the displayed information can also be generated.

FACILITIES RESTRICTION LEVEL	
PATH NAME	**PURPOSE**
facilities-mgmt facility-rstr trunk-groups	Displays and changes the FRL assigned to a trunk group(s).
facilities-mgmt facility-rstr extensions	Displays and changes the FRL assigned to an extension(s).
facilities-mgmt facility-rstr alternate-set	Displays, changes, turns on, and turns off alternate FRLs.

Force Administration Data System

Description

The FADS (Force Administration Data System) feature collects and stores traffic-related information for CAS (Centralized Attendant Service) and/or for UCD (Uniform Call Distribution) groups. This information allows the customer to intelligently adjust the make-up of the CAS attendants and/or UCD groups to suit the call-load requirements.

> **NOTE:** The UCD feature is not available in switches after Release 2, Version 1. The call distribution services provided by UCD have been replaced and enhanced first by the EUCD feature (Release 2, Version 2) and subsequently by the ACD feature (Release 2, Version 3). The following material that relates FADS to UCD should not be construed as a current offering. This material is meant to. serve as a reference source for those using earlier software packages. The FADS feature may be used to monitor CAS traffic activity in all software packages.

FADS Output

Output of FADS information is available from a display unit or in hard-copy form obtained from an optional printer. The traffic records available contain information such as:

- Calls attempted
- Calls completed
- Call attempts abandoned, and
- The amount of time that calls remain in queue.

The Display Unit: The FADS display unit consists of a 4-digit display field, a 3-digit display field, and a touch-tone dialing pad. The dialing pad is used to request a display of individual traffc items or to request a complete printout of every traffic item.

Display Unit Commands: The commands that control the FADS display unit are easily executed on a display unit dialing pad A command is executed with one or two button proses. As an example, some of the basic commands are:

- Pressing the "*" button clears the previous command and displays the time of day.
- The command "*9" displays the number of abandoned calls to a group.
- The "#" button may be used alone or in conjunction with the digits (e.g., "# 3") to test the display unit.
- The command "*6" requests a complete printout of the current traffic information.

Other commands are available and can be found in the appropriate user's documents.

Display Time Out: When executing cornmands on the display unit, the maximum display time for an item is 60 seconds. If another command is not executed within 60 seconds, the display goes blank.

Traffic Data

The system records time in CCS (hundred call seconds) where 1 call lasting 100 seconds equals 1 CCS. The FADS feature collects and compiles information on an hourly basis.

While this feature is intended primarily to measure traffic data for CAS and UCD group calls, the FADS feature also measures regular traffic data for attendant-related calls.

CAS Traffc Data

The traffic data for CAS calls include:

- Number of calls handled by each attendant
- Total time (expressed in CCS) that an attendant is busy processing calls
- Number of calls received from each branch location over release link trunks
- Total calls received from all branch locations
- Total time (expressed in CCS) that each branch location's release link trunk handles calls
- Total cals placed in queue waiting for the attendant to answer
- Total time (expressed in CCS) that all calls remain in the attendant queue
- Total times a caller hangs up while the call is in queue before the attendant answers.

UCD Traffic Data

The traffic data for UCD calls include:

- Number of calls handled by each group member
- Total calls handled by all members of a group
- Total time (expressed in CCS) that a voice terminal group member is busy processing group calls
- Number of incoming calls received over each dedicated UCD trunk group
- Total incoming trunk calls received over all dedicated UCD trunk groups
- Total traffic handled by each dedicated UCD trunk group (expressed in CCS)
- Number of calls placed in queue for a group (every UCD call enters the queue)
- Total time (expressed in CCS) that calls remain in a UCD group queue
- Number of times a caller hangs up while the call is waiting in queue before a UCD group member becomes available.

Considerations

Measurement Limits

The FADS feature only measures the first 12 UCD groups. Forty group members can be measured in each of the first four groups. Twenty-four group members can be measured in groups 5 through 12. The FADS feature does not measure groups or group members in excess of these parameters.

The FADS feature is able to measure Direct Inward Dialing trunks.

Display Units

The maximum number of display units per system is 13: 1 unit for CAS and 1 for each of the 12 UCD groups.

Report Retension

After compiling the hourly traffic reports, the FADS feature retains the reports for 1 hour for inspection and / or printing.

UCD Group Administration

The UCD groups should be administered to FADS channels consecutively (e.g., 1, 2, 3, 4, 5, 6, 7 instead of 1, 3, 4, 7, 9, 11, 12). If not, the Queue Usage Count will not be measured.

Button Response Time

The buttons on the FADS display unit should be held down long enough for the system to recognize that a button has been pressed. This time is usually short enough not to be noticeable. However, during heavy traffic periods, it may be necessary to hold the buttons down slightly longer than dining light traffic.

With the printer turned on, it is necessary to pause briefly between button presses on the display unit. This is because the printer responds more slowly than the display unit Rapid button presses may result in the printer missing a request for service.

Interactions With Other Features

None.

Hardware Requirements

The FADS feature requires the following additional or special hardware.

- A 102F1-A display unit for CAS
- A 102G1-A display unit for each UCD group measured (maximum 12)

- A211A power unit for each display unit used (maximum 13)
- A B-19252, L7 adapter for each display unit used (maximum 13)
- A TN403 low-speed data channel per display unit used (16 channels per TN403)
- A 9042-2 ADDMASTER* printer (optional) for each display unit (maximum 13).

Feature Administration

Assignment of this feature is on a per-system basis when used with CAS. When used with UCD, assignment of the feature is on a per-UCD group basis.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is administered using the DEFINITY Manager II.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — FORCE ADMINISTRATION DATA SYSTEM			
PROCEDURE	WORD	PURPOSE	SMT
253	1	Administers a data channel circuit of a TN403 circuit pack.	No
284	1	Sets the software clock.	Yes
426	1	Administers feature assignments for FADS.	Yes

* Registered trademark of Addmaster Corporation

Foreign Exchange Access

Description

The FX (Foreign Exchange) Access feature provides connectivity to CO (Central Office) trunks from areas outside of the local service area where the System 85 or DEFINITY Generic 2 switch is located. This feature provides the same functionality to the FX CO service area as is provided by local CO trunks in the local dialing area. This includes use of features such as DID (Direct Inward Dialing), DOD (Direct Outward Dialing), LDN (Listed Directory Number), Remote Access, and Personal Central Office Line.

Use of the FX Access feature in effect extends the "free" dialing area to the exchange served by the FX CO. This can represent a significant cost savings where considerable business is done in an area that otherwise would require toll call access.

Feature History and Development

This feature was first available in System 85 with R1. It was included in a feature called Public Network Access and was reintroduced in Release 2, Version 3 under the name Foreign Exchange Access. It has otherwise remained unchanged since its introduction.

User Operations

The following are the user operating procedures for this feature.

Attendant Handling of Incoming Calls

1. Press the associated switch loop button,

   or

   Press the **[ANSWER]** button. [PA lamp goes out.] (Calling party is connected to attendant.)

2. Obtain the desired destination for the incoming call.
3. Press the **[START]** button. [Dial tone]
4. Press the appropriate DXS button

   or

   Dial the desired extension number. [Call-progress tone, RING lamp lights.]

5. Press the **[RELEASE]** button. [PA lamp lights.]

Attendant Handling of Outgoing Calls

If ADTGS (Attendant Direct Trunk Group Selection) is available:

Press the appropriate FX trunk group button.

If ADTGS is not available:

1. Press the **[START]** button. [Dial tone]
2. Dial the FX trunk group access code.

Depending on calling instructions:

3. Press the **[HOLD]** button to remain connected to the call and finish dialing the final destination number (or set up a conference if desired),

   or

   Press the **[RELEASE]** button to drop from the call and allow the calling party to finish dialing.

Direct Outward Dialing FX Access Calls

1. Go off-hook. [Dial tone]
2. Dial the FX trunk group access code. [Dial tone]
3. Dial the desired destination number. [Call-progress tone]

Considerations

Tariff Concepts

FX Access can be a cost-effective alternative to DDD (Direct Distance Dialing) or WATS (Wide Area Telecommunications Service) when a large volume of long-distance traffic is placed to a distant metropolitian area. The following list shows the components of FX billing.

- Flat-Rate Billing

  Each FX trunk is billed at a flat rate, regardless of usage.

- Flat Rate Based on Distance

  The cost per trunk is a function of the distance from the closed end (originating switch) to the open end (CO in the distant exchange).

- Small Usage-Sensitive Component

  There can also be a "usage sensitive" component in the price of FX Service. This component, when applied, is affixed at the open end of the trunk by the distant LEC (Local Exchange Carrier). This cost is usually only a small percentage of the flat-rate cost of acquiring each trunk.

Access Code

On System 85 and DEFINITY Generic 2.1 switches, the AAR (Automatic Alternate Routing) and ARS (Automatic Route Selection) features can be used to provide FX service. When this is done, the AAR or ARS access code is used and the FX trunk group dial access code does not need to be used. For DEFINITY Generic 2.2, the AAR and ARS features are replaced by the WCR (World Class Routing) feature. With WCR, a network access code is again needed, however, depending on administration it can but does not need to be exclusively for FX routing.

Hard and Soft Processor Swaps

Stable calls over FX trunk groups endure a hard processor swap. However, calls cannot be placed over FX trunk groups during a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

The Abbreviated Dialing feature is fully compatible with the FX Access feature. That is, the FX Access code or a full FX dialing string can be loaded into an Abbreviated Dialing list location or AD button. When this is done, the Abbreviated Dialing feature can be used in invoke FX Access.

AAR (Automatic Alternate Routing)

On System 85 and DEFINITY Generic 2.1 switches, the AAR feature is fully compatible with FX Access. FX trunk groups can be assigned to AAR patterns and preferences. When this is done, the AAR feature selects the FX trunk group when appropriate and the FX trunk group dial access code does not need to be used.

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, the ARS feature is fully compatible with FX Access. FX trunk groups can be assigned to ARS patterns and preferences. When this is done, the ARS feature selects the FX trunk group when appropriate and the FX trunk group dial access code does not need to be used.

DID (Direct Inward Dialing)

Using incoming (or 2-way) FX trunks, the DID feature can be used by a caller in the FX CO service area to reach a specific station on the switch.

DOD (Direct Outward Dialing)

With outgoing (or 2-way) trunks, the DOD feature is used to call stations served by the FX CO over FX Access trunks.

DS1 (Digital Service 1) Interface

The DS1 Interface feature is compatible with FX Access service. Current network tariffs require that if a DS1 span is used for FX service it must be on a dedicated basis. That is, no trunks (channels) on the same DS1 span can be used for any other purpose.

ISDN—PRI (Primary Rate Interface)

The FX feature is compatible with the ISDN—PRI feature. However, FX service is not currently offered on a Call-By-Call Service Selection basis (see the ISDN—PRI feature chapter). This means that an ISDN span can provide FX service on an all or nothing basis only. This is a constraint of the network and not of the System 85 or DEFINITY Generic 2 switch or ISDN. The System 85, Release 2, Version 4 and DEFINITY Generic 2 switches will support FX Service on a Call-By-Call Service Selection basis when it is available from the network.

Look-Ahead Interflow

From a sending (or tandeming) switch, calls can interflow over the public network using ISDN—PRI trunk groups assigned as FX Trunk Type 22 or 24. When this is done, these interflow calls will succeed if every public-network switch between the sending switch and the destination telephone number is ISDN—PRI-capable.

At a sending switch, incoming FX Access calls can terminate to a VDN with a vector assigned that contains commands for Look-Ahead Interflow. When this is done, these incoming FX Access calls will interflow normally (i.e., according to the commands in the sending and receiving vectors).

Multiple LDN (Listed Directory Numbers)

The Multiple LDN feature can be used to direct incoming FX Access calls to the attendant queue.

Remote Access

The FX Access features is fully compatible with the Remote Access feature. That is, based on restrictions applied, incoming FX Access calls can use the Remote Access feature and Remote Access calls to the switch can use outgoing FX Access trunks.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, the WCR feature replaces the networking features AAR and ARS. The FX feature works with WCR in the same way it did with the earlier networking features. If FX trunks are included in WCR routing patterns as preferences, they can be accessed automatically through the WCR feature rather than by dialing the FX trunk group dial access code.

Restricting Feature Use

The Restriction—Voice Terminal Restrictions and the Attendant Control of Voice Terminals features can be used to restrict individual station access to FX Access trunks.

The Miscellaneous Trunk Restrictions and Code Restriction features can be used to restrict access to FX Access trunks on a more generalized basis.

If FX Access is via AAR, ARS, or WCR, the FRL (Facilities Restriction Level) and Authorization Code features can be used to restrict access to these trunks.

Hardware Requirements

The FX Access feature uses the following specific hardware.

For Traditional Modules:

- SNQ230, Trunk Circuit Pack (four circuits per circuit pack).

For Universal Modules:

- TN747B, Trunk Circuit Pack (eight circuits per circuit pack).

Feature Administration

Assignment of the FX Access feature is on a per-system and per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) and the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — FOREIGN EXCHANGE ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
010	3	Assigns trunk restriction group and FRL to an extension class of service.	Yes
100	1	Assigns trunk group dial access code, and trunk type. For System 85, R2 V3 and earlier, also assigns Route Advance patterns. The applicable trunk-types are as follows: 21 1-way automatic incoming attendant-completing 22 1-way outgoing DOD 23 1-way out DOD with party test 24 2-way automatic attendant-completing in / DOD 25 2-way with party test.	No
101	1	Assigns trunk group features.	No
102	1	Assigns Miscellaneous Trunk Restrictions (if desired).	Yes
103	1	Assigns trunk group FRL.	Yes*
115	1	Assigns incoming termination of a trunk group (if desired).	No
150	1	Administers trunk group number and unattended console service to the equipment location.	No
202	1	Assigns trunk group to Attendant Trunk Group Select buttons and trunk group warning levels.	No
204	1	Administers the ICI (Incoming Call Identification) for the Attendant Console.	No
* Display only procedure for the SMT.			

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — FOREIGN EXCHANGE ACCESS	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns miscellaneous trunk restrictions and FRL to an extension class of service.

Hold

Description

This feature allows a voice terminal user to hold incoming outgoing, or intraoffice calls. This frees the same extension number to originate another call, activate a feature, or return to a previously held call. This last capability also allows a user to alternate between two calls, holding one while connected to the other. Additionally, an existing call can be placed on hold to prevent the held party from overhearing office discussions.

Types of Hold

A call is held differently on different types of voice terminals.

Soft Hold

Users of single-appearance (analog) voice terminals (2500, 7101A, and 7103A) can place a call into soft hold by pressing the RECALL (R) button or by momentarily pressing (flashing) the switchhook. Soft hold is a transient call state with limited functionality. While a call is in soft hold, the user can either access *a limited set* of System 85 or Generic 2 features, or place the call into "hard hold." If the user goes on-hook, the soft held call is lost.

Hard Hold

The hard hold function places the call into a more durable hold state than soft hold. Hard hold can be activated using an access code or a features button. The 7103A voice terminal may be assigned a HOLD button that dials the access code automatically. All AT&T multi-appearance voice terminals have Hold Buttons that activate the hard hold function. With hard hold active, the user can access more features than are allowed with soft hold. The call remains in hard hold until the access code is redialed or until the user goes on-hook.

Feature History and Development

This feature was first available on System 85 in Release 1.

During the Release 2, Version 2 time frame (but not initially in Version 2), the ability to use the Call Hold access code was added. This enhancement was retrofitted back to Release 1.

User Operations

The following are the user operating procedures for this feature.

To Hold a Call

At a 2500 voice terminal without an R button:

1. Be sure there is a 2-way connection.
2. Press the switchhook for about half a second (from 0.2 seconds to1.2 seconds). [Recall dial tone, the connected call is placed in the soft hold state.]
3. Dial the Call Hold access code. Dial tone, the held call is placed in the hard hold state.]
4. Perform other desired actions.

At a 2500 with an R button, 7101A, or 7103A voice terminal:

1. Be sure there is a 2-way connection.
2. Press **[RECALL]** or **[R]** . [Recall dial tone, the connected call is placed in the soft hold state.]
3. Dial the Call Hold access code

   or

   Press **[HOLD]** on a programmed 7103A terminal. [Dial tone, the held call is placed in the hard hold state.]
4. Perform other desired actions.

At a multiappearance voice terminal:

1. Be sure there is a 2-way connection.
2. Press **[HOLD]** . [The green status lamp on the held apperance flutters, the connected call is placed directly into the hard hold state.]
3. Perform other desired actions.

To Return to the Previously Held Call

At a 2500, 7101A, or 7103A voice terminal:

Go on-hook. [Switch rings back the analog voice terminal with the held call.]

At a 2500 voice terminal without an R button:

1. Momentarily press the switchhook. [Dial tone]
2. Dial the Call Hold access code. [Switch re-connects the held call.]

At a 2500 with an R button, 7101A, or 7103A voice terminal:

1. Press **[RECALL]** . [Dial tone]
2. Dial the Call Hold access code. [Switch re-connects the held call.]

At a multiappearance voice terminal:

Press the appearance button of the held call. [Switch re-connects the held call.]

Considerations

Attendant Calls

Attendant calls cannot be placed on hold by a voice terminal user.

Trunk Restrictions

The Hold feature cannot be used by a voice terminal when connected to an attendant console, a recorded announcement, a recorded telephone dictation trucnk, or a loudspeaker paging trunk.

Reconnection to Held Calls

A single-appearance terminal user may hold only one call at a time. If the user completes the conversation, flashes the switchhook or presses RECALL, then dials the Call Hold access code, the user is reconnected with the held call. If the user goes on-hook after completing the conversation, the switch attempts to ringback the user with the held call. This reduces the possibility of losing held calls. However, queued or waiting calls may be serviced before the held call. Using the Call Hold access code ensures being reconnected to the held call.

Alternating Between Two Calls

A user of a single-appearance terminal can alternate between two calls, holding one while speaking to the other. All three parties may not be placed on the same talking connection. The two held calls cannot be connected together.

A user of a multiappearance voice terminal can also alternate between two calls. Moreover, this user can place all three parties on the same talking connection by placing one of the parties on hold using the CONFERENCE button, manually selecting the appearance with the other held party, and pressing the CONFERENCE button again. (See the Conference—Three Party feature for a description of the user operation for conferencing.)

Deactivation of Hold

This feature is deactivated when the held call is retrieved and no other call is present. Also, the feature is deactivated if the held party abandons the call by going on-hook. A new call must be placed to contact the party.

RECALL Button and Switchhook Functions

A switchhook should only be pressed for 200 milliseconds to 1200 milliseconds. A switchhook pressed for more time will cause a disconnect. A switchhook pressed for less time is ignored by the switch. The RECALL button on a single-appearance terminal generates a timed flash of the correct time interval.

Hard and Soft Processor Swaps

If a call is being held on a voice terminal when a hard processor swap occurs, the held call does not endure the hard swap.

During a hard processor swap, calls cannot be placed on hold.

The Hold feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

ACD (Automatic Call Distribution)

If an ACD agent (including Message Center agents) with queue-status display places an ACD call on hold, then the queue-status display is automatically updated as the agent returns to the held call for subsequent handling.

When multiple call handling is assigned to an ACD split, ACD agents equipped with a multiappearance voice terminals can place ACD calls on hold and remain available for additional ACD calls. (This operation is appropriate for Message Center agents.)

Attendant Call Waiting

Attendant Call Waiting is denied when the called single-appearance voice terminal has a call on hold.

Automatic Callback

The Hold feature interacts with the Automatic Callback feature as follows:

- A voice terminal user can hold a call while Automatic Callback is in effect toward the terminal.
- If a calling single-appearance terminal user has activated Automatic Callback and has a party on hold, the held call takes priority (rings back the calling voice terminal) when the calling terminal user hangs up. When there are no calls in hold and both terminals become idle, the automatic callback sequence takes place.
- For calling multiappearance voice terminals with a call on hold, the callback routes to an idle appearance (if available).

Bridged Call

While a straight line set has a call on soft hold (by pressing the RECALL button, or momentarily pressing the switchhook) bridging is not allowed for a multiappearance terminal sharing the appearance.

While a straight line set has a call on hard hold (by dialing the Hold dial access code), bridging is allowed for a multiappearance voice terminal sharing the appearance. The bridging multiappearance voice terminal is added to the active connection. However, the held call cannot be retrieved until the bridged multiappearance terminal leaves the connection.

Busy Verification of Lines

If terminal A has terminal B on hold, terminal A can be busy verified but terminal B cannot (unless the extension being verified has multiple appearances).

Call Coverage

If a covering user places a Coverage call on hold and the principal then bridges onto the held call, the covering user who originally placed the call on hold drops off and cannot reenter the connection.

Call Forwarding—Busy and Don't Answer

Hard hold ringbacks for single-appearance voice terminals are not forwarded. Instead, the held call rings back the terminal that placed the call on hold when the called terminal goes on-hook.

Call Forwarding—Don't Answer

Hard hold ringbacks for single-appearance voice terminals are not forwarded. Instead, the held call rings back the terminal that placed the call on hold when the called terminal goes on-hook.

Call Forwarding—Follow Me

Hard hold ringbacks for single-appearance voice terminals are not forwarded. Instead, the held call rings back the terminal that placed the call on hold when the called terminal goes on-hook.

Call Pickup

A single-appearance voice terminal user is denied use of the Call Pickup feature while holding a call on hard hold and soft hold at the same time.

A voice terminal user is allowed to place a call on hard hold, and then answer a call using Call Pickup. If this is done using soft hold, the held call is moved to hard hold and can be retrieved using the access code.

Call Waiting

A voice terminal with a call on hold and a call in waiting goes on-hook. The waiting call is connected first. Using the Call Hold access code is the recommended method for returning to a held call.

The switch denies Call Waiting toward a voice terminal that has been placed on hold.

Call Waiting is denied if the calling voice terminal has a call on soft hold, but is allowed if the calling terminal has a call on hard hold.

Conference—Attendant Five Party

Voice terminal users who are participating in an attendant conference cannot use the Hold feature.

Conference—Attendant Six Party

An attendant can establish a multiparty conference connection of up to six conferees in addition to the attendant. Voice terminals which are participating in an attendant conference connection are denied access to the Hold feature.

Conference—Three Party

Used in conjunction with hard hold, this feature allows the calling voice terminal user to transfer a second call while holding the first call. After transferring the second call, the calling voice terminal user flashes the switchhook and dials the Call Hold access code to return to the call in hold.

Intercom—Automatic, Intercom—Dial, and Intercom—Manual

The Hold feature is denied on an intercom line.

Last Number Dialed

The LND (Last Number Dialed) feature operates while a call is on "hard hold" or "soft hold." The number of the last call (the call placed after the previous call was put on hold) is stored in LND memory.

Multiappearance Preselection and Preference

The Hold feature interacts with the Multiappearance Preselection and Preference feature as follows:

- If the HOLD button on the terminal is pressed while the terminal is on-hook, any appearance preference for call origination (i.e., Last Appearance Preference, Prime Appearance Preference, or Manual Preselection) is lost. When going off-hook, the user does not connect to any appearance in these cases. Idle Appearance Preference will select a line when going off-hook.
- If any appearance is placed on hold and the controlling terminal user goes on-hook, the I-use lamp is relit on the Prime Appearance button if that terminal has Prime

Appearance Preference. If the controlling terminal user then goes off-hook, the terminal connects back to the Prime Appearance and that call releases from hold.

- If the HOLD button is pressed while the terminal is in the on-hook ringing state and the Ringing Appearance preference is active, the Ringing Appearance Preference feature becomes inoperative. When the user goes off-hook, no appearance is selected or an idle line is selected if the voice terminal has Idle Appearance preference.

Music-on-Hold Access

If a 3-party conference call is placed on hold, the two remaining parties can continue with the call and are not provided music even if Music-on-Hold Access is active. A single party on hold is provided with music.

Override

The Hold feature interacts with the Override feature as follows:

- Override is allowed toward a voice terminal that has a call on hold.
- Override of a voice terminal's appearance that is on hold is denied. Busy tone is heard.

Personal Central Office Line

Personal Central Office Line calls can be placed on hold.

Priority Calling

Priority Calling is denied when either soft hold or hard hold is active at a called single-appearance voice terminal.

Priority Calling to a busy terminal is also denied if the calling voice terminal has a call on soft hold. However, Priority Calling is allowed when the called terminal is idle or when the calling terminal has a call on hard hold.

Privacy—Manual Exclusion

An appearance that is placed on hold while Manual Exclusion is active remains in Manual Exclusion.

Queuing

A voice terminal user can be placed in a ringback queue while holding another call. Queuing allows a voice terminal user to dial a busy outgoing trunk group, be automatically placed in a queue and called back when a trunk is available. If queuing is activated and the calling voice terminal has a call on hold, going on-hook returns the held call first (Hold takes precedence over queuing callback). After the calling voice terminal user goes on-hook and the dialed trunk group is idle, the calling voice terminal is rung back.

Restriction—Voice Terminal Restrictions

The user of an origination-restricted single-appearance voice terminal can place a calling party on hold. From this state, the restricted voice terminal can be used to originate a call, activate a feature, or return to the held call.

When the origination-restricted user places a call on hold, the voice terminal is treated as a ***fully unrestricted terminal*** unless another restriction (e.g., Outward Restriction or Terminal-to-Terminal Only Calling) has also been applied to the terminal. If another restriction does apply, the origination restricted user's dialing capabilities are limited to the capabilities allowed by the second restriction.

Trunk Verification—Attendant and Voice Terminal

If the trunk to be verified is being held by the Hold feature, Trunk Verification is denied.

Uniform Call Distribution

Going on-hook to return to a held call may result in the voice terminal being connected to the next call in the distribution-group queue. Using the Call Hold access code is the recommended method for returning to a held call.

Restricting Feature Use

A voice terminal user cannot be restricted from using the Hold feature when the feature is assigned in the applicable line class of service.

There is no limit to the number of voice terminals which can be assigned to this feature.

Hardware Requirements

None.

Feature Administration

Assignment of the Hold feature is on the line class-of-service and then on a per-line basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal).

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — HOLD			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the class of service to an extension number.	Yes
010	1	Assigns the Hold feature to a voice terminal class of service.	Yes
051	1	Assigns a voice terminal and related translations including the HOLD button which is automatically assigned.	Yes
054	3	Displays the button type of the fixed HOLD buttons.	Yes
350	1	Assigns the first digit of the dial access code (if required).	No
350	2	Assigns the access code to access hard hold using a single-appearance voice terminal. The applicable encode is: 4 Call Hold.	No

Notes:

Host Computer Access

Description

The Host Computer Access feature provides a DCP (Digital Communications Protocol) interface between a System 85 or Generic 2 switch and a local Host Computer. This permits switched digital access between data endpoints on the local switch and the host computer.

With the Host Computer Access feature, the host is connected to a DCP port on the switch through a Data Module, or the connection can be to an EIA or Digital Line port circuit via a MADU (Multiple Asynchronous Data Unit). The data modules or MADU convert the EIA RS-232 signals from the host computer to the DCP format of the switch and vice versa. Both formats are digital, but the communications and signaling arrangements differ. Another digital host access arrangement is provided by the DMI (Digital Multiplexed Interface) feature. The DMI feature is described separately in this manual. Figure 61–1 shows a Host Computer Access configuration for System 85, Release 2 switches or Generic 2 switches using the traditional module. For universal modules on the Generic 2 switch, the arrangement remains the same except that the SN270 circuit packs change to TN754 circuit packs.

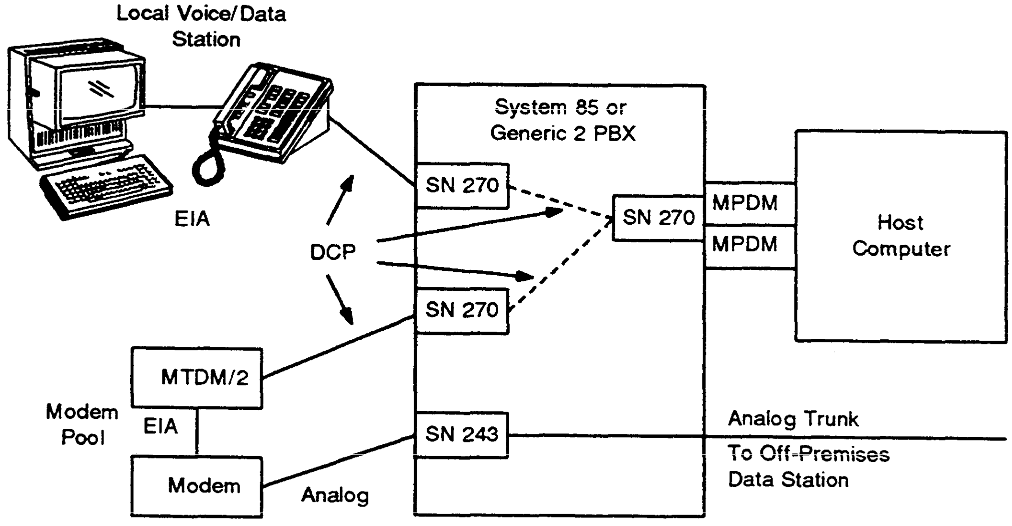


Figure 61-1. Host Computer Access Feature

Feature Access

Host Computer Access ports are accessible from either on- or off-premises facilities. For off-premises facilities, the Extension Number Steering feature is used. For off-premises calls that use analog or voice grade digital trunk facilities, the Modem Pooling feature must also be provided. An on-premises analog terminal must appear as a Remote Access extension or an off-premises terminal in order to use the Modem Pooling feature.

Host Computer Access ports can be set up as either lines or trunks. However, arranging the Host Computer Access ports into a trunk group allows trunk access features to be used which can enhance feature operations (see Considerations).

Reduced Cost Configuration

Figure 61–2 shows a reduced cost configuration for the Host Computer Access feature. For System 85, Release 2 switches and for traditional modules on the Generic 2 switch, this configuration uses SN238 EIA circuit packs and MADUs (Multiple Asynchronous Data Units). With this reduced cost arrangement, two SN238s and one MADU provide an 8-port interface. The same eight ports in the conventional configuration would require two SN270 circuit packs and eight MPDMs.

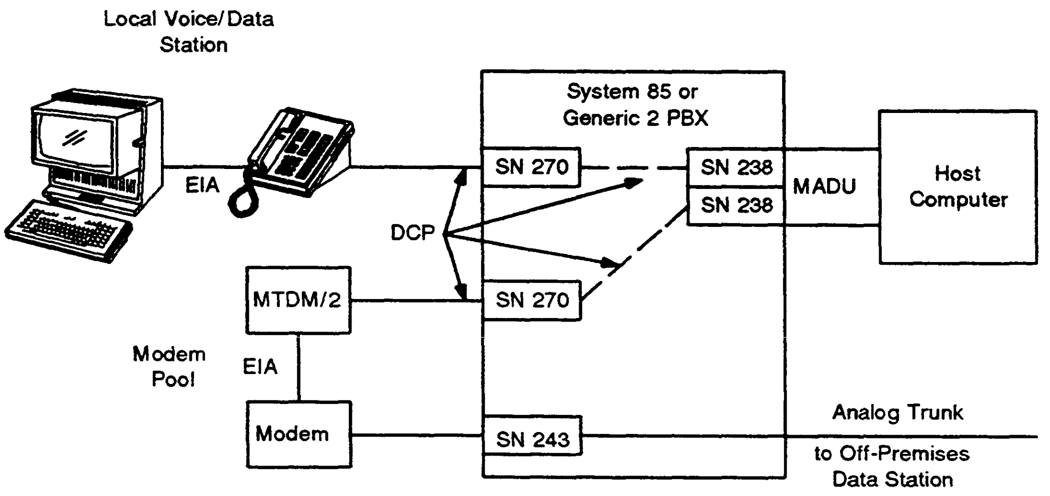


Figure 61-2. Host Computer Access, Reduced Cost Option

For universal modules on the Generic 2 switch, the same reduced cost configuration can be used. The difference is that one TN726 Data Line circuit pack is used instead of the two SN238s, and TN754 Data Line circuit packs are used to replace the SN270s on the terminal side (where the terminal is also located on a universal module). This arrangement is shown in Figure 61–3.

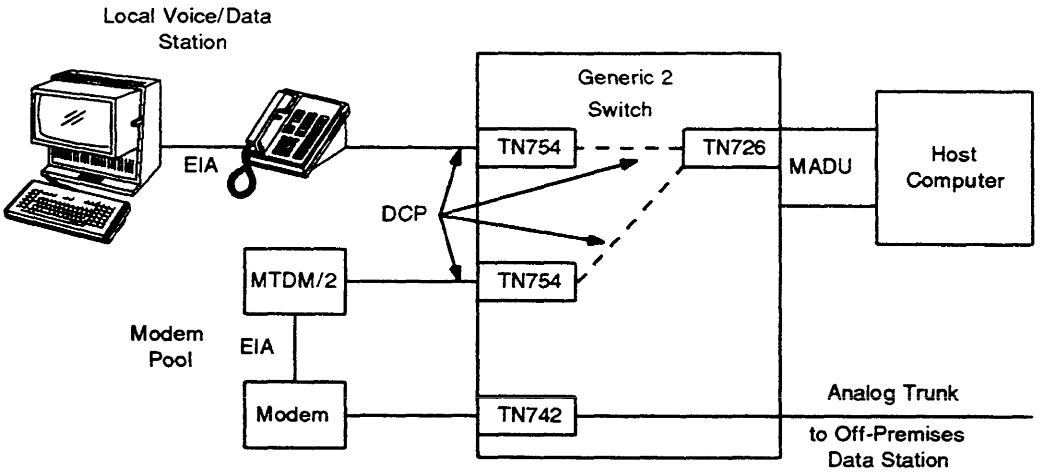


Figure 61-3. Host Computer Access, Reduced Cost Option with Universal Module

Host Computer Automatic Dialing

Host computer automatic dialing permits the host computer to place calls through the switch without the use of an ACU (Automatic Calling Unit). Host software routines can send data terminal (keyboard) dialing signals (ASCII characters in a 10-bit, start/stop format) to the data module. The data module (or MADU) converts outgoing signals to the DCP format and sends them to the switch as though a data terminal were being used It also converts call progress messages from the switch to RS-232 format and returns them to the host. The host routines must be capable of interpreting and responding to these call progress messages. Such routines allow the host to emulate a data terminal to place either on-premises or off-premises calls, thus eliminating the need for ACUs.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to the feature itself since its introduction. New hardware options have been added, specifically

- MADU (Multiple Asynchronous Data Unit) and EIA port for use with the reduced cost conjuration
- Universal module options available with Generic 2
- The 7400A and 7400B data modules have been added.

User Operations

The Host Computer Access feature is operated by the switch. There are no user operations, as such, for this feature. The Data Call Setup feature is used to access this features and the user operations for that feature apply.

Considerations

Attendant Handling

The attendant cannot access the Host Computer Access feature directly. To extend an attendant-seeking call to Host Computer Access, it must first be extended to a voice terminal capable of performing the necessary transfer functions. The voice terminal user can then transfer the call to the host computer.

Data Modules

Any of the following DCP data modules can be used to interface host computers with the System 85 or Generic 2 switch.

- 7400A Data Module

  The 7400A Data Module can be configured in either a DTE (Data Terminal Equipment) interface or a DCE (Data Communications Equipment) interface. The DCE configuration is generally recommended for Host Access purposes. The 7400A should not be configured as a DTE interface for Host Access in situations where calls are expected that will use Modem Pooling. The digital members of Modem Pooling conversions resources are DTE configured. Two DTE configured devices cannot handshake with each other (under normal circumstances). Therefore, Host Access seeking calls that use Modem Pooling cannot successfully complete to a Host Access port that is interfaced with a 7400A (or any other data module) configured as DTE.

  The 7400A Data Module can be set to operate like a MPDM or in the Hayes "Smart Modem" mode. Either configuration will work for Host Access purposes. However, the Hayes "Smart Modem" mode may have some advantages in situations where calling devices (such as Personal Computers) also use the Hayes "Smart Modem" mode, or where the host will be used to originate calls.

- 7400B Data Module

  The 7400B Data Module is configured as a DCE interface only. The 7400B also functions only in the Hayes "Smart Modem" mode. With both of these constraints, there should be no problems using the 7400B Data Module for the Host Computer Access feature.

- MADU (Multiple Asynchronous Data Unit)

  The MADU is used in conjunction with the EIA Port for the Reduced Cost Host Computer Access feature configuration.

- MPDM (Modular Processor Data Module)

  The MPDM is the traditional data module used with the Host Computer Access feature. This data module provides a DCE interface that is compatible for use with Modem Pooling.

- MTDM (Modular Trunk Data Module)

  The MTDM should not be used for Host Access in situations where calls are expected that will use Modem Pooling. THE MTDM uses the DTE configured interface, and digital members of Modem Pooling conversions resources are also DTE configured. Two DTE configured devices cannot handshake with each other under normal circumstances. Therefore, Host Access seeking calls that use Modem Pooling cannot successfully complete to a Host Access port that is interfaced with a MTDM.

The 7500 ISDN—BRI Data Module is also available for use with the Universal Module on the Generic 2 switch. On the Generic 2 switch, the 7500 Data Module can be used in the DCE configuration and therefore could be used for Host Computer Access. However, the ISDN—BRI feature is a line side only feature on Generic 2, therefore, Host Access Ports interfaced with 7500 Data Modules could not be arranged into trunk groups and would not have the full functionality normally expected of the Host Computer Access feature. Therefore, 7500 Data Modules are not recommended for use with the Host Computer Access feature.

Lines vrs Trunks

Host Computer Access ports can be administer as either lines or trunks. However, some severe limitations are encounter when line side service is selected. For example, lines can be arranged into Hunt Groups so that a maximum of 30 lines can be searched for an available port Trunks are arranged into trunk groups with up to 99 trunks in a host access trunk group. Furthermore, these trunk groups can be assigned to Route Advance patterns which provide for up to five trunk groups (with a maximum of 495 trunks) in a single search pattern.

Port (Trunk) Group Limits

Host Computer Access ports, translated on the switch as trunks, are limited to a maximum of 99 ports (trunks) in a trunk group. This limitation remains in force for host computer access trunk groups, even on switches that have a higher basic trunk group limit.

The System 85, Release 2, is limited to 255 trunk groups in Version 3 and earlier switches. The System 85, Release 2, Version 4 and the Generic 2 switches can have 999 trunk groups. System 85 is also limited to a total of 2250 physical trunks in Version 1, 5000 trunks in Version 2, and 6000 trunks in Version 3 and 4. The Generic 2 can also have 6000 physical trunks. These facilities (both trunks and trunk groups) must be shared by all the features using trunk appearances (see Appendix A for applicable features and limits).

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

The Abbreviated Dialing feature is fully compatible with the Host Computer Access feature. Abbreviated Dialing is used by the Data Call Setup (Default Dialing) and the Hotline features to automatically place calls to Host Computer Access ports.

Bearer Capability

In Generic 2, all HCA ports must be assigned a BCCOS (Bearer Capability Class of Service). If the predefined BCCOS 1 (the default) is used, call processing to HCA ports will be essentially the same as it was in System 85, R2 V4 and earlier, except that when ***Voice Terminal Data Call Setup*** is used from a DCP terminal, ***Data Preindication is mandatory.*** If some other BCCOS is used (a user defined BCCOS that specifies application unique data call characteristics for the host port supported), then call processing to (and possibly from) that specific port is controlled by the data characteristics assigned to that BCCOS. For example, you could set up a BCCOS that would allow only synchronous calls or only mode 3 data calls to be completed to a particular group of host ports. See the Bearer Capability feature for detailed information on customizing BCCOSs.

CDR (Call Detail Recording)

The CDR feature provides information on and specific identification of data calls, including Host Computer Access calls. This enables managers to allocate resources to meet dynamic needs based on use.

Data Call Setup

The Data Call Setup feature is fully compatible with the Host Computer Access feature.

Data Protection

Data Protection-Permanent should be assigned to all Host Computer Access trunk groups to prevent system generated tones from interrupting data communications. These tones are issued in response to the activation of certain features such as Override and Priority Calling. These features are not to be used toward data extensions, but Data Protection prevents inadvertent interruptions.

Hunting

The Hunting feature is used to check if an alternate circuit in a Hunt Group can serve a call when the circuit dialed is busy. This feature can be used for Host Access if port circuits are administered as lines. However, Host Computer Access ports are usually administered as trunks. The hunting principle is also used for trunks that are assigned to the same trunk group. This hunting service is automatically provided for trunk circuits in the same trunk group. When dialing a Host Computer Access trunk group, access is provided to as many as 99 computer ports. Trunk groups can also be combined using Route Advance to provide additional access.

ISN (Information Systems Network) Interface

The Host Computer Access feature is available to ISN stations through the ISN Interface feature.

ISDN—BRI (Basic Rate Interface)

The ISDN—BRI feature is compatible with the Host Computer Access feature in that callers using an ISDN—BRI data station can access HCA ports like DCP data station users. However, ISDN—BRI is a line side only feature on the Generic 2 switch. While host ports can be set up using lines, this would result in a loss of functionality and is not recommended. Therefore, ISDN—BRI data modules (7500 Data Modules) are not recommended for use with the Host Computer Access feature.

Last Number Dialed

The Last Number Dialed feature completely stores and redials HCA calls to a host computer.

Modem Pooling

The Modem Pooling feature is fully compatible with the Host Computer Access feature. However, one administrative consideration is worthy of note. The digital member of a Modem Pooling conversion resource is always configured as DTE. The data module used for Host Computer Access may be configured as DTE or DCE (generally the DCE configuration is recommended). An MTDM, if used for Host Access, is always configured as DTE while a 7400A Data Module can be either DTE or DCE. Two DTE configured devices cannot (normally) handshake with each other. This means that if the data module used for Host Access interface is configured as DTE, incoming (or outgoing) calls that must use Modem Pooling cannot complete successfully to that Host Access port.

Queuing

The Queuing feature provides a waiting list when all resources are busy. As resources become available, the first user in queue is served. Queuing can be used with Host Computer Access. Voice terminal users can be provided with on-hook or off-hook queuing. Data terminal users can be provided off-hook queuing. If every Host Computer Access port is busy, users dialing the Host Computer Access access code receive confirmation tone (on-hook queuing), music-on-hold, a recorded announcement, or silence (off-hook queuing). If reorder tone is received, it may be an indication that the queue is full. It might also mean the system is in heavy use. The user should go on-hook and try again.

Route Advance

The Route Advance feature provides access to up to five trunk groups using a single access code. When every circuit in the first trunk group is busy, the system checks the next trunk group, and so on. Since Host Computer Access trunk groups can contain up to 99 trunks, Route Advance can provide access to as many as 495 computer ports with a single access code or extension number.

Tenant Services

Host Computer Access (DCP access to a host computer) is a partitioned feature on System 85 and Generic 2.

Line-side computer access is partitioned using partitioned extension numbers. Each extension number is assigned to a data module and is usually included in a hunt group. In turn, the data module's extension number is assigned to an extension partition allowing data-terminal access for users in that partition and Extension Partition 0.

Trunk-side computer access is partition using partitioned trunk groups. Since the trunk types (103 and 104) can be partitioned, access to these trunk groups can be dedicated to or shared by an extension partition. However, if a computer is accessed from outside the switch, ***trunk-side*** partitioning would have no effect. There are no partitioning checks between the incoming trunk group and the outgoing HCA trunk group.

Restricting Feature Use

Attendant Control of Trunk Group Access

Calls to Host Computer Access ports can be restricted by applying Attendant Control of Trunk Group Access to the Host Computer Access trunk groups. Any attempt to access a Host Computer Access port is redirected to an attendant. If the attendant is to screen these calls, a voice terminal equipped with transfer capabilities must be provided near the attendant console to perform the transfer. Data call transfers cannot be performed from the attendant console because of the DCP interface at the Host Computer Access port. Modem Pooling is not provided for attendant-extended calls.

Attendant Control of Voice Terminals

An attendant can restrict selected terminals from access to the Host Computer Access trunk groups with the Attendant Control of Voice Terminals feature. An attempt to access a Host Computer Access port from a restricted terminal is redirected to an attendant. If the attendant is to screen these calls, a voice terminal equipped with transfer capabilities must be provided near the attendant console to perform the transfer. The transfer cannot be performed from the attendant console.

FRL (Facilities Restriction Levels)

Facilities Restriction Levels provide a hierarchy of access to trunk groups. Each Host Computer Access trunk group is assigned an FRL to restrict certain users from access to the host computer. An attendant or the system manager (user of Applications Processor System Management applications) can invoke an alternate set of FRLs when desired.

Miscellaneous Trunk Restrictions

Local voice terminal lines may be restricted from accessing a Host Computer Access trunk by the Miscellaneous Trunk Restriction feature.

Trunk Group Restrictions

Access to Host Computer Access ports via a trunk-group dial access code can be restricted by activating "dial access denied" for the trunk group. This makes Host Computer Access possible only via the Extension Number Steering feature which uses indirect access to the Host Computer Access ports via a previously unused extension number.

Hardware Requirements

The following specific hardware items are required for the Host Computer Access feature.

Standard Configuration

For Traditional Modules:

- SNZ70, GPP (General Purpose Port) Circuit Pack

  One SN270 circuit pack provides four DCP circuits.

For Universal Modules:

- TN754, Digital Line Circuit Pack

  One TN754 circuit pack provides eight DCP circuits.

Regardless of the Module Type:

- Data Module

  A data module is used to interface the host computer to the SN270 or TN754. The 7400A Data Module in the DCE configuration, the 7400B Data Module, or the MPDM (Modular Processor Data Module) provide the recommended interface for a conventional Host Computer Access arrangement.

  The MTDM always uses a DTE configured interface and the 7400A can be configured as DTE. Ports using a DTE interface cannot connect to other data endpoints that also use a DTE interface. This includes the Modem Pooling feature and other Host Computer ports (multiple processor applications).

Reduced Cost Configuration

For Traditional Modules:

- SN238, EIA Trunk Circuit Pack

  One SN238 circuit pack provides four digital circuits. The SN238 provides the digital interface to the MADU (Multiple Asynchronous Data Unit).

For Universal Modules:

- TN726, Data Line Circuit Pack

  One TN726 circuit pack provides eight digital circuits. The TN726 provides the digital interface to the MADU (Multiple Asynchronous Data Unit).

Regardless of the Module Type:

- MADU (Multiple Asynchronous Data Unit)

  One MADU provides eight interface circuits (connects to two SN238 EIA circuit packs or one TN726 circuit pack).

Feature Administration

Assignment of the Host Computer Access feature is on a system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using SMT (System Management Terminal), TCM (Terminal Change Management) feature, or FM (Facilities Management) feature.

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — HOST COMPUTER ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
051	1	Assigns an ADFTC to an equipment location for data trunk testing.	Yes
100	1-5	Assigns trunk type including Host Computer Access and Access Code, Route Advance, data characteristics, and dial access restriction. **NOTE:** Applicable Words vary with different versions. The applicable trunk-type encodes include: 103 Host access (PDM) 104 Host access (TDM) 105 3BAP DCPI 106 EIA 4 port 107 ISN/EIA port.	No
101	1	Assigns trunk group characteristics.	No
102	1	Assigns miscellaneous trunk restrictions.	Yes
150	1	Assigns trunk circuits to trunk groups.	No
350	1 & 2	Assigns feature dial access codes (as required).	No

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — HOST COMPUTER ACCESS	
PATH NAME	**PURPOSE**
terminal-change group trunk-restrictions	Assigns miscellaneous trunk restrictions.

The following are the applicable FM path names used with the AP 16.

FM SCREENS — HOST COMPUTER ACCESS	
PATH NAME	**PURPOSE**
facilities-mgmt data-switching host-access	Assigns an equipment location (SN270 circuit) to a Host Computer Access trunk group.
facilities-mgmt data-switching trunk-attributes	Assigns the attributes for a Host Computer Access trunk group. These attributes are: ● The dial access code and trunk group number ● The extension number and equipment location ● A route advance list ● The transmission characteristics ● The trunk type ● Turn the Data Restriction feature on or off.
facilities-mgmt facility-restriction trunk-groups	Assigns FRLs to trunk groups.
facilities-mgrnt facility-restriction alternate-set	Assigns the alternate set of FRLs, and activate/deactivate them when desired. The attendant can also activate/deactivate the Alternate FRLs.
facilities-mgmt trunk-group queuing	Assigns Queuing to trunk groups.

Notes:

Hot Line

Description

The Hot Line feature provides automatic dialing of a predesignated number by going off-hook. Either single-line analog voice terminals or data modules administered for default dialing can be assigned as Hot Line stations.

Hot Line Voice Service

An analog voice terminal is used for Hot Line voice service. The voice terminal is assigned a personal abbreviated dialing list (see the Abbreviated Dialing feature). When administered as a Hot Line station, the first entry in the "A" list is dialed automatically when the terminal goes off-hook. The Hot Line number can have up to 20 digits. This provides Hot-Line service to either on- or off-premises destinations. Any other entries in the list are ignored with this arrangement, and it is not generally possible to activate other calling features and services from a Hot Line station. Unless separately restricted, Hot Line stations can receive incoming calls normally.

Hot Line Data Call Service

The Hot Line feature can be used for data calling as well as for voice service. In System 85, Release 2, Version 3, Hot-Line service was limited to analog service only. Beginning with Release 2, Version 4, either analog or digital data stations can be used with the Hot Line feature.

Analog Data Call Hot Line Stations

For analog data call setup, the Hot Line feature works the same as for an Analog Voice Hot Line stations. A personal abbreviated dialing list is assigned to the analog voice terminal associated with a modem. The first number in the "A" list is automatically dialed when the voice terminal goes off-hook. When ready tone is returned, call control is transferred to the modem in the normal manner (see the Data Call Setup feature, analog data calls).

The voice extension can be used to answer incoming calls. This allows the voice terminal to receive data calls from data end points other than the one assigned as the Hot Line destination. The voice terminal is restricted to one data endpoint for call placement only.

Digital Data Call Hot Line Stations

A data module, providing a DCP (Digital Communications Protocol) interface, and administered for default dialing can be assigned as a Hot Line station. The ***default dialing*** capability of the Data Call Setup feature is used for digital Hot Line call setup. A data module with default dialing administered becomes a digital data Hot Line station through administration (Procedure 000, Word 3). Once assigned, the user simply presses the BREAK key on the keyboard or the Originate / Disconnect button on the data module, and the default number is automatically dialed by the switch.

If a DATA button is assigned anywhere in the system for a digital data Hot Line station, the DATA button can be used to bypass Hot-Line service for the data terminal. With a DATA button, the One Button Transfer or Voice Terminal Dialing functions of the Data Call Setup feature can be used to receive incoming data calls for the data Hot Line station or to initiate data calls for the Hot Line data terminal to a data end point other than the Hot Line destination.

Hot Line Service Applications

There are many potential applications for Hot-Line service. The following are some of the more typical applications in current use.

Manual Line to Attendant

A Manual Line connects the user directly to the attendant when going off-hook. Hotels and airports use this arrangement for voice terminals located in the lobby or passenger terminal. A Manual Line can be set up in System 85 or DEFINITY Generic 2 by administering a Hot Line with the attendant access code as the Hot Line destination code. The attendant can be local or a CAS (Centralized Attendant Service) attendant as best suits the application.

Paging Service

The Hot Line destination code can also be the paging access code. When the ***paging Hot Line terminal*** is taken off-hook, the user is automatically connected with the paging service trunk.

Emergency Services

Hot-Line service is frequently used to provide automatic connections to emergency services such as security services fire reporting and medical facilities.

Dedicated Use Data Terminals

Hot Line data call stations can be set up to ensure that a particular data terminal is always used to access a specific application point in a data system. This use can apply to reservation services and automatic ordering stations.

Hot Line Service Versus Dedicated Switch Connections

The Dedicated Switch Connections feature can be used for many applications where Hot Line Service would be used. The merits of the Hot Line feature versus the Dedicated Switch Connections feature must be determined on an individual case basis.

- The ***Dedicated Switch Connections*** feature ensures that a connection is always available. In cases where emergency response is required or where immediate (real time) recording is needed, this may be an advantage. No time is lost in call setup and processing, and there is no possibility that the distant terminal might be busy on another call or that the switch might be blocked (overloaded).
- The ***Dedicated Switch Connections*** feature ties up a time slot on the switch and the terminals at both ends of the connection, even when not in use. For many low density requirements, this is wasteful of valuable resources.
- The ***Dedicated Switch Connections*** feature is not generally available for DCP (Digital Communications Protocol) interfaced data terminals.

- The ***Hot Line*** feature can be used for either analog or DCP interfaced data terminals. If DCP is used for all other applications, this could avoid the introduction of different, and generally incompatible data interfaces for limited use applications.
- The ***Hot Line*** feature consumes a time slot only when it is actually in use. This reduces the likelihood of blockage on high usage switches.
- The ***Hot Line*** feature also makes it possible to use the terminals at each end of a Hot Line path for other purposes. That is, unless otherwise restricted, a Hot Line terminal can receive incoming calls from stations other the Hot Line destination station.
- The ***Hot Line*** feature can also be used to direct calls to a multiple resource answering service such as an ACD (Automatic Call Distribution) split or attendant pool. This allows more than one call to be handled at a time when many Hot Line calls are being placed at once.

Feature History and Development

The Hot Line feature was first available on System 85 in Release 2, Version 3. In its initial form, it was available only with analog terminals.

- In System 85, Release 2, Version 4, the Hot Line feature was enhanced to make it available for DCP Data Call Setup from data modules.

User Operations

The following are the user operating procedures for this feature.

Placing a Hot Line Call From an Analog Voice Station:

Go off-hook on a designated Hot Line voice terminal. [Call-progress tones are returned.]

Placing a Hot Line Call From an Analog Data Station:

1. Go off-hook on a designated Hot Line Modem Voice Terminal. [Call-progress tones are returned. Ready tone is heard when the destination modem answers.]
2. Press the **[DATA]** button on the modem. [Call control is passed to the modem; ready tone is silenced.]

Placing a Hot Line Call From a DCP Data Station:

1. Press the **[ORIGINATE / DISCONNECT]** button on the data module, or the **[BREAK]** key on the data terminal keyboard. [If Keyboard Dialing is assigned, call-progress messages as follows are displayed on the data terminal screen.]

   DIAL: HOT LINE
   RINGING
   ANSWERED

2. When the call setup is complete, the appropriate login sequence or other initiation process is used.

NOTE: Key designations may vary depending on terminal model being used.

Considerations

Access to Other Features

The Hot Line extension cannot use most of the switch's voice terminal features since it immediately dials the Abbreviated Dialing list entry when going off-hook. For example, once the call has been dialed, flashing the switchhook is ignored (e.g., to attempt a transfer). The destination code will be redialed.

System Limit

There can as many as 52,223 Hot Line numbers administered in an R2 V4 switch (13,055 in R2 V3) if no nonsystem abbreviated dialing lists are used for the Abbreviated Dialing feature. This number is reduced by the number of personal and group abbreviated dialing lists assigned for purposes other than Hot Line. For digital data Hot Lines, the number is affected by the number of default dialing terminals in use.

Straight Line Sets

The Hot Line feature cannot be assigned to straight line sets. Hot Line terminals must be fully administered in Procedure 000, not in Procedure 051. As a result, Hot Line terminals cannot be bridged to multiappearance terminals.

Hard and Soft Processor Swaps

Stable Hot Line calls will endure a hard processor swap. However, Hot Line calls cannot be placed during a hard swap.

Hot Line destinations are stored in a translation portion of switch memory. Therefore, these destinations will endure a hard processor swap.

The Hot Line feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

The Hot Line feature uses the Abbreviated Dialing feature to perform the dialing function for voice Hot Line stations and for analog data hot line stations. Abbreviated Dialing must be provided to Hot Line stations.

Data Call Setup

Default Dialing

The default dialing capability of the Data Call Setup feature is used to perform the dialing function for digital data Hot Line stations. Data modules used for Hot Line stations must be assigned default dialing. The default dialing number assigned to the data module becomes the hot line destination number when the data module is administered as a hot line station.

If a DATA button is assigned for a digital data Hot Line station, the DATA button can be used to bypass the Hot Line designation. This is done by setting up a data call with the voice terminal and then using 1-button transfer to connect the data terminal to the call.

One-Button Transfer

The 1-button transfer function of the Data Call Setup feature can be used to override the Hot Line assignment for a Hot Line data station and complete a call to a destination other than the Hot Line destination for that station.

Host Computer Access

The Host Computer Access feature is 1-way compatible with digital data Hot-Line service. That is, a Host Computer Access port can be the Hot Line destination for a digital data Hot Line station, but a Host Computer Access port cannot be a Hot Line station. Host Computer Access ports are administered as trunk appearances, and Hot Line stations must be administered as line appearances. A computer port administered as a line appearance (not using the Host Computer Access feature) can be a digital data Hot Line station.

ISDN—BRI (Basic Rate Interface)

The ISDN—BRI feature is fully compatible with the Hot Line feature. If ISDN—BRI Voice/Data stations are assigned separate extension numbers for the voice and data appearances, the data extension can be assigned to Data Hot Line Service without affecting the voice extension.

IPA (Interpartition Access)

Hot Line numbers are not checked for legality at the time they are entered. Illegal Hot Line numbers are denied when the digits are outpulsed by the switch. When a user attempts to place a Hot Line call over another partition's trunk group or to an extension in another partition group, the switch returns intercept treatment to the calling party.

Tenant Services

Hot Line numbers are not checked for legality at the time they are entered. Hot Line calls are denied if they attempt to access facilities that are not allowed from the calling extension (i. e., if the Hot Line destination number would place a call over another partition's trunk group or to an extension in another partition). The switch returns intercept treatment to the calling party.

Restricting Feature Use

The use of a voice extension as a Hot Line can be temporarily denied by either blanking the first entry in the abbreviated dialing list or removing the list entirely. If the first entry in the abbreviated dialing list is blank, the user does not receive dial tone (silence) when going off-hook, however, a number can be dialed. If the list is removed entirely, the user receives intercept tone when going off-hook.

The following restriction features can also be applied

- Voice Terminal Restrictions can be used to limit the call placing and call receiving capabilities of any System 85 or DEFINITY Generic 2 terminal including a Hot Line station.
- Attendant Control of Voice Terminals can be used to limit the call placing capabilities of individual terminals or groups of terminals, including Hot Line stations, on a temporary basis.
- Attendant Control of Trunk Group Access restricts calls, including Hot Line calls from being placed directly over the restricted trunk groups. Attendant assistance is required for the use of these trunk groups.

If a Hot Line Station is to be reserved for Hot Line use only, it should be inward or termination restricted to prevent it from being used to receive incoming calls.

For Digital Data Hot Line Stations that are to be restricted to Hot Line Service use, do not assign a DATA button for the data module used. The 1-button transfer capability provided by the DATA button can override the default dialing function that is used for digital data call Hot-Line service.

See the Data Call Setup feature for more detailed information on voice terminal data call setup, default dialing, and 1-button transfer.

Hardware Requirements

None.

Feature Administration

Assignment of the Hot Line feature is on a per station basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can also administer this feature using the SMT (System Management Terminal) or TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

Hot Line can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — HOT LINE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	3	Assigns Hot-Line service to a voice terminal or data module extension number.	Yes
059	1	Assigns a 5-item personal abbreviated dialing list to an extension number.	Yes
059	2	Assigns destination or access codes to personal abbreviated dialing lists. For Hot-Line service, only the first entry in list "A" should be assigned.	Yes
059	4	Assigns the default dialing number for a data module. This is used as the Hot Line destination number for digital data hot line stations. Not applicable prior to Release 2, Version 3.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — HOT LINE	
PATH NAME	**PURPOSE**
terminal-change extension attributes	Assigns Hot Line to an extension number.
terminal-change group abbreviated-dialing	Assigns an extension to an abbreviated dialing list and assigns the list entries.

Notes:

Hunting

Description

The Hunting feature routes calls directed to a busy terminal to other terminals in a predetermined group (hunt group). The status (busy or idle) of extensions within the group is checked in order. If an extension is busy, the call routes (hunts to) to the next extension. This hunting process continues until an idle extension is found or the hunt group has been exhausted, depending on the type of hunting that is used. There are two types of hunting or search patterns that can be used, circular or linear.

- Circular Hunting

  In circular hunting, the hunt starts with the called extension and proceeds to check ***all*** extensions in the hunt group. The call completes to the first idle extension found The hunt routine can check up to 30 terminals. In hunt groups with less than 30 terminals, some terminals are checked again. If every extension number in the hunting sequence is busy, the switch returns busy tone to the caller.

- Linear Hunting

  In linear hunting, the hunt starts at the called extension and proceeds to check the ***remaining*** extensions in the hunt group. If the called extension is not the first number in the hunt group sequence, the extensions preceding the called number in the hunt group are ***not*** checked. If the remaining extension numbers are busy, the switch returns busy tone to the caller.

The Hunting feature provides efficient use of voice terminals when heavy incoming calling is experienced in a particular department This feature improves customer service by allowing as many as 30 people to respond to a call. Also, since the switch hunts for an idle terminal, the caller is less likely to receive busy tone.

Feature History and Development

The Hunting feature was first available on System 85 with Release 1. This feature has remained unchanged since its introduction.

User Operations

The Hunting feature provides a service for calls coming into a System 85 or Generic 2 hunt group. There are no user operating procedures, and its operations are transparent to callers.

Considerations

Group Size

Any number of extensions can be assigned to a hunt group. However, the hunting algorithm will not check more than 30 extension numbers.

Local Restriction

Hunting can only be done within the local switch.

Hunting Destination

A terminal can only hunt to one other terminal. However, Call Forwarding—Busy and Don't Answer can be used to route calls out of the hunt group to an attendant position.

Sequencing

A hunting group is established sequentially. This is done by assigning the next (hunt to) destination for each extension in the group.

Dual Hunting to Same Point

Hunt groups can be set up with more than one terminal hunting to the same extension. The last terminal in one or more linear hunt groups can hunt to a circular group.

Hunting to Multiappearance Voice Terminals

When the switch hunts to an extension with more than one appearance, the algorithm attempts to distribute the call to every appearance with terminating capabilities before hunting to the next extension in the group.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

When the called terminal is part of a hunting group, hunting is performed before the Attendant Call Waiting feaute is implemented. If an idle terminal is not found in the hunt path, the call waits on the originally called terminal.

ACD (Automatic Call Distribution)

Any individual ACD agent extension can be included in a hunt group other than the ACD split. Hunting functions normally in this case. However, an ACD split's controlling extension cannot be assigned to a hunting sequence.

Automatic Callback

If Automatic Callback is used toward a busy terminal in a hunt group, hunting does not take place. The call does not complete until the specific called terminal becomes idle.

Busy Verification of Lines

Hunting is not performed when the Busy Verification of Lines featue is used toward a terminal in a hunting group.

Call Coverage

Normally, Call Coverage takes precedence of Hunting. That is, if the called extension has both Call Coverage active and has a hunt path assigned, an incoming call will follow the coverage path. However, if the coverage point is a VDN (ACD split) and there are no agents staffed, the call will follow the Hunting path.

When a call routes to coverage, hunting is only done from the final coverage point. At this point, hunting is limited to the first nine members of the hunt group (after the coverage point).

CDR (Call Detail Recording)

The call record shows the hunted-to extension number (the extension that actually answers the call) as the called number.

Call Vectoring

A vector directory number cannot be assigned to a Hunting sequence. When this is attempted, an administration error will occur.

A member of a hunt group can be assigned as the destination of a "route to" vector step. If the "route to" step is the final effective step in the vector and the routed-to extension is busy, vector processing checks for an idle extension in the hunt group. If the "route to" step is not the final effective step in the vector and the routed-to extension is busy, vector processing goes on to the next vector step without checking for an idle extension in the hunt group.

Call Forwarding—Busy and Don't Answer

When a call is forwarded to a busy terminal in a hunt group, hunting takes place. Another terminal in the hunt group may receive the call rather than the terminal selected in the call forwarding arrangement.

When an incoming call hunts to a terminal in a hunt group with Call Forwarding—Busy and Don't Answer active, the call forwards as specified by the call forwarding arrangement (call forwarding takes precedence).

Call Forwarding—Don't Answer

When a call forwards to a busy terminal in a hunt group, hunting occurs and another member of the hunt group may answer the call. If an incoming call reaches a terminal in a hunt group with Call Forwarding—Don't Answer active, the call forwards if it is not answered. When the forwarding terminal is busy, the call will hunt.

Call Forwarding—Follow Me

When a call is forwarded to a busy terminal in a hunt group, hunting takes place. Another terminal in the hunt group may receive the call rather than the terminal selected in the call forwarding arrangement.

When a terminal in a hunt group uses Call Forwarding—Follow Me, that terminal is bypassed. No attempt is made to connect or forward the call; however, the extension record is checked to identify the next ***Hunt to*** extension.

If the terminal specified by Call Forwarding—Follow Me has Call Waiting active and is in a hunt group, a call forwarded to that busy terminal hunts for an idle appearance first and then waits on the specified terminal if no idle appearance is found.

Call Waiting

When a called terminal assigned Call Waiting is busy and the terminal is in a hunting group, an incoming call hunts before the Call Waiting feature is implemented. If every terminal is busy, the call waits on the originally called terminal.

EUCD (Enhanced Uniform Call Distribution)

Any individual EUCD agent extension may be included in a hunt group. Hunting functions normally in this case. However, an EUCD split's controlling extension cannot be assigned to a hunting sequence.

Extension Number Portability

An extension number must be removed from a hunting sequence before that extension number can be ported to another node.

Host Computer Access

The Hunting feature can be used to check if an alternate circuit in a Hunt Group can serve a call when the dialed circuit is busy. To combine the Hunting feature with Host Computer Access, administer the port circuits as lines.

The Hunting principle is also used for trunks that are assigned to the same trunk group. This hunting service is automatically provided for trunk circuits in the same trunk group. When dialing a Host Computer Access trunk group, access is provided to as many as 99 computer ports. (Trunk groups can also be combined using Route Advance to provide additional access.)

Leave Word Calling

Leave Word Calling messages are always left for the principal originally called, even when the Hunting Feature is active. The only exception to this is calls redirected to an attendant. Leave Word Calling is not allowed when a call is redirected to an attendant.

Override

An Override call does not hunt. Override calls terminate to the dialed extension.

Priority Calling

Hunting is not performed when Priority Calling is used toward a terminal in a hunting group. Priority calls either wait on a called single-appearance voice terminal or terminate to an idle appearance of a multiappearance voice terminal (if available).

Queuing

When the local switch attempts a Queuing callback, the callback call does not hunt.

Callback Calling

If a Callback Call is made over a tandem tie trunk, the call appears as a normal incoming tie trunk call to the subtending switch. If the called terminal is in a hunt group and is busy when a callback attempt is made, the callback call will hunt.

Restriction—Attendant Control of Voice Terminals

When a call is placed directly to a restricted voice terminal in a hunt group, the switch returns Intercept Treatment to the calling party. The call does not hunt.

However, when a call is placed to an unrestricted voice terminal in a hunt group, the call can hunt to the restricted terminal and beyond. The hunting pattern is not affected.

Restriction—Voice Terminal Restrictions

When a call is placed directly to a restricted voice terminal in a hunt group, the switch returns Intercept Treatment to the calling party. The call does not hunt.

However, when a call is placed to an unrestricted voice terminal in a hunt group, the call can hunt to the restricted terminal and beyond. The hunting pattern is not affected.

Tenant Services

There are no tests in System 85 or Generic 2 software to prevent hunting to another partition. It is the responsibility of the system manager to ensure that the extensions in a hunt group do not cross partition boundaries.

Timed Reminder

When the STA ID button is pressed on the attendant console, the extension number shown on the attendant display is the extension the call hunted to rather than the originally called extension.

Hardware Requirements

None.

Feature Administration

Assignment of the Hunting feature is on a per-terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

Hunting can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE — HUNTING			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the class of service to an extension number.	Yes
000	2	Assigns a hunt-to extension number to each extension in a hunt group.	Yes
010	1	Assigns "stop hunt" to a voice terminal class of service.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — HUNTING	
PATH NAME	PURPOSE
terminal-change extensions hunt	Displays or prints a report of the hunting sequence for a specified extension or a set of extensions.
terminal-change extensions attributes	Asigns a hunt-to extension number to each extension in a hunt group.

Information Systems Network Interface

Description

The ISN (Information Systems Network)

The ISN is a packet switched LAN (Local Area Network) that connects digital processors and devices in an asynchronous distributed processing system. The distribution system and carrier medium can be all-wire or a combination of wire and optical fiber. The ISN Packet Controller is a high-speed packet switch that moves packets of information from one network component to another. A single packet controller can service as many as 1920 asynchronous data devices (ISN stations). Additional information on the ISN can be found in Introduction, Information Systems Network, 555-300-020, and Reference Manual, Information Systems Network, 555-300-210.

The System 85/DEFINITY Generic 2 Interface to ISN

EIA (Electronics Industry Association) Trunk Ports

System 85 and DEFINITY Generic 2 ("circuit switches") interface to the ISN through 1-way EIA trunk ports. The "one way" is a trunk function (incoming calls or outgoing calls only) and not a data flow function. These ports provide full duplex, RS-232 asynchronous, 10-bit start/stop data transmission (see Figure 64-1).

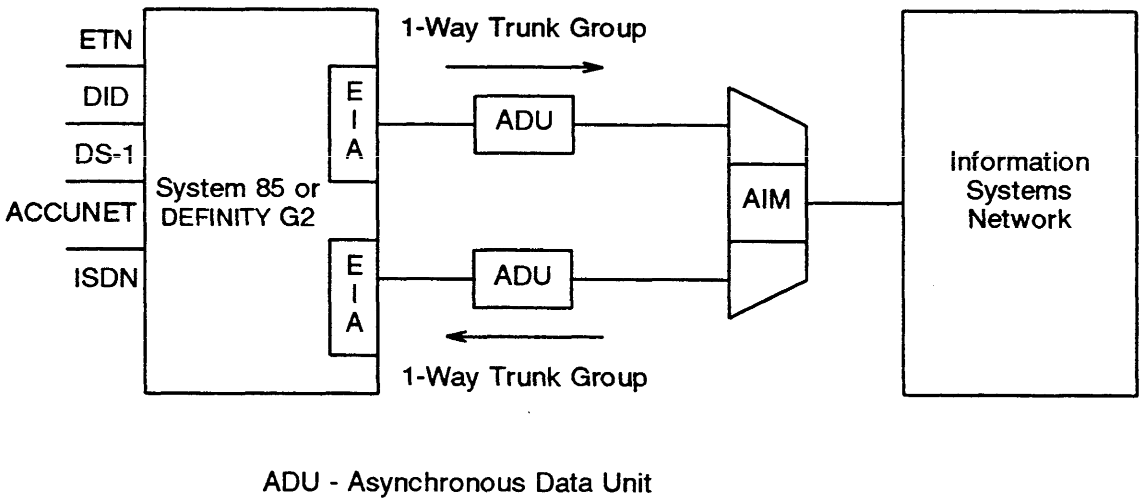


Figure 64-1. Circuit Switch to ISN Connectivity

Z3A3 ADU (Asynchronous Data Unit)

An ADU is used in the connection between the EIA port of the circuit switch and the AIM (Asynchronous Interface Module) of the ISN. The Z3A3 ADU connects to the EIA port via a cross-connect cable and serves as a signal buffer that extends the range of the EIA signal from 50 feet to between 2,000 feet and 40,000 feet depending on data rate and conductor gauge.

AIM (Asynchronous Interface Module)

These trunks connect to an AIM located in either the Packet Controller or Concentrator. The bit rate for each trunk group is fixed at a standard asynchronous data rate (300; 1,200; 2,400; 4,800; 9,600; or 19,200 bps). When accessing an ISN Interface trunk group, selection is made based on the data rate needed.

Protocol Conversion: The EIA circuit provides protocol conversion between EIA (on the ISN side) and DCP (on the switch side). To the circuit switch, this circuit board looks and responds like a PDM (Processor Data Module).

Synergism

The ISN Interface feature enables the ISN and the circuit switch to work together to enhance each other's data communications capabilities. The circuit switch provides gateway services for the ISN for connection to ETN (Electronic Tandem Network), DCS (Distributed Communications System), ISDN (Integrated Services Digital Network), and other external networks. This integration also enhances the data-switching capabilities of the circuit switch. Services and enhancements include the following

- Connections between the circuit switch and the ISN are set up using standard Data Call Setup dialing techniques.
- Access to applications on the Applications Processor is available from either the circuit switch or the ISN terminals.
- Access to the public and private networks is available to ISN terminals through the circuit switch.
- Access to ISN stations is available from external, public, and private network data stations through the circuit switch.
- Access to the ISN's high-speed multiplexed networking capabilities is available to data stations on the circuit switch.
- Integrated administration is available for both ISN and the circuit switch from a single System Control Terminal.

Feature History and Development

The ISN Interface feature was first available with System 85 in Release 2, Version 2.

User Operations

Access to the ISN from the circuit switch uses the Data Call Setup feature.

Keyboard Dialing:

1. Press the **[BREAK]** key on the keyboard,

   or

   Press the **[ORIGINATE/DISCONNECT]** button on the associated data module.
   [Dial prompt]

2. Dial the appropriate ISN Interface trunk group access code or extension number.*
[Second dial prompt]
3. Dial the ISN destination number.

Voice Terminal Dialing:

1. Dial the desired ISN Interface trunk group access code or extension number?
[Second dial tone]
2. Press the appropriate **[DATA]** button to transfer control of the call to the assigned data terminal.
3. Complete dialing to the ISN destination number from the connected data terminal.

Host Computer Dialing:

It is possible to use Host Computer Dialing to access selected ISN stations including ISN Host Computers. The off-hook/on-hook signals and dialing instructions are generated from software provided by the user.

Disconnecting Voice Terminal Controlled ISN Calls:

1. At the controlling voice terminal, go off-hook.
2. Press the appropriate **[DATA]** button. (The ISN call is transferred back to the voice extension.)
3. Go on-hook. (The ISN Interface trunk seizure is dropped.)

Disconnecting ISN Interface Calls Using Keyboard Dialing:

Press the **[BREAK]** key for 2 seconds if long-space disconnect is active,

or

Press the **[ORIGINATE/DISCONNECT]** button on the data module.

See the Data Call Setup feature for a more detailed description of the procedures and options available to circuit-switch users.

Accessing the Circuit Switch From ISN

To access the circuit switch from the ISN, use 1- or 2-stage keyboard dialing. Two-stage dialing is the same as above; except that all dial prompts come from the ISN and are different from the prompts given by the circuit switch. The ISN controls all the address signaling. One-stage dialing is performed by entering the circuit-switch trunk group access code and the destination code on a single input line.

* ISN Interface trunk groups have fixed data rates. Select a trunk group with a data rate that matches your terminal.

Considerations

Common Terminal Administration

A 500 BCT (Business Communications Terminal) connected to an AP (Applications Processor)-16 can be used to perform remote administration of the ISN through the ISN Interface connection. This allows the same terminal to be used to administer all three connected systems (ISN, circuit switch, and AP-16).

An EIA terminal, such as the 513 BCT or a 515 BCT, connected to the circuit switch can also be used to administer the ISN. In all cases, the terminal used to administer the ISN must be operating in a "Dumb Terminal Mode" (TTY Emulation for the 500 and 515 BCTs). A 500 BCT connected to a 3B5 AP does not have a dumb terminal mode ("TTY Emulation) and cannot be used to administer the ISN.

Switch administration, using the Terminal Change Management feature, can also be done from the same AP terminal or 515 BCT. Switch administration can also be done from an ISN terminal.

Common Printer

The ISN Interface feature allows facilities, such as printers connected to the AP, to be shared by terminals and work stations on either the ISN or the circuit switch.

Differences in Dialing Procedures

2-Stage Dialing: Dialing from the circuit switch to an ISN destination requires a separate response to each of two dial prompts: one from the circuit switch and the other from the ISN Controller.

1-Stage Dialing: ISN users can access stations on the circuit switch using 1-stage keyboard dialing. This is done by entering the circuit-switch interface access code and the destination number on a single input line.

Interface Trunk Characteristics

Trunks used for the ISN Interface are administered as 1-way, either in or out (as seen from the circuit switch). These trunks are also assigned a fixed data rate. Users on the circuit switch side must know what trunk groups (trunk group access codes or extension numbers) match the data rates of their terminal.

On the ISN side of the interface, flow control is available to modify data rates so that matching data rates to ISN terminals are unnecessary. It is necessary, however, for ISN users to select the appropriate trunk group to match the circuit-switch data end point when placing an ISN to circuit switch call, unless the circuit switch data end point has an autobaud capability.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the use of the ISN Interface feature.

AAR/ARS (Automatic Alternate Routing and Alternate Route Selection)

On System 85 and Generic 2.1 switchers, ISN data stations can use the AAR or ARS feature when placing private- or public-network calls through the circuit switch. When this is done:

- Modem Pooling may be required if the trunk group accessed contains analog trunks
- Two-stage dialing is required from the ISN station.

Calls to or from the ISN interface are handled by the AAR or ARS feature just like any other data calls on the switch.

CDR (Call Detail Recording)

The CDR feature provides information on and specific identification of data calls including ISN Interface calls.

Data Call Setup

The Data Call Setup feature is used to access the ISN Interface from the circuit-switch side. When keyboard dialing is used, circuit-switch users receive "TRY AGAIN" when all facilities are busy, while ISN users receive "BUSY."

Data Protection

Data Protection—Permanent should be assigned to all ISN Interface trunk groups to prevent system generated tones from interrupting data communications. These tones are issued in response to the activation of certain features, such as Override and Priority Calling. These features should not to be used toward data connections. Data Protection prevents inadvertent interruptions by barge in features.

DS1 Interface

When the DS1 Interface feature with AVD (Alternate Voice / Data) trunks or ISDN—PRI trunks are used to link circuit-switch locations, end-to-end digital connections between ISN stations and remote locations are possible. These arrangements do not require support from the Modem Pooling feature.

DID (Direct Inward Dialing)

When an off-net voice terminal is used to set up a data call to the ISN Interface, stage one dialing can be performed from the voice terminal (that is, to dial the ISN Interface access code); however, stage two dialing must be used for keyboard dialing (to dial the ISN endpoint).

Host Computer Access

The Host Computer Access feature is available to ISN stations through the ISN Interface feature.

ISDN—BRI (Basic Rate Interface)

The ISDN—BRI feature is compatible with the ISN Interface feature. ISDN—BRI stations can access ISN endpoints and ISN stations can access ISDN.nr;G 1 I endpoints via the ISN Interface.

ISDN—PRI (Primary Rate Interface)

The ISN Interface feature and the ISDN—PRI feature are compatible. ISN stations can access ISDN—PRI facilities (and ISDN—PRI calls can access ISN stations) through the interworking function.

Modem Pooling

Modem Pooling on the circuit switch provides 2-way connectivity between ISN endpoints and remote stations via either public or private network trunks. For example, a remote data station user can dial the circuit-switch number assigned to the ISN trunk group, receive the ISN dial prompt, and dial the ISN destination code. An ISN station user can also dial through to the remote data station. Use of Modem Pooling requires that extension number steering be available to access ISN Interface trunks.

Queuing

Circuit-switch endpoints can use Queuing if all ISN Interface trunk circuits are busy, and if the queue is not full. Voice terminal users can be provided with on-hook (if administered) or off-hook queuing. Data terminal users can use off-hook queuing only. Queuing is not provided by ISN. That is, when a call from the circuit-switch side has passed the ISN Interface point and is under ISN control, Queuing is no longer available. Also, Queuing is not available to ISN station users until after the call has passed the interface point.

Route Advance

The Route Advance feature provides access to as many as five trunk groups with a single access code. When all circuits in the first trunk group are busy, the system checks the next trunk group and so on. With up to 256 trunks in a trunk group, this provides up to 1280 port appearances with a single access code.

Tenant Services

A partitioned System 85 or Generic 2 can serve as an endpoint in a tenant's data network. As long as the partitioned switch is serving as an endpoint to the ISN, an extension partition within the partitioned switch can access the ISN over dedicated trunk groups. However, a partitioned circuit switch should not reside between the ISN and another switch. Acting as a tandem, the partitioned circuit switch does not provide trunk-to-trunk partitioning to the ISN.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switchers, ISN data stations can use the WCR feature when placing private- or public-network calls through the circuit switch. When this is done:

- Modem Pooling may be required if the trunk group accessed contains analog trunks
- Two-stage dialing is required from the ISN station.

Calls to or from the ISN interface are handled by the WCR feature just like any other data call on the switch.

Restricting Feature Use

For circuit-switch users, the Data Call Setup feature is used to access ISN Interface. Therefore, the same restrictions that apply to the Data Call Setup feature also can be applied to ISN Interface.

Temporary Station Restrictions: The circuit-switch attendant can use the Restriction-Attendant Control of Voice Terminals feature to restrict use of this feature by circuit-switch stations and to limit access to circuit-switch stations via this feature.

Fixed Station Restrictions: Fixed restrictions that apply to terminals that could be used with the ISN Interface feature include the following

- Origination Restriction
- Termination Restriction
- Terminal-to-Terminal Only Restriction
- Inward
- Outward.

Trunk Restrictions: Also, because ISN Interface ports are trunk appearances on the circuit switch, the following trunk restrictions can be applied.

- Attendant Control of Trunk Group Access

  CAUTION: *If an attendant is to screen calls, a voice terminal equipped with transfer capabilities must be provided near the attendant console to perform the transfer. Data transfers cannot be performd from the attendant console, and Modem Pooling support is not provided* to *attendant-extended calls. Also, the attendant-console cannot handle Keyboard Dialing originated calls.*

- Facilities Restriction Levels
- Miscellaneous Trunk Restrictions
- Trunk Group Restrictions.

Hardware Requirements

The following circuit-switch hardware items are required to implement the ISN Interface feature:

For Traditional Modules:

- SN238 EIA Port Circuit Packs (four circuits per SN238).

For Universal Modules:

- TN726 Data Line Circuit Packs (eight circuits per TN726).

Regardless of the Module Type:

- Z3A3 ADU (Asynchronous Data Unit) Interface Unit, one for each EIA trunk circuit used. This unit is sometimes referred to as a Limited Distance Modem.

On the ISN side, an AIM (Asynchronous Interface Module) optioned to present a DTE (Data Terminal Equipment) interface is required. The AIM provides eight port connections.

Feature Administration

The trunk ports that connect the circuit switch to the ISN are assigned on a per-trunk basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel) or SMT (System Management Terminal).

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — ISN INTERFACE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
070	1,2,3,4	Displays detailed information for an equipment location including. equipment type, state of health, bit rate (bps), parity, transmission mode and type, and EIA leads.	Yes
100	1	Assigns a trunk group dial access code and trunk type to a trunk group. For incoming trunk groups (ISN to circuit switch) the following encodes apply: ● Trunk Type = 107 ISN/EIA ● Dial Access Restriction = 1 This prevents terminals assigned to the circuit switch from accessing these trunks, but permits testing via the Trunk Verification by Station feature.	No
100	2	For System 85 switches: Administers data characteristics for trunk groups assigned to ISN Interface. For DEFINITY Generic 2 switches: Assigns Bearer Capability Class of Service to trunk groups.	No
150	1	Assigns trunks (by equipment location) to trunk groups.	No
354	2	Assigns extension number steering.	No

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — ISN INTERFACE	
PATH NAME	**PURPOSE**
terminal-change terminal equipment	Assigns an equipment location to an EIA port. Also assigns an extension number to the EIA port.

Notes:

BRI (Basic Rate Interface)

Description

The ISDN—BRI feature provides a station direct ISDN interface within the DEFINITY Communications System Generic 2 switching network. With ISDN—BRI service, ***each BRI station*** has its own interface that sends and receives control and signaling messages.

Feature History and Development

The ISDN—BRI feature was introduced with Generic 2. This feature is not available on any System 85 Switches.

The ISDN—BRI feature was enhanced in Generic 2.1 3.0 with the addition of the following

- Prime Data Line and data appearances
- Service SPID (Service Profile Identifier)
- ISDN—BRI PC Interface.

Terms

Ports and Interfaces

The terms "port" and "interface" are essentially synonymous when used in connection with BRI. A BRI port consists of two B-channels and one D-channel. Note that "port" and "channel" are **not** the same thing. The expression "BRI Interface" is redundant (Basic Rate Interface Interface) and will not be used in the rest of this discussion.

Initializing and Non-initializing Terminals

There are two types of ISDN—BRI terminals, initializing and non-initializing. The difference between these two terminal types is that the initializing terminals go through an initialization sequence with the switch (similar to handshaking for a data module) when they are first placed into service and again any time the switch is stopped and restarted. Non-initializing terminals do not go through this sequence.

The advantage to initializing is that this sequence involves condition and maintenance checks to insure that the terminal and its interface are ready and able to function properly. The disadvantage is that the initialization process consumes switch processor time, but only at the time of initialization when call activity is often low.

MINs (Management Information Messages)

Management Information Messages are a special class of ISDN messages. These MIMs are passed between the switch processor and an ISDN—BRI station and are used in maintenance and management checking to query the station for status and to advise the switch of conditions at the station. Functions performed include: interrogation of terminal ID and SPID by Administration tasks, terminal maintenance checks, terminal initialization, and data call option queries (mode, data rate, etc.) by call processing.

SPID (Service Profile Identifier)

The SPID is a unique number (10 digits or less) assigned to a BRI station that includes an initializing terminal. Note that if two terminals (for example, one voice terminal and one data terminal) are assigned to the same station (that is, the same ELL), they both share the same SPID. The SPID is used by the switch during the initialization process as a recognition tool for BRI terminals. See Considerations in this chapter for additional information on the SPID.

Service Spid

The service Spid is a SPID that can be assigned to a test terminal and then used by maintenance and service personnel to test any BRI station (port). The Service SPID, when used, is common to all BRI stations on the switch. However, the Service SPID provides limited functionality and should not be used for normal station operations.

Stations, Work Stations, and Terminals

In reference to BRI, the terms station and work station are not synonymous with terminal. A station can consist of two different terminals (one voice terminal and one data terminal) *connected to the same ELL (Equipment Line Location).* This arrangement is called a voice / data station. The key here is that both terminals are connected to the same ELL, thus forming an single station. Another option is the ISDN—BRI PC Interface work station which consists of a PC and from one to four voice terminals or voice calling devices (handsets or headsets). With this arrangement, each combination of PC and voice-calling device uses a separate ISDN Interface card and connects to a separate port (ELL). From the switch perspective, each ISDN Interface card constitutes a separate station. (Each separate station may or may not have its own separate extension number.)

The Interface Configuration

Each ISDN—BRI is served by two "B" (bearer) channels and one "D" (data) channel. In ISDN terminology, this arrangement is called a **2 B plus D** configuration.

B-Channels

Each B-channel provides 64 Kbps digital communications services. That is, the B-channels carry the conversation (or data stream) for the terminals. Each B-channel can support a separate connection (a voice terminal or a data terminal), providing for two separate terminals that can operate simultaneously on the same interface circuit.

D-Channel

For the BRI feature, the D-channel is a 16 Kbps digital link (as opposed to 64 Kbps for PRI) that carries call control and signaling messages. The main function of the D-channel is signaling and control for the associated B-channels.

BRI Station Configurations

ISDN—BRI stations can be configured in the same ways as the DCP stations. There are three possible configurations:

- Voice terminal only
- Data terminal only
- Both voice and data terminals at the same station.

The BRI Voice/Data Station

The basic ISDN—BRI terminal configuration is in the form of the *Voice/Data Station.* This configuration takes maximum advantage of the power and versatility of the ISDN—BRI and is similar to the *DCP Voice/Data station.* From 65-1 shows a comparison between a BRI Voice / Data Station and a DCP Voice / Data Station.

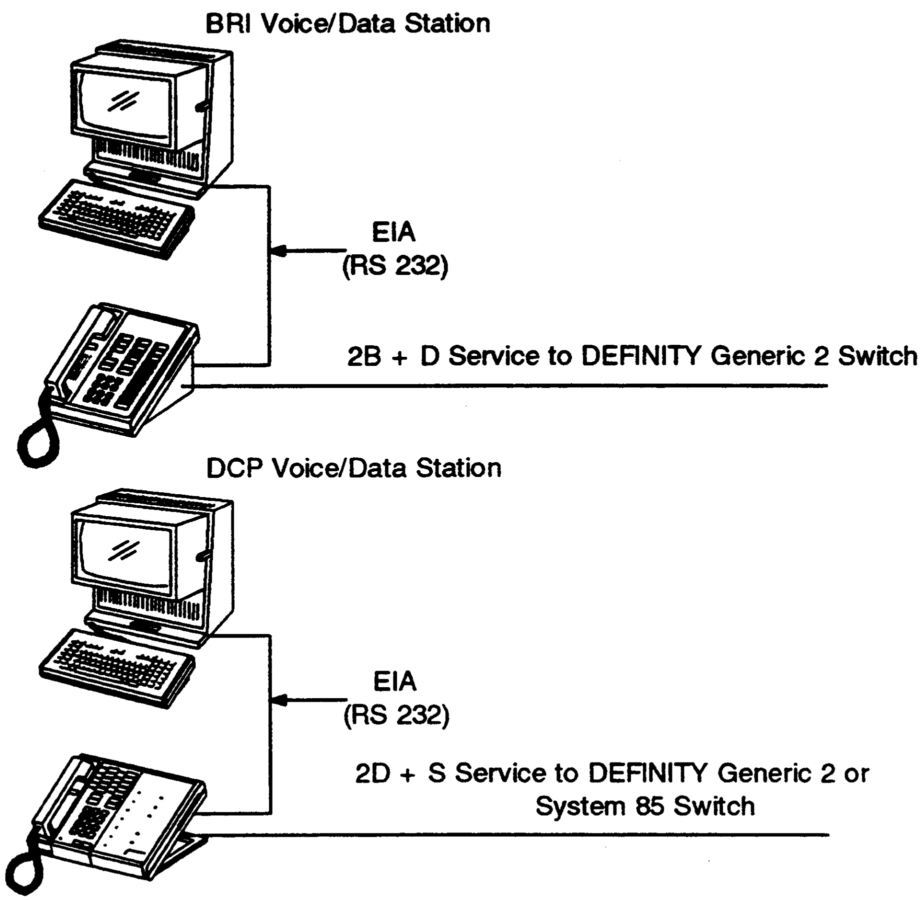


Figure 65-1. BRI and DCP Voice / Data Stations

Notice that both types (BRI and DCP) consist of a voice terminal and an associated data terminal. Both configurations are also served by two communications channels (B-channels for the BRI station), and one call control and signaling channel [D (data) channel for the BRI station). Both BRI and DCP use the DMI specification.

These configurations are similar because both BRI and DCP are designed from the same conceptual model. Both protocols do essentially the same things although the detailed method by which they do them is different. It is this close conceptual relationship between BRI and DCP that allows these two protocols to work well together.

Prime Data Line

The ability to administer both a data appearance and a prime data line to an ISDN—BRI station is added in Generic 2.1 3.0. The prime data line is a data only appearance on a BRI voice terminal equipped with an ADM (Asynchronous Data Module).

A prime data line functions similarly to the *Prime Appearance Preference* as described in the Multiappearance Preelection and Preference feature, except that it is for data calls only. When administered, the first data call (either originating or terminating) associated with the ISDN—BRI Voice / Data Station, will automatically select the prime data line appearance if that appearance is available.

It is also possible, with Generic 2.1 3.0, to administer a non-prime data appearance. A data appearance (non-prime) functions similarly to a normal appearance except that it is for use with data calls only. That is, an appearance administered as a data appearance (either prime or non-prime) cannot be used to either originate or terminate a voice call.

Data Appearance

The prime data line is a data appearance with the added attributes of the preselection and preference function.

Call Origination:

When a *data call* is originated from a BRI terminal with a prime data line assigned, the appearance selected for that call will be the prime data line if that appearance is available. If the prime data line is not available for any reason, the remaining appearances will be searched in squence, beginning with the first appearance following the prime data line, for an available appearance. The first available appearance (whether a data appearance or not) will be selected.

When a *voice call* is originated from a BRI terminal with one or more data appearances assigned, all data appearances (both prime and non-prime) are skipped when searching for an available appearance.

It should be noted that it is not necessary to assign either a prime data lime or a data appearance to a BRI terminal for that terminal to be able to handle data calls. For BRI terminals without data appearances assigned, data calls will select the first available standard appearance.

Call Termination:

When a data call terminates to a BRI station with one or more data appearances assigned, the call will be routed to the first available data appearance (including Prime Data Lines if assigned). If no data appearance is available, the first available standard appearance (voice or data) will be selected. For data call termination, one immage of each BRI data appearance must be designated as being on the home terminal. Note that this differs slightly from the data call origination case.

When a *voice call* is terminated to a BRI terminal with one or more data appearances assigned, all data appearances (both prime and non-prime) are skipped when searching for an available appearance.

Bridged Images of Data Appearances

Unlike DCP data appearances, it is possible to administer bridged images of ISDN—BRI data appearances. These images provide a means of monitoring (through status lamp states) the state of the data appearance. They also provide a separate data appearance for the purpose of data call origination (but, because they are not on the home terminal, they cannot terminate a data call). Unlike bridged images of a standard appearance (with a voice call active), they cannot be used for access to (bridge onto) an active data call.

> **NOTE:** With a bridged image of a standard BRI appearance when a data call is active, bridging onto the active data call is not allowed (intercept tone is returned) even without Data Protection assigned.

Bridged images of a data appearance can be designated as prime data lines or simply as data appearances. That is, multiple images of the same data appearance, can be designated as a prime data line on any, all or none of the stations where they occur, whether or not they are designated as the prime data line on the home terminal for that appearance. Specific constraints include the following any given station can have only one prime data line; an image of any appearance can occur only once on any terminal, and an appearance that is designated a data appearance on any of its images is at data appearance on all of its images.

Connectivity

In Generic 2, the ISDN—BRI feature provides line side service only. That is, it operates within the network provided by the Generic 2 switch and cannot be used where a trunk type appearance is needed (for example the Host Computer Access feature). ISDN—BRI Off-Premise stations are not available with DEFINITY Generic 2.

ISDN Connections

Access to external (off-net) features and services is provided to BRI terminals by the ISDN—PRI feature, or through ***interworking*** by other Generic 2 trunking and networking features. Some of the ISDN connections that may be available to Generic 2 BRI terminals are shown in Figure 65-2.

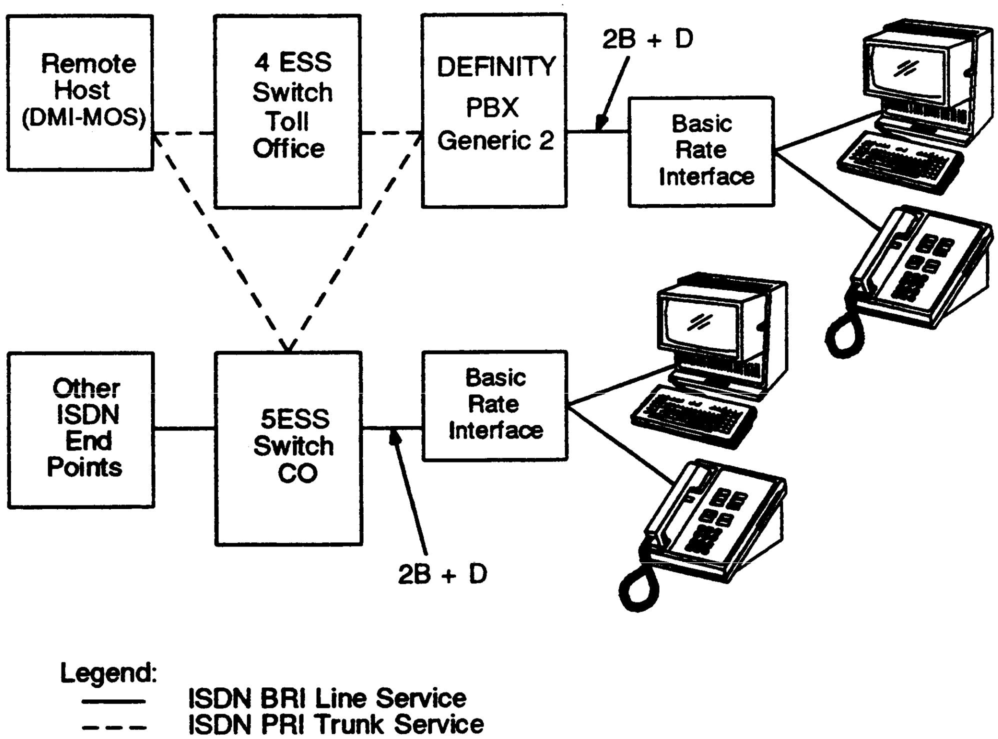


Figure 65-2. ISDN—BRI connectivity to ISDN End Points

Connections to Non-ISDN End Points

It should not be inferred from Figure 65-2 that ISDN—BRI terminals can only call other ISDN end points. The ***interworking*** function provides connectivity to virtually every end point available to other digital terminals. Figure 65-3 shows typical non-ISDN connections available to ISDN—BRI terminals.

The DEFINITY Communications System, Generic 2 Implementation

Bearer Capability

Bearer capability applies to all calls and call support facilities, but is of primary significance to data calling requirements. For ISDN—BRI voice/data stations, the call discrimination provided by bearer capability is very significant.

A ***BCCOS*** (Bearer Capability Class of Service) is assigned to each extension, trunk group, and routing preferance. For non-ISDN non-data module calls, and as a last resort for all calls that are lacking needed information in the call setup message, this BCCOS can be used to obtain a ***default*** value for bearer capability requirements information.

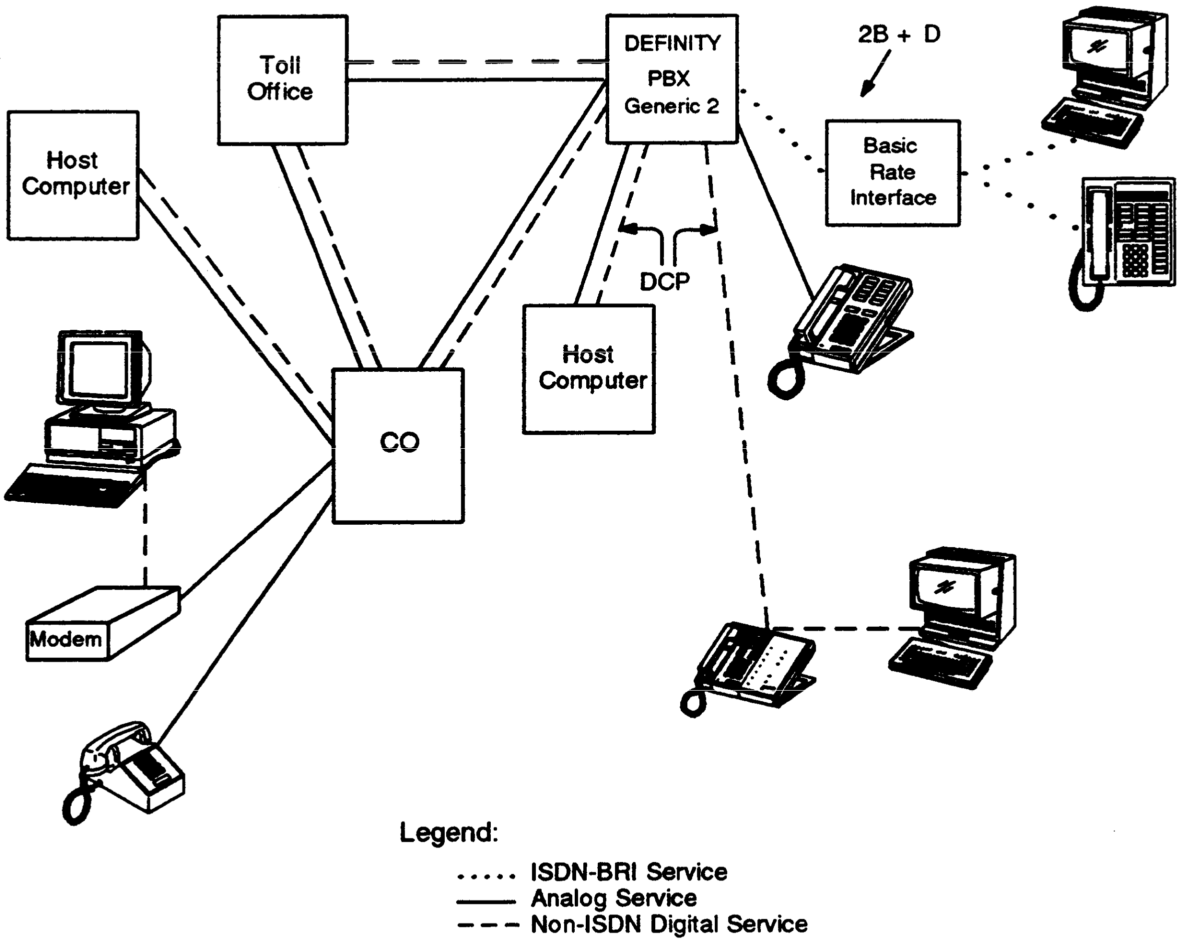


Figure 65-3. ISDN—BRI connectivity to Non-ISDN End Points

Predefine Bearer Capability Class of Service

In Generic 2 switches, nine BCCOSs are predefined. Some of these predefined BCCOSs are used as defaults. For example, on Generic 2 switches, all origination, termination, and call processing support facilities come from the factory with a predefine BCCOS assigned. Voice terminal extensions default to BCCOS "0", while BCCOS "1" is the default for DCP data modules. Note that BCCOS applies to non-ISDN facilities (such as analog and DCP lines and trunks) as well as to ISDN facilities. Table 65-A lists the nine predefined BCCOSs and the closest corresponding System 85, Release 2, Version 4 bearer capability code (if any).

For ISDN—BRI extensions administered as "voice", the default BCCOS is "0". This is the same BCCOS that would be used for a DCP voice extension or an analog voice terminal. In a few cases, this BCCOS does not work properly for calls from ***international*** sources or calls from other vendor's switches. This is because of differences in signaling conventions used. If problems are experienced on these types of calls, the BCCOS can be modified to accommodate these differences *(see* ***Bearer Capability Class of Service,*** under Considerations, later in this chapter).

TABLE 65-A. Predefined Bearer Capability Classes of Service

Default BCCOS DEFINITY G2	BCC System 85 R2V4	Type of Call Supported
0	(0)	Voice only*
1	2	Mode 2 Data
2	—	BRI Voice / Data
3	—	Unknown Digital
4	(0)	Unknown Analog
5	(0)	Voice Grade Data*
6	4	Mode 0 Data
7	1	Mode 1 Data
8	3	Mode 3 Data

* Bearer services available on System 85, Release 2, Version 4 but in a single BCC

For an ISDN—BRI extension that is to be used for data only, the appropriate BCCOS is usually "1". This is the same BCCOS that is used for DCP data modules. These BCCOS will normally be assigned by default during standard switch translation.

Table 65-A shows BCCOS "2" as BRI Voice / Data. BCCOS 2 is used for the special case where both the voice and data terminal share the same extension number. This application works except when incoming calls are classified as either ***unknown analog*** or ***unknown digital.*** With BCCOS 2, neither the switch nor the interface can tell which terminal (voice or data) should receive an incoming call classified as ***unknown.*** Also, when calls are forwarded from a terminal where both voice and data calls use the same extension number, all calls (both voice and data), are forwarded to the same extension.

Effect of BCCOS on Call Processing

Bearer capability values are used by call processing software to decide how to handle a specific call. For example, BCCOS 4 in Table 65-A indicates an analog call of an unknown nature. If an incoming call with BCCOS 4 is directed to a BRI data extension with BCCOS 1, the switch recognizes that this call probably needs a Modem Pooling conversion resource and inserts one. On the other hand, if the same call is directed to a voice only extension (BCCOS 0), the switch infers that this is a voice call and circuit switches the call. In either case, if the inference is not correct, the call would fail anyway. A data call completed to a voice terminal or a voice call to a data terminal will not work, whether a conversion resource is inserted or not. New BCCOSs can be created to deal with other circumstances.

ISDN—BRI Voice/Data Stations

Call processing for BRI voice / data stations works at two levels, at the switch level (like any other type of terminal) and at the interface level. This is because the BRI is an intellegent interface. With the voice / data station, each terminal should have its own extension number. That is, there should be one extension number for the voice terminal and a different extension number for the data terminal. When this is done, each extension number is assigned a BCCOS that is appropriate for the type of call (voice or data) that will be handled by that extension number. Two types of calls are possible that are not cut and dried: the ***unknown digital*** (BCCOS 3) and ***unknown analog*** (BCCOS 4) calls. Either of these types of calls could be either voice or data calls.

For ***unknown type calls,*** callers are given credit for knowing what they are doing. That is, if an unknown type call is directed to an extension number that is assigned BCCOS 0 (voice only), we assume that it is a voice call. Conversely if an unknown type call is directed to an extension number that is assigned a data BCCOS, it is treated as a data call. If these conclusions are incorrect, the call will fail.

If a ***voice call (BCCOS 0) is directed to the data extension number,*** call processing takes the opposite tack. In this case, it is clear that the data extension is unable to handle a voice call; therefore, the interface offers this call to the voice terminal, even though it was origionally directed to a data extension. The same type of handling is provided for a data call (BCCOS 1) directed to the voice extension number.

For a DCP originated data call that is set up using ***Voice Terminal Data Call Setup*** procedures, an entirely different problem can be encountered. If data preindication is not used, the call will be misdirected regardless of which extension number is used. This is because without data preindication the call will carry the BCCOS of the voice terminal. For additional information on BCCOS, see the Bearer Capability feature.

Switch Features and Services

Generic 2, voice and data features and services provided to BRI terminals are the same as the features and services available to DCP terminals. Additional capabilities and services, unique to ISDN are also available at the BRI terminal level.

Feature Differences

While all Generic 2 terminal features are available to ISDN—BRI terminals, there are differences in the way some of them work. Most differences are in the way the feature operates (internal to switch operations and call processing) and are not readily apparent to the user. Feature interactions in general are described in detail later in this chapter. T'he following specific differences are worthy of special note.

Data Call Setup

- ***Keyboard Dialing***

  The BRI data terminal operates in a "Command Mode" resulting in different operator procedures. See "User Operations" for details on ISDN—BRI Keyboard (Terminal) Dialing Procedures.

- ***Call Progress Monitoring and Control***

  Call progress monitoring and control messages are sent to a DCP data terminal (keyboard dialing operations) by the switch. Most corresponding messages displayed at a BRI data terminal are generated by the BRI data interface and are different from those that appear on DCP data terminals.

- ***Voice Terminal Data Call Setup***

  Voice Terminal Data Call Setup on ISDN—BRI terminals works in a completely different way than it does on DCP terminals. While there is a DATA button on the BRI voice terminal, it is an appearance button for the data extension. On a DCP voice terminal the DATA button is a feature button.

— Data Button Functions on DCP Terminals

The DATA button on a DCP terminal is a feature button used with voice terminal data call setup. As such, it performs the feature functions: ***One-Button Transfer, Return to Voice*** and ***Data Preindication.***

— Data Button Functions on BRl Terminals

On BRI terminals, the DATA button is an appearance button for the data terminal and not a feature button. The feature functions (One-Button Transfer, Return to Voice, and Data Preindication) are not used on BRI terminals. An operation that appears similar to one-button transfer can be performed; however, it is not the same. There is no return to voice or data preindication function.

— Additional Data Buttons

On DCP terminals, additional DATA buttons can be assigned. That is, a DCP voice terminal can have more than one DATA button and control more than one DCP data terminal. Also, more than one DATA button can be assigned for the same DCP data terminal.

On BRI terminals, only one DATA button can be assigned and it can be used only for the associated BRI data terminal (linked to the same terminal adapter ADM—T [Asynchronous Data Module—T Interface]). A BRI data terminal cannot be assigned to a DATA button on any non-associated terminal. For example, a BRI data terminal interfaced by a 7500 data module cannot be assigned to a DATA button (anywhere).

Also, DATA buttons for DCP data terminals cannot be assigned to BRI voice terminals.

Call Forwarding

ISDN—BRI data terminals can forward their calls, but cannot use Call Forwarding Off Net. They can also receive forwarded calls.

Dedicated Switch Connections

ISDN—BRI data stations can be assigned to Dedicated Switch connections; however, BRI voice terminals cannot.

Capabilities and Services Unique To lSDN—BRl Terminals

The following capabilities and services are specifically associated with ISDN—BRI terminals and are not generally available on DCP terminals.

Channel Negotiation

For ISDN—BRI, channel negotiation applies to calls between the switch and a BRI terminal. Two separate scenarios apply to BRI channel negotiation. When a call is ***originated by the BRl terminal,*** either a ***preferred*** or ***exclusive*** channel is indicated.

- ***Preferred Channel Option***

  If the preferred B-channel is available, it will be used for the call; otherwise the other B-channel will be used (if available).

- ***Exclusive Channel Option***

  With the exclusive channel option, if the requested B-channel is available it will be used; otherwise the call is denied.

Channel negotiation for BRI terminals is performed by the interface an is not user administrable.

B-Channel Reservation

The *reserved status* of a B-channel can affect how the switch will respond to a B-channel negotiation. When a BRI terminal has a voice call assigned (for example on Hold), the switch will attempt to keep one B-channel reserved for voice calls on that interface. While a channel is reserved for voice, all voice calls to that terminal will be given the reserved channel, independent of a "preferred" channel request from the terminal. An "exclusive" request for the other channel (non-reserved channel) will be rejected. The reserved channel is not available for data calls and only the reserved channel is available for voice calls.

Flow Control

The ISDN Flow Control enhancement (also known as Hyperactivity Management) is a defensive mechanism in Generic 2. It is based on the hyperactivity structure for DS1 introduced in System 85, Release 2, Version 4. The concern is that excessive message processing over ISDN facilities (both PRI and BRI) could overload the call processing capacity of the switch. This particular situation is referred to as ***applications hyperactivity.***

The flow control mechanism monitors ISDN D-channels and keeps track of the message flow rates by type of application (BRI facility, PRI facility, etc.). Actual message flow rates are checked against nominal standards for each facility type. These nominal standards are maintained and updated statistically by the switch as experience factors are developed.

If a particular ISDN D-channel significantly exceeds the standards for its type over a designated period of time, that facility is flagged as a hyperactivity suspect. If suspect status continues, an additional flag is raised and the offending facility is automatically removed from service and reported on a trouble audit.

AT&T ISDN—BRI Telephones

The ISDN—BRI feature, on the Generic 2 switch, requires the use of specific terminals that uses the 2B + D configured "T" interface. AT&T produces the 7500 series ISDN—BRI voice terminals that use the 2 B + D configured "T" interface. A "U" interface is used with some BRI terminals. Terminals configured for the "U" interface will not work on the Generic 2 switch.

7500 Series Telephones

The ISDN BMT (Basic Modular Telephone) 7505

The 7505 is an initializing, digital voice terminal that supports MIMs (Management Information Messages). An optional ADM (Asynchronous Data Module)-T for simultaneous voice and data operations is available. If used, the ADM-T is part of the 7505s base [TA (Terminal Adapter)]. Figure 65-4 shows the 7505 BMT.

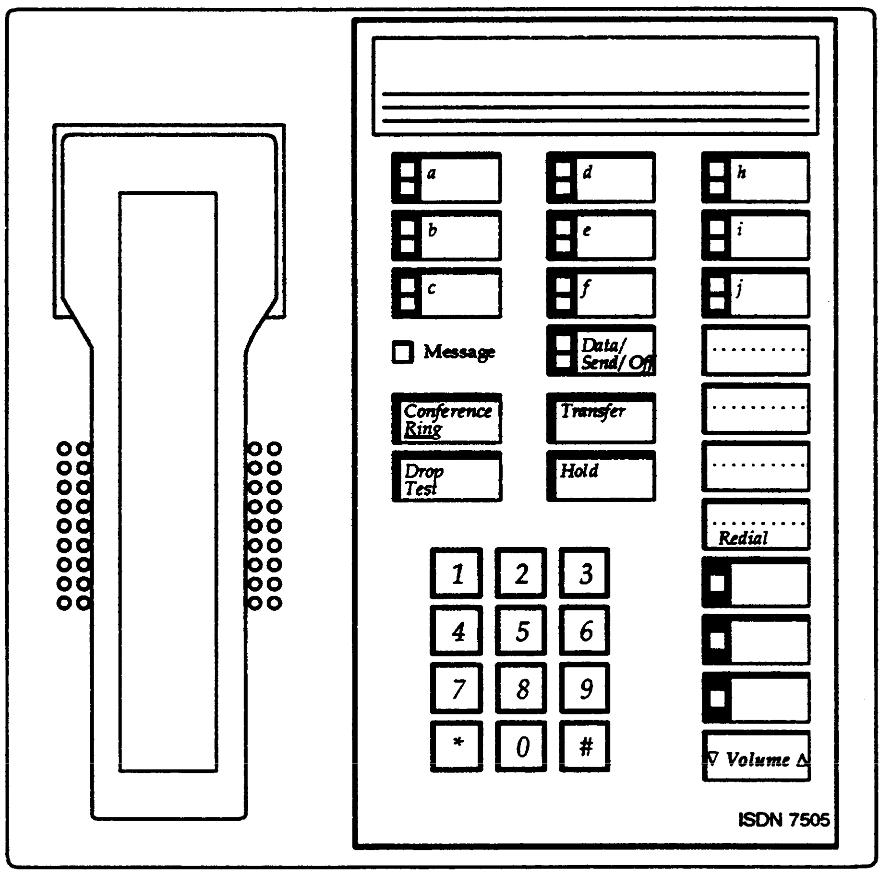


Figure 65-4. ISDN—BRI Voice Terminal 7505, BMT (Basic Modular Telephone)

As a voice terminal, the 7505 BMT provides the following features:

- Appearance/Feature Buttons:

  There are ten appearance or feature buttons with both in-use (green) and status (red) lamps. These can be used for call appearances or for administrable feature buttons. If the optional ADM-T is used for an associated data terminal, appearance button number 7 is used for the data appearance.

  When button number 7 is assigned as a data appearance, it becomes a dual function button. When used with the Shift/Select button, the second function is to set data call options.

- Shift/Select

The Shift button selects the operation of dual function buttons.

- Dual Function Fixed Feature Buttons:

There are two dual function, fixed feature buttons.

Conference / Ring

The Conference button is used to set up the Conference — Three Party feature. Used with the Shift / Select button, this button provides the ability to select a personal ringing pattern.

Drop / Test

The Drop button is used to disconnect from an active call drop from an established conference call. Used with the Shift / Select button, this button initiates an on demand ISDN—BRI self-test for the terminal.

- Single Function Fixed Feature Buttons:

There are two single function fixed feature buttons.

Transfer

The Transfer button is used to set up the Transfer feature.

Hold

The Hold button is used to place an active call appearance on hold. This can be used to answer another incoming call, place a call, or perform some other task. It places the active call in a state where the other party does not hear background sound (conversations). If available, Music on Hold is provided.

- Administrable, Dual Function Feature Buttons:

There are three administerable feature or function buttons. These can be assigned dual functions, providing for six administrable feature functions. These are single lamp (status) buttons and cannot be used for call appearances.

- Adjunct Jack:

An adjunct jack (located on the bottom of the set) is used to connect a voice terminal adjunct such as the 500A Headset Adapter for hands free operations.

- Message Waiting Lamp:

The Message Waiting Lamp functions with the Message Waiting — Automatic feature.

- Speakerphone:

There is a built in speakerphone and SPOKESMAN® loudspeaker. This is useful for hands free operation and for group calls and conferencing.

- Mute Button:

  A Mute button, used to control the microphone function of either the handset or the speakerphone. The status lamp of the Mute button shows (when lighted) when the mute function is in operation.

- Volume Control Button:

  A Volume button that controls the speaker volume for both the handset and the speakerphone.

- Speaker Button:

  A Speaker button that controls (turns on and off) the speakerphone function.

The ISDN MDT (Modular Display Terminal) 7506

The 7506 is functionally similar to the 7505 except that it has a built in Digital Display. This digital display provides a 2-line display field, with 24 characters per line. This allows the user to take advantage of the Display — Voice Terminal feature provided by the switch and, when available, the user-to-user information function called for in the ISDN standards. Figure 65-5 shows the 7506 MDT.

As a voice terminal, the 7506 MDT provides the following features:

- Appearance/Feature Buttons:

  There are ten appearance or feature buttons with both in use (green) and status (red) lamps. If the optional ADM-T is used (for an associated data terminal), button number 7 is used for the data appearance.

- Adjunct Jack:

  An adjunct jack (located on the bottom of the set) is used to connect a voice terminal adjunct such as the 500A Headset Adapter for hands free operations.

- Shift/Select Button:

  A Shift/Select button that selects the operation of dual function buttons. Dual function buttons include the following

  Redial

  Calls the Last Number Dialed from the terminal keypad. The second function of this button is assigned in system administration.

  Conference/Ring

  The Conference button is used to set up the Conference — Three Party feature. Used with the Shift/Select button, this button provides the ability to select a personal ringing pattern.

  Transfer/Clock

  The Transfer button is used to set up the Transfer feature. Used with the Shift/Select button, this button activates the Date/Time functions of the Display — Voice Terminal feature.

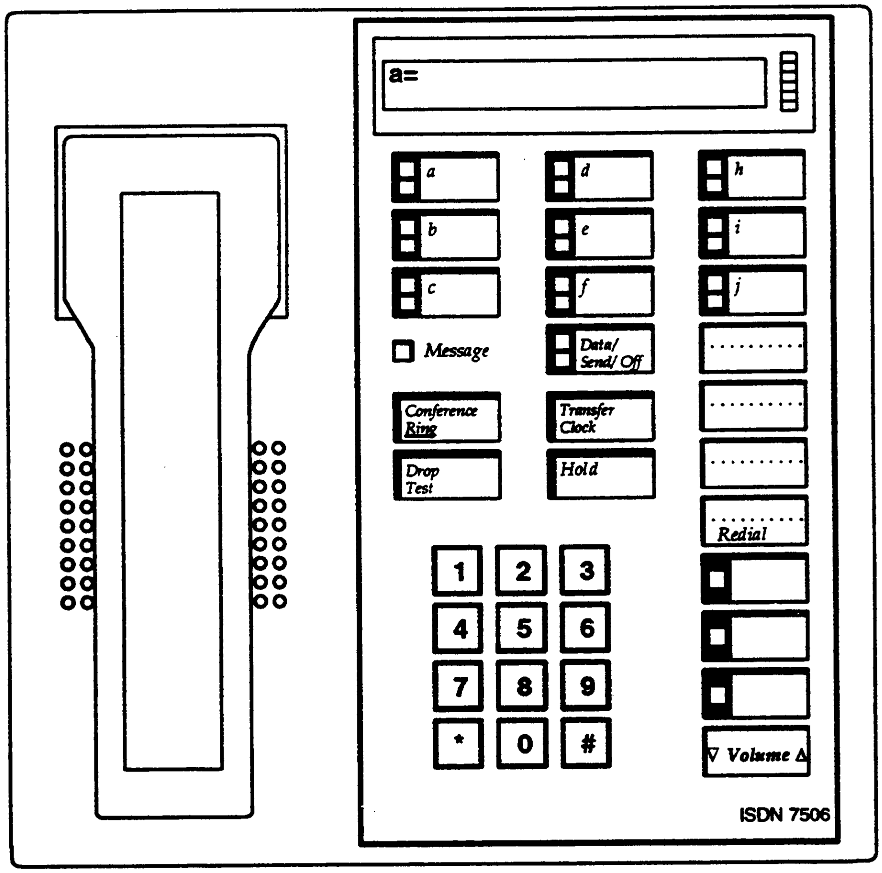


Figure 65-5. ISDN—BRI Voice Terminal 7506, MDT (Modular Display Telephone)

Drop/Test

The Drop button is used to disconnect from an active call drop from an established conference call. Used with the Shift/Select button, this button initiates anon demand ISDN—BRI self-test for the terminal.

- Administerable Feature/Function Buttons:

There are three administerable feature or function buttons. These can be assigned dual functions, providing for six administrable feature functions. They cannot be used for call appearances.

- Hold Button:

The Hold button is used to place an active call appearance on hold. The user can then answer another incoming call, place a call, or perform some other task without abandoning the first call. Hold places the active call in a state where the other party does not hear background sound (conversations). If available, Music on Hold is provided.

- Message Waiting Lamp:

  A Message Waiting Lamp that functions with the Message Waiting — Automatic feature

- Speakerphone:

  A built in speakerphone and SPOKESMAN® loudspeaker. The speakerphone supports hands free operation and is useful for group calls and conferencing.

- Mute Button:

  The Mute button is used to control the microphone function of both the handset and the speakerphone. The status lamp of the Mute button shows (when lighted) when the mute function is in operation.

- Volume Control Button:

  The Volume button controls the speaker volume for both the handset and the speakerphone.

- Speaker Button:

  The speaker button controls (turns on and off) the speakerphone function.

The ISDN IDT (Integrated Display Telephone) 7507

The 7507 IDT supports all of the features and services of the 7506 MDT and provides some of these at an enhanced level of service. Figure 65-6 shows the 7507 IDT. The 7507 IDT features the following capabilities and characteristics:

- Digital Display

  The display field of the 7507 is larger than that of the 7506 (2-lines with 40 characters pre line, as opposed to 2-lines with 24 characters per line), which allows more information to be displayed at a time.

- Appearance Feature Buttons

  There are 31 appearance or feature buttons with both in-use (green) and status (red) lamps. If the optional ADM-T is used for an associated data terminal, button number 31 is used for the data appearance.

- Adjunct Jack:

  An adjunct jack (located on the bottom of the set) is used to connect a voice terminal adjunct such as the 500A Headset Adapter for hands-free operations.

- Shift/Select Button:

  The Shift button that selects the operation of dual function buttons.

- Dual Function Fixed Feature Buttons:

  There are three dual function, fixed feature buttons: Conference/Ring Transfer/Clock, and Drop/Test.

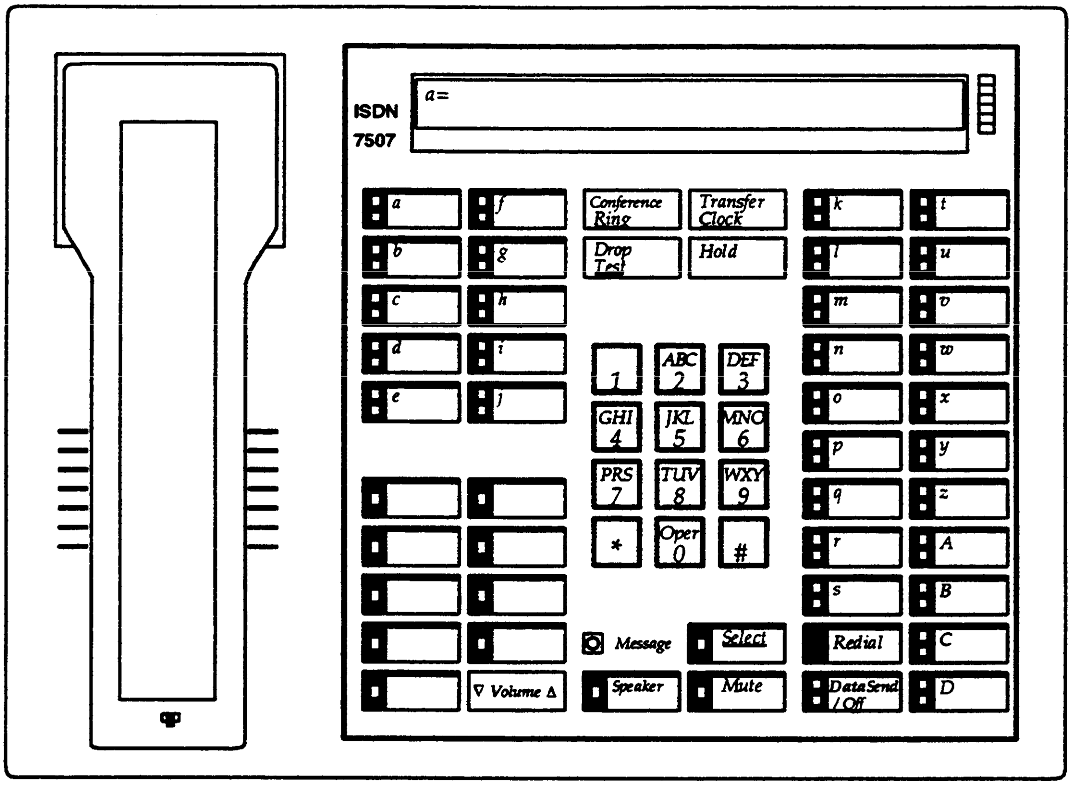


Figure 65-6. ISDN—BRI Voice Terminal 7507, IDT (Integrated Display Telephone)

- Hold Button:

  The Hold button is used to place an active call appearance on hold. This can be used to answer another incoming call, place a call, or perform some other task. It places the active call in a state where the other party does not hear background sound (conversations). If available, Music on Hold is provided.

- Message Waiting Lamp:

  The Message Waiting Lamp functions with the Message Waiting — Automatic feature.

- Speakerphone:

  A built in speakerphone and SPOKESMAN® loudspeaker

- Mute Button:

  A Mute button, used to control the microphone function of either the handset or the speakerphone. The status lamp of the Mute button shows (when lighted) when the mute function is in operation.

- Volume Control Button:

  The Volume button controls the speaker volume for both the handset and the speakerphone.

- Speaker Button:

  The Speaker button controls (turns on and off) the speakerphone function.

Table 65-B compares the capabilities and functions of each of the AT&T manufactured BRI voice terminals.

TABLE 65-B. Characteristics of AT&T BRI (Basic Rate Interface) Telephones

Feature or Characteristic	7505 BMT (Basic Modular Terminal)	7506 MDT (Modular Display Terminal)	7507 IDT (Modular Integrated Display Terminal)
Data Capable With Optional ADM	Yes	Yes	Yes
Display Capability	No	2-Lines 24 Characters	2-Lines 40 Characters
Terminal Set Personalized Ringing	Yes	Yes	Yes
Administerable Appearance or Feature Buttons	10	10	31
Administrable Dual Function Feature Buttons	3	3	No
Fixed, Dual Function Feature Buttons	2	3	3
Redial Button	Yes*	Yes*	Yes
Mute Button	Yes	Yes	Yes
Hold Button	Yes	Yes	Yes
Speakerphone	Yes	Yes	Yes
Message Waiting Lamp	Yes	Yes	Yes
Terminal Adjunct Jack	Yes	Yes	Yes

* Administerable dual function.

More detailed information on specific ISDN—BRI voice terminals is available in the Voice Terminals Reference Manual NNN-NNN-NNN, and in the following User's Manuals:

- 555-021 -701, ISDN 7505 Modular Terminal and 7506 Display Terminal
- 555-021 -702, ISDN 7507 Display Terminal.

NOTE: Some operations described in terminal equipment User's Manuals may not be available on the Generic 2 switch.

The ISDN Terminal Adapters

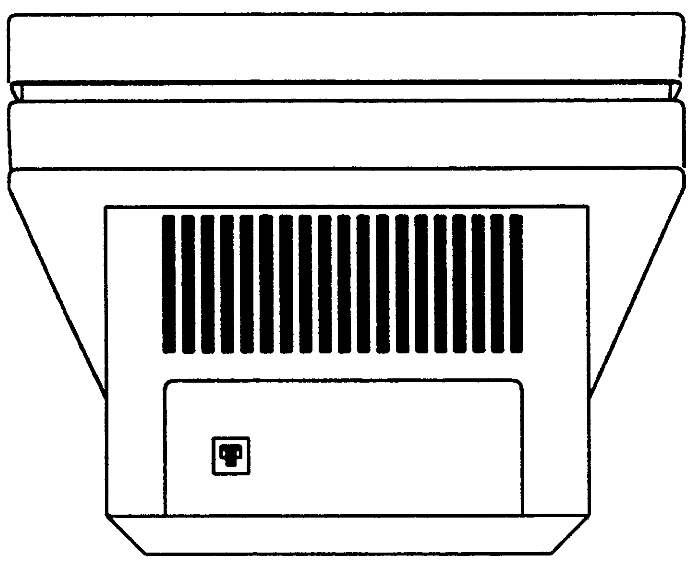
The ISDN TAs (Terminal Adapters) provide a station level ISDN interface for ISDN voice and data terminals. The two basic TAs come in the form of phone stands. They are the VOM-T and the ADM-T.

VOM-T (Voice Only Module with T Interface)

The ISDN VOM-T is an adjunct phone stand used with the ISDN—BRI voice terminals. It can be used with either the 7505, 7506, or 7507; and provides ISDN—BRI connectivity to the voice terminals that do not have an associated data terminal connected to the same interface port.

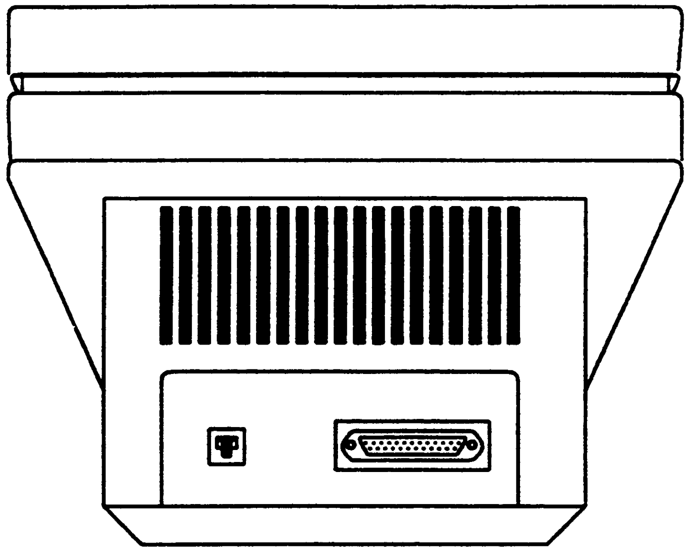
ADM-T (Asynchronous Data Module With T Interface)

The ISDN ADM-T is an optional adjunct used with the ISDN—BRI voice terminals. It can be used with either the 7505, 7506, or 7507; and provides ISDN—BRI data terminal connectivity to the BRI voice / data stations. The ISDN Terminal Adapters (both VOM-T and ADM-T) are shown in Figure 65-7.



BRI Phone Stand VOM-T (Voice Only Module)

Figure 65-7. ISDN—BRI Terminal Adapters



BRI Phone Stand ADM-T (Asynchronous Data Module)

Figure 65-7. ISDN—BRI Terminal Adapters (Continued)

Additional information on the ISDN—BRI terminal adapters is available in the User's Manual, 555-021-708.

NOTE: Some operations described in terminal equipment User's Manuals may not be available on the Generic 2 switch.

ISDN—BRI PC (Personal Computer Configurations)

Specific AT&T PCs and other IBM* compatible PCs can be provided with an ISDN—BRI connection. This is done through the use of a set of PC-to-PBX interface products called the PC/ISDN Platform. Up to four ISDN interface cards can be installed in the expansion bus of an appropriate PC. This provides up to four stations (each interface card provides a separate interface that requires its own ELL) for the PC. Note that each separate station may or may not have a separate extension number (see Appendix E). This arrangement can provide both voice and data connections for the PC and one or more associated voice-calling devices.

Associated voice-calling devices can include any of the 7500 series voice terminals described previously, or an AT&T R-Type replacement handset (PEC X10150), or a suitable headset such as the Plantronics StarSet† Series model StarMate† MH0228-3. The ISDN-PC arrangements and capabilities are described in greater detail under the PC Interface feature.

* Registered trademark of the IBM Corporation

† Registered trademark of the Plantronics Corp.

Non-initializing Terminal Type

The PC/ISDN Platform can be used in a stand alone (no associated voice terminal) configuration. When used this way, or with a handset or headset rather than one of the 7500 series voice terminals, the PC/ISDN Platform is a non-initializing terminal that does not support MIMs (Management Information Messages). This means that these "stations" do not go through an initialization procedure with the switch. It also means that they do not require a SPID, and that they do not initiate or respond to MIMs.

NOTE: When more than one PC/ISDN interface card is used on the same PC, each separate card provides a separate interface and is translated on the switch with its own ELL (Equipment Line Location) as a separate station. If one card has an associated 7500 series voice terminal, that station is translated as initializing and supporting MIMs. If additional interface cards (on the same PC) do not have associated 7500 series voice terminals, those stations are translated as non-initializing stations that do not support MIMs.

See the Interactions with Other Features section or the PC Interface feature for more details on these arrangements.

Stand Alone Data Module

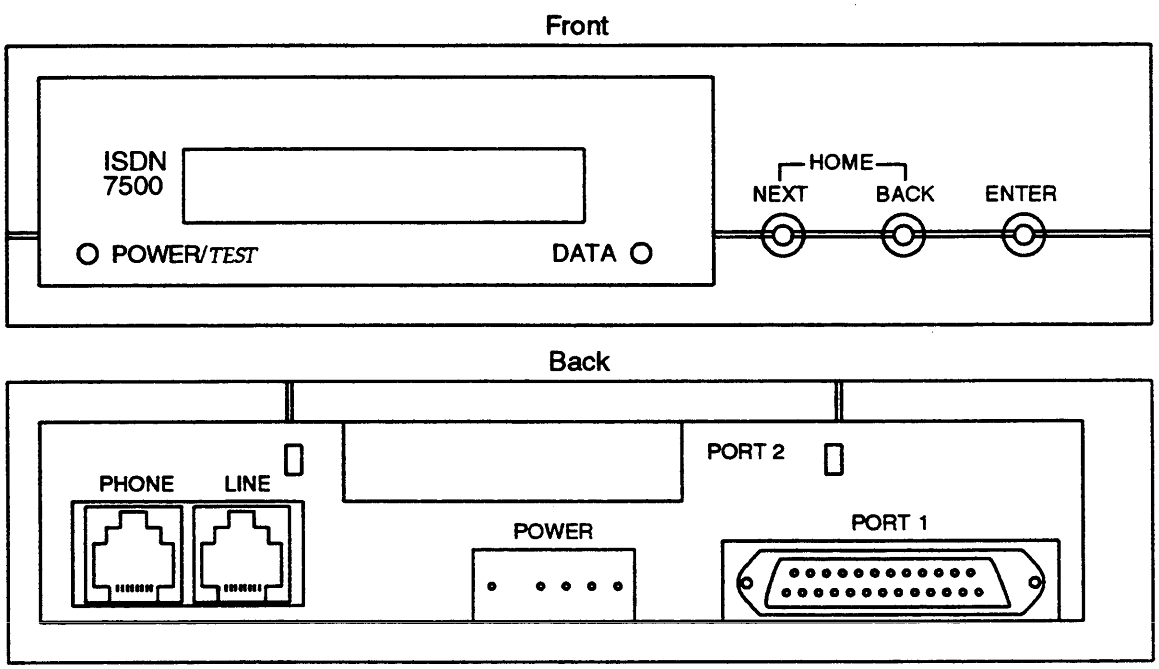


Figure 65-8. ISDN—BRI Data Module, 7500 Data Module

7500 Data Module

The AT&T 7500* data module provides ISDN—BRI connectivity for data terminals that do not need an associated voice terminal. The 7500 is a MIMs supporting, initializing terminal type. The 7500 is a stand alone data module that can be used in a rack mounted or table top configuration. Figure 65-8 shows the front and rear panel views of the 7500. On the rear panel view, a phone jack is visible. This is provided for future use and is not functional at the present time (linked voice terminal operations are not available with the 7500).

The 7500 uses the EIA RS-232D interface standard for connection to the supported data terminal. This connection is compatible with data terminals that use the older RS-232C standard interface. Depending on internal configuration options, the 7500 can also support RS-366 or V.S5 interface connectivity.

The 7500 provides BRI data interface support on the Generic 2 switch as DCE (Data Communications Equipment) in either the asynchronous or synchronous mode.

The 7500 data module supports the standard data rates (300 through 19.2K bps) in the asynchronous modes. In synchronous mode, the following data rates are supported

- 1200 bps
- 2400 bps
- 4800 bps
- 9600 bps
- 19.2K bps
- 48K bps
- 56K bps
- 64K bps.

The 7500 data module provides asynchronous data calling services that are similar to those provided by the ADM-T. The 7500 supports data modes 0 through 3 and mode 2/3 adaptive asynchronous data operations.

In the absence of an associated voice terminal, most data call functions that can be performed by the voice terminal in an ADM—T arrangement, can be performed using the 7500 data module.

* The 7500 Data Module (also known as the Z7500) functions as a UDM (Universal Data Module) on the 5ESS switch While the hardware is the same, it does not function as a UDM on the Generic 2 switch

Additional information on the 7500 data module is available in the 7500 Data Module User's Manual, 555-021-703.

NOTE: Some operations described in terminal equipment User's Manual are not initially available on the Generic 2 switch.

Other Manufacturers ISDN—BRI Terminal Equipment

The ISDN—BRI is an open standard (see Appendix G), therefore, ISDN—BRI terminal equipment manufactured by companies other than AT&T should work on the Generic 2 switch. The CCITT standard provides for basic connectivity; however, it does not specify functionality in all cases. While other manufacturers equipment should work, AT&T cannot warrant functionality and cannot support any such equipment until it has been specifically verified for compatibility by AT&T.

User Operations

Voice Terminal Operations

In general, user operations for an ISDN—BRI telephones are similar to user operations for a DCP voice terminal (7400 Series). Differences are primarily in the internal operations of a feature or call processing, and are generally transparent to the user. One noticeable difference is that 7500 series ISDN—BRI telephones do not have Reed or Disconnect buttons. A recall button can be administered to an available feature button.

For BRI terminal user operations that are not given here, see the appropriate terminal user's guide or the applicable feature description.

To place a voice call:

1. Go off-hook. [An idle appearance is selected automatically (see the **Multiappearance Preselection and Preference** feature), dial tone is heard, and the user is connected to the selected appearance.]

   or

   Press an idle appearance button [the red in-use lamp lights] and then go off-hook. [The green status lamp lights, dial tone is heard, and the user is connected to the selected appearance.]

2. Dial the desired extension number or feature dial access code. [The appropriate call-progress tones (ringing tone, confirmation tone, reorder tone, second dial tone, etc.) are heard]
3. When the connection is no longer needed, go on-hook. [The green status lamp goes out. The red in-use lamp may or may not go out depending on the appearance used and the multiappearance preference selection.

To answer an incoming voice call:

Go off-hook [ringing appearance will be selected automatically]

or

Select appearance by pressing the desired alerting appearance button (green status lamp is flashing) and then go off-hook. [Ringing stops (if the appearance is ringing), the green status lamp lights steadily, and the user is connected with the calling party.]

To set personalized ringing:

1. Press the **[SELECT]** button (on 7507) or **[SHIFT/SELECT]** button (on 7505 or 7506). [The green status lamp lights].
2. Press the **[CONFERENCE RING]** button. [Current ringing pattern is heard.]
3. Press the **[*]** key (on the dialing pad) or the **[CONFERENCE RING]** button to hear the next alternate ringing pattern.

   Repeat this step until the desired ringing pattern is heard.
4. Press the **[#]** key (on the dialing pad) to save the selected ringing pattern. [Two burst confirmation ringing tones are heard.]

Dual-Function Button Operation

To use the primary function (first listed) of a dual-function button:

Press the desired function button. [First listed function is selected (i.e., Conference on the Conference/Ring button). Appropriate switch response is returned depending on the feature or function selected.]

To use the alternate function (second listed) on a dual-function button:

1. Press the **[SELECT]** button (on 7507) or **[SHIFT SELECT]** button (on 7505 or 7506). [The green status lamp lights.]
2. Press the desired function button. [Alternate (second listed) function is selected (i.e., Ring on the Conference/Ring button). Appropriated switch response is returned depending on the feature or function selected.]

To record the SPID (Service Profile Identifier) on a BRI terminal:

CAUTION: *Performing this operation after your BRI terminal has been installed could result in loss of service.*

1. Press the **[SELECT]** button (on 7507) or **[SHIFT SELECT]** button (on 7505 or 7506). [The green status lamp lights.]
2. Press the **[MUTE]** button. [On display capable terminals, `[PROGRAM]` appears on the display.]

3. Enter **[I][D]** on the touch-tone key pad.

GHI 4	DEF 3

[On display capable terminals, the following messages appear on the display:

[SPID:NNNNNNN] and [[X] Change [#] Save] .]

4. Enter the SPID (unique number of 10 digits or less), if a new SPID is needed.
5. Press **[#]** on the touch-tone key pad to record new SPID or retain old SPID if a new number was not entered. [Display returns to normal configuration; green status lamp (on **[SELECT]** , or **[SHIFT SELECT]** button goes dark.]

To place a data call from the BRI Telephone:

(Applies only to ISDN—BRI telephones equipped with the optional ADM-T terminal adapter.)

1. Make sure the data terminal is turned on and options are set correctly. (Do not go off-hook.)
2. Press the **[DATA/SEND/OFF]** button. [The red in-use lamp lights. On display capable terminals [DIAL] appears on the display.]
3. Dial the extension number or dial access code desired (a mnemonic cannot be entered on a voice terminal).
4. Press the **[DATA/SEND/OFF]** button again. [The red in-use lamp stays on and the green status lamp blinks. Green status lamp may respond on a second appearance button, other than the Data/Send/Off button depending on administered options and appearance selected for the call. When the far end returns answer tone, the green status lamp goes to a steady on state and the red in-use lamp remains in the steady on state. The data terminal screen displays:

   [CONNECTED - MODE 2] and [FAR END SPEED - 19200]

   NOTE: The line appearance seized may not be the one assigned to the data button. Also, the figures for data mode (mode 2) and data rate (19200) shown in these examples are used for example purposes only. The mode and rate that will appear on your display will reflect the actual state of the far end of your connection and may differ from the examples shown.

5. Press **[RETURN]** on the data terminal keyboard and proceed with appropriate log on procedures.

To end a data call from the telephone:

Press the **[DATA/SEND/OFF]** button. [Both the red and green status lamps go dark.]

To set data calling options from the BRI telephone:

1. Press the **[SELECT]** button (on 7507) or **[SHIFT SELECT]** button (on 7505 or 7506).
2. Press the **[DATA/SEND/OFF]** button. [The red in-use lamp lights.]
3. Press the **[1]** key (on the dialing pad) for B-channel mode 2.
4. Press one of the following dialing pad keys to set the corresponding data rate:

1	for 300 bps
ABC 2	for 1200 bps
DEF 3	for 2400 bps
GHI 4	for 4800 bps
JKL 5	for 9600 bps
MNO 6	for 19.2 Kbps.

5. Press one of the following dialing pad keys to set the corresponding parity option:

1	for no parity
ABC 2	for mark parity
DEF 3	for even parity
GHI 4	for odd parity.

6. Press one of the following dialing pad keys to set the desired local mode:

1	for CMD
ABC 2	for off
DEF 3	for AT.

NOTE: If the local mode is set to "off" the data options cannot be changed from the data terminal.

Data Terminal Operations

ISDN—BRI data terminals operate in the "Command" mode. This results in some differences in ***Keyboard Dialing*** operations between ISDN—BRI and DCP data terminals.

Also, ISDN—BRI data terminals can operated in either asynchronous or synchronous modes. Only the asynchronous DCE operations common to both ISDN—BRI and DCP data terminals are described here. For information on other types of operation, see the Users Manual for the data module being used

To place a data call from the BRI data terminal:

1. Make sure the data terminal is turned on and options are set correctly.

   If the `[CMD:]` prompt does not appear on the screen, press the **[BREAK]** or **[RETURN]** key and then enter "`AT`" to make certain that speed and parity are correct.

2. Enter "`dial`" (or "`d`") and the number to be called (extension number, mnemonic, access code, etc.)

3. Press the **[RETURN]** key. [Call progress messages, `[CALLING nnn]` `[Type E to end call:]` , `[RINGING]` , `[NNN NNN]` , `[CONNECTED - MODE 2]` `[FAR END SPEED - 19200]` are displayed].

4. Press the **[RETURN]** key. [Far end login prompt or other response is retuned.]

To end a data call from the data terminal:

1. Enter the attention sequence "+++" [Terminal returns to command mode and `[CALL STATUS:]` , `[Data Call Active]` , `[Type H for help.]` and `[CMD:]` appear on the screen.]

2. Enter "end" or "e". [The call progress messages `[CLEARING]` , `[ENDED]` , `[Call Status: Idle]` , `[Type H for help.]` appear on the screen.],

or

You can log off from the far end (host etc.). This will cause the distant end to terminate the call. [The progerss messages [CLEARING], [ENDED], [FAR END REQUESTED], [Call Status: Idle], [Type H for help.] appear on the screen.]

CAUTION: *The above methods (using the attention sequence and end command or logging off at the far end) provide a quick and "graceful" means of ending a data call from an ISDN—BRI terminal. A popular alternative method, tuming the data terminal itself off,* **is not recommended.** *Wile this alternative method works well for DCP data terminals it does not work well from BRI data terminals. With BRI, tuming the power off on the data terminal does not actully end the call. The call remains up (no disconnect) until the system (far end or switch, whichever occurs first) times out.*

NOTE: To end a call during call setup (while still in the command mode, before [CONNECT] is displayed), enter "end" or "e".

To use Last Number Dialed from a data terminal:

1. At the [CMD:] prompt, enter "redial"
2. Press the **[RETURN]** key. [Call progress messages, [CALLING nnn] [Type E to end call:], [RINGING], [NNN NNN], [CONNECTED - MODE 2] [FAR END SPEED - 19200] are displayed].
3. Press the **[RETURN]** key. [Far end login prompt or other response is retuned].

Considerations

Bearer Capability Class of Service

The predefined BCCOS "0" is the default BCCOS for voice extensions. For BRI Voice/Data stations where both voice and data calls use the same extension number, the predefined BCCOS "2" is used In some cases, the predefined BCCOSs need to be modified (or a different BCCOS needs to be assigned) for certain ISDN—BRI stations.

While the CCITT, ISDN protocols provide for standardization in general form and character, they do not specify all message structures and signaling conventions. The currently available 7500 series BRI voice terminals expect certain types of messages for incoming voice calls. Some sources of incoming ISDN voice calls do not provide the expected messages. Consequently, incoming calls from these sources do not terminate to a voice extension on a 7500 series terminal. Specifically, this problem has been encountered on ISDN calls from *international sources* and on calls originated at some non-AT&T switches. Some calls that have been processed through the *interworking* function have also experienced this problem. This problem applies only to early versions of ISDN—BRI terminals and does not occur with analog or DCP extensions that also use BCCOS 0.

On DEFINITY Generic 2.1, Issue 3.0 and later switches, if this problem is experienced, the BCCOS for these terminals can be changed to accommodate these calls. This can be done in either of two ways:

- ***By creating new BCCOSs for affected terminals***

  1. Copy the default BCCOSs (BCCOS 0 and 2 if used) to new BCCOSs using Procedure 014, Words 1 and 2.
  2. For the new BCCOSs, use Procedure 014, Word 1, to change the encode value in Field 5 to "3".
  3. Assign the new BCCOS to the affected extensions using Procedure 000, Word 3, Field 5.

- ***By modifying the predefined BCCOSs***

  For the affected BCCOSs, use Procedure 014, Word 1, to change the encode value in Field 5 to "3".

Class of Service

While ISDN—BRI terminals can use the same classes of service that would normally be assigned to a DCP terminal of like characteristics, the ISDN muting options (which are assigned as part of the class of service) should be at the least "ISDN preferred". This will be true, even if few ISDN—PRI routes are available.

Equipment Location

The ISDN—BRI is available on Generic 2, Universal Modules and XE Modules. BRI is not available on traditional modules.

Legal Considerations

Local laws vary considerably and are constantly subject to change. Legislation concerning the non-voluntary disclosure of information has been proposed in some areas. It is possible that passing, recording, or displaying some information provided by ISDN Messages may be, or become contrary to local privacy regulations. It is the responsibility of local management (switch administrator, communications director, etc.) to ensure that available features and services are not used in such a way as to violate local laws and regulations.

SPID (Service Profile Identifier)

The SPID is a unique number (10 digits or less) assigned to each BRI terminal (end point). The SPID is used by the switch during the initialization process as a recognition tool for initializing BRI terminals. Note that some BRI terminals are non-initializing. These terminals do not require a SPID.

The SPID is stored in two separate locations: one in switch memory, and the other in the BRI terminal itself. When a BRI station is initialized (after a power failure or when an initializing terminal is replaced or moved), the switch and terminal exchange SPIDs as a means of recognition. If the SPID recorded on the switch does not match the SPID recorded at the terminal, initialization will fail.

For ease in accomplishing terminal exchange and testing it is strongly recommended that the primary extension number be used as the SPID for BRI terminals. Where the same extension number is used as the primary extension for more than one station, a suffix should be used to distinguish between two stations. A two or three digit suffix will suffice to accommodate all occurrences of shared extensions.

With a voice / data station, where two separate terminals are assigned to the same station (ELL), only one SPID is needed. Actually, only the telephone (7500 series) is initializing, and it is this terminal that needs the SPID and participates in the initialization process with the switch.

Non-Initializing Terminals

The BRI version of the PC Interface feature, the PC / ISDN Platform, in a stand alone configuration (no associated 7500 series telephone) is a non-initializing terminal and does not require a SPID. In fact, in the stand alone configuration, this type of terminal cannot be assigned a SPID.

Service SPID

The Service SPID is an administrable option that allows maintenance and service personnel to use a common SPID administered to both the switch and one or more test telephones for testing any BRI station position. This is an optional capability that must be specifically translated on the switch.

When administered on the switch and on the test terminal, the test terminal can be plugged into any BRI port and will initialize properly. The test terminal will also acquire the characteristics (class of service, feature assignments, etc.) of the terminal that was originally translated for that port.

> **NOTE:** The Service SPID provides limited functionality and should not be assigned to terminals that are intended for general use.

Mode 0 Data Calls

During the initial releases of Generic 2, ISDN—BRI data terminals are not compatible with DCP interfaced data end point (either terminals or host ports) when Mode 0 is used. This is due to differences in handshaking routines.

Interactions With Other Features

Abbreviated Dialing

The Abbreviated Dialing feature works the same for ISDN—BRI terminals as it does for DCP terminals on the Generic 2. Either type of interface can be configured as a Voice / Data Station. When this is done, both the telephone and the data terminal have the same ELL (Equipment Line Location). The Abbreviated Dialing feature is assigned to the ELL and therefore treats both terminals the same. BRI terminals (both voice and data) located on the same port share the same Abbreviated Dialing lists and permissions.

This creates the same situation as for a DCP Voice / Data Station [voice terminal with an attached data module (DTDM)]. In both cases, a data terminal and a telephone have access to (or share) the same abbreviated dialing lists. Care must be taken to ensure a clear separation between data call and voice call list items. This is particularly important where the Data Call Setup Default Dialing or the Hot Line Service features are involved. For data terminals (including ISDN—BRI data terminals) with ***default dialing*** or ***Hot Line*** service assigned, the first index item of Abbreviated Dialing List A is used for the default dialing or Hotline destination number. The user must be aware of this relationship, especially when abbreviated dialing list items are programmed from the voice terminal. For example, it is possible to change the default dialing destination for the data terminal from the voice terminal by reprogramming item 1 of list A. This could have confusing results if users don't fully understand what they are doing.

NOTE: If an ISDN—BRI Voice / Data Station were administered with ***default dialing*** assigned to the data terminal and ***Hot Line*** service assigned to the telephone, both terminals would automatically dial the same extension number. Hot Line Service, should not be assigned to an ISDN—BRI Voice / Data Station.

Redial Button

ISDN—BRI 7500 series telephones are equipped with Redial buttons. These buttons provide a service similar to the LND (Last Number Dialed) feature, except when Abbreviated Dialing is used.

For calls made using Abbreviated Dialing the LND feature records and redials station-to-station numbers called using list-stored or button-stored numbers.

When the Redial button is used rather than the LND feature, only numbers actually dialed from the keypad are recorded and redialed.

ACCUNET™ Service Interface

The ISDN—BRI feature is compatible with ACCUNET Service Interface with one restriction. The ISDN—BRI ADM-T (Asynchronous Data Module with T Interface) does not support either the 56 Kbps data rate or mode 1 data operations in the Generic 2 time frame and cannot be used for ACCUNET Service Interface. Both the 7500 data module and the PC / ISDN Interface, will work with the ACCUNET Service Interface feature. BRI stations using either the 7500 data module or the PC / ISDN Interface can be used for ACCUNET Service Interface calls.

AAR (Automatic Alternate Routing)

ISDN—BRI is a line side feature in Generic 2. BRI terminals access off-net features and services through the ***interworking*** function or via the ISDN—PRI feature. Once off-net service is accessed, the AAR feature provides the appropriate routing for private network calls on Generic 2.1 switches, in the same way it supports other terminals.

BCCOS (Bearer Capability Class of Service)

In Generic 2, routing is based on BCCOS, and two methods are available for routing the call.

1. The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (such as Mode 2 data, 1200 bps, restricted channel). If a match is found and a trunk is available, the action taken is "circuit switch the call".

2. If a match is not found the algorithm attempts to connect the call to a preference for which the action to take ***is not*** "block the call". With currently available options, this would be a preference where the action is "insert modem pool".

ACD (Automatic Call Distribution)

ISDN—BRI terminals can be used in an ACD split. However, a BRI terminal cannot be used to perform the service observing function on switches prior to Generic 2.1, Issue 2.0. A BRI Data Line can be assigned to an ACD agent station, however, the first appearance on the station should never be used for a data line. This is because many of the ACD call distribution checks are based on the status of the first appearance on the set If the first appearance is a data line, it will not be available for a voice call and the station will be skipped by the ACD call distribution algorithm.

ARS (Automatic Route Selection)

ISDN—BRI is a line side feature in Generic 2. BRI terminals access off-net features and services through the ***interworking*** function or via the ISDN—PRI feature. Once off-net service is accessed, the ARS feature provides routing for public network calls, on Generic 2.1 switches.

Bearer Capability

The Bearer Capability feature interacts with the ISDN—BRI feature to provide optimum call muting and call handling. The Bearer Capability feature provides default information on call support requirements and capabilities of extensions, trunks, and routing pattern preferences, in the form of an administered BCCOS (Bearer Capability Class of Service). For non-ISDN facilities, this default information used to match information contained in the ISDN call setup message. This information is used by call handling facilities including the ISDN—BRI to make call processing decisions.

Bridged Call

For voice calls, the Bridged Call feature works on BRI terminals and appearances in the same way as for other terminals. However, unlike DCP data call appearances, BRI data appearances can be shared (assigned bridged images). The bridged images of BRI data appearances (on other than the "Home Terminal") can be used for monitoring the state of the data appearance (active, idle, alerting) and under certain circumstances to initiate a data call. See the discussion earlier in this chapter under Data Prime Line for details.

Call Coverage

The Call Coverage feature is compatible with ISDN—BRI and works with BRI telephones in the same way as it does for other digital voice terminals. ISDN—BRI telephones can be used for the principal's voice terminal and can also be in the coverage path (covering user voice terminals).

Call Detail Recording

Calls to or from ISDN—BRI terminals are recorded in essentially the same way as calls for any other type of terminal on the DEFINITY Generic 2 switch.

Call Forwarding — Busy and Don't Answer

ISDN—BRI terminals are fully compatible with the Call Forwarding — Busy and Don't Answer feature. This includes ISDN—BRI data terminals as well as voice terminals except that data terminals cannot forward off-net.

Call Forwarding — Don't Answer

ISDN—BRI terminals are fully compatible with the Call Forwarding — Don't Answer feature. This includes ISDN—BRI data terminals as well as voice terminals.

Call Forwarding — Follow Me

ISDN—BRI terminals are fully compatible with the Call Forwarding — Follow Me feature. This includes ISDN—BRI data terminals as well as voice terminals except that data terminals cannot forward off-net

Call Pickup

ISDN—BRI data terminals cannot be assigned as members of Call Pickup Groups.

Conference — Attendant Five Party

ISDN—BRI telephones work with the Conference — Attendant Five Party feature in the same way as other types of voice terminals on Generic 2.

Conference — Attendant Six Party

ISDN—BRI voice terminals work with the Conference — Attendant Six Party feature in the same way as other types of voice terminals on Generic 2.

Conference — Three Party

All AT&T ISDN—BRI 7500 series telephones are equipped with a fixed Conference feature button. For these telephones, the Conference — Three Party feature works the same as it does for a DCP voice terminal.

Data Call Setup

The Data Call Setup feature works for an ISDN—BRI data terminal in much the same way as for a DCP data terminal. ***Terminal Dialing, Mnemonic Dialing,*** and ***Hot Line*** all work the same way, except for the interposition of the ***interworking*** function which is transparent to the user.

Default dialing

Default dialing also works the same for ISDN—BRI data terminals except that the default dialing number is stored in the associated Abbreviated Dialing List A as index item 1, rather than in a separate default dialing storage location.

Voice Terminal Data Call Setup

Voice Terminal Data Call Setup works differently for ISDN—BRI Voice / Data Stations than for DCP terminals. For outgoing calls, the nature of the call (voice or data) is indicated by the ISDN call setup message. The distinction is handled by the ISDN message content.

- ***The Data Button***

  For ISDN—BRI Voice / Data Stations, a data button exists on the telephone; however, it is an appearance button for the associated data terminal and not a DATA function button as on DCP voice terminals.

  Unlike DCP DATA buttons, BRI DATA buttons are assigned only to Voice / Data Stations (telephones with an ADM-T terminal adapter) and only one DATA button can be assigned. DATA buttons for non-associated data terminals (terminals assigned to a different interface) are not allowed.

- ***Procedures***

  There are ***Voice Terminal Data Call Setup*** procedures (see User Operations) that can be used with ISDN—BRI Voice / Data Stations, however, they work differently and have different limitations than those for DCP Voice / Data Stations.

- ***Data Button Functions***

  The DCP data button functions ***(One Button Transfer, Preindication, and Return to Voice)*** do not exist on ISDN—BRI voice / data stations.

Call Progress Monitoring and Control

With AT&T ISDN—BRI terminals, most displays will be similar to or the same as the displays that appear on DCP terminals; however, BRI displays are generated by the BRI interface from display messages received either from the network or from the switch.

> **NOTE:** The process described here applies only to AT&T manufactured ISDN—BRI equipment. The CCITT recommendations do not specify terminal functions and other vendors equipment may or may not perform in a like manner.

In Generic 2, three call progress messages are provided to ISDN—BRI data terminals by the switch (like DCP data terminals). Figure 65-9 shows these three messages.

Switch Generated Messages
for ISDN—BRI Data Terminals

Position in queue message.

05 IN QUEUE

Modem Pool member identification message.

RINGING 070 008

Hot Line notification message.

HOT LINE

Figure 65-9. ISDN—BRI Call Monitoring and Control Messages

DSC (Dedicated Switch Connections)

ISDN—BRI data terminals can be used with the Dedicated Switch connections feature. ISDN—BRI telephones cannot be used for DSC When an ISDN—BRI data terminal is assigned to a DSC, there can be only one appearance administered to the station (interface) and there can be no shared (bridged) images of that appearance. The DSC may or may not be designated as a data appearance. In the case of a DSC, since there is only one image of the appearance, the data appearance designation (field 10, Procedure 052, Word 1) has no effect.

DMI (Digital Multiplexed Interface)

ISDN—BRI data terminals can access the DMI feature either through ***interworking,*** or directly if the DMI port uses the MOS (Message Oriented Signaling) version. If direct connections are available (local switch or via the ISDN—PRI feature to a DMI MOS host), ***ISDN End-to-End*** connectivity can be provided.

Dial Access to Attendant

The Dial Access to Attendant feature works the same for ISDN—BRI telephones as it does for other Generic 2 voice terminals.

Display — Voice Terminal

The Display — Voice Terminal feature works for ISDN—BRI terminals with a display capability. There are some differences in how and where some display messages are generated, however, these are generally user transparent. The displays shown on BRI terminals and those on display-capable DCP terminals are generally the same. One noticeable difference occurs in timing on outgoing (trunk) calls. With BRI terminals, the display is initially controlled by the local (at the terminal) interface. For outgoing trunk calls, display information generated by sources other than the local interface (for example, the local switch) is not displayed immediately. In some cases, the display maybe delayed for several seconds. For example, when the Abbreviated Dialing feature is used to place an outgoing trunk call, the dialed number display information is generated by the switch rather than the terminal. In this case, the display information does not appear on the terminal display until answer supervision is returned. Depending on the calling situation, this delay can be in excess of 10 seconds.

DCS (Distributed Communications System)

ISDN—BRI terminals located on a Generic 2 switch can be used for DCS network calls just like any other terminal on the switch.

FX (Foreign Exchange) Access

FX trunk service is available to ISDN—BRI terminals either through the ***interworking*** function, or via the ISDN—PRI feature if the FX trunks are set up as ISDN trunks.

Host Computer Access

ISDN—BRI data terminals can call Host Computer Access ports. The ***interworking*** function supports these connections. However, ISDN—BRI data modules cannot be used with the Host Computer Access feature to provide an ISDN interface for the computer ports. The Host Computer Access feature remains a DCP feature

Hot Line

The ISDN—BRI feature is compatible with the Hot Line feature. Hot Line is assigned on a per extension basis. Therefore, if a BRI Voice / Data Station is given two separate extension numbers, one for the voice appearance and another for the data appearance, the Hot Line assignment can be made for the data extension without affecting the voice appearance.

ISN (Information Systems Network) Interface

ISDN—BRI data stations use the ISN Interface feature in the same way as DCP data stations. The ISN Interface feature uses DCP, therefore ISDN end-to-end connections are not supported.

ISDN—PRI (Primary Rate Interface)

The ISDN—BRI feature is a line side feature in Generic 2. For off-net ISDN capabilities and services, BRI terminals can place calls through the ISDN—PRI feature. If the called extension is also an ISDN—BRI terminal or a DMI MOS port, this can provide ISDN ***End-to-End Connectivity.***

Intercept Treatment

The Intercept Treatment feature works with ISDN—BRI terminals in the same way as it does for other types of terminals on the DEFINITY Generic 2 switch.

Last Number Dialed

The Last Number Dialed feature stores and redials the digits dialed during an ISDN call. On an AT&T BRI Telephone, this function is performed by the terminal itself. The REDIAL button on an AT&T BRI 7500 series telephone is a fixed feature button that dials the last number stored by the terminal. An administerable feature button to activate the Last Number Dialed feature provided by the switch can also be assigned if desired.

Leave Word Calling

The Leave Word Calling feature works for ISDN—BRI terminals on local calls and within a DCS as it does for other type terminals. Leave Word Calling does not work on calls to or from the public network or for a non-DCS private network.

Modem Pooling

Modem Pooling supports ISDN—BRI terminals in the same way it does DCP terminals, through the ***interworking*** function. In System 85, the decision to use a modem pooling conversion resource is made by the switch, based on translations for the trunk and line involved in the call. With Generic 2, and when an ISDN—BRI end point is involved, this decision is based on a much broader spectrum of options. Call related information is obtained from the best available source as follows:

- ***Call Setup Messages:*** Call control information contained in the ISDN call setup message associated with each specific call is the primary Source of information for call handeling requirements. Call control setup messages (originating from ISDN facilities) contain IEs that indicate the type of call (i.e., voice, Mode 0 data, etc.), data rate, and other information needed to identify required resources including Modern Pooling conversion resources.
- ***Optional Query:*** AT&T data modules* (both BRI and PRI) have the ability to respond to requests for additional information from the switch. For information that is needed but not available in a specific call setup message, this optional query ability is used
- ***Default BCCOS Values:*** The last resort for determining resources needed for a specific call is the customer administered BCCOS assigned to the origination points, carrier and support facilities, and termination points. This BCCOS provides default requirements and characteristics for specific facilities. The default BCCOSs are associated with the facility and not with a specific call.

* The query response capability mayor may not be available from non-AT&T data modules.

If call processing must, for any reason, use only the BCCOS default values, the resulting call handling will be consistent with call handling provided by an R2 V4 switch for the same call. In effect, the default processing for Generic 2 equates to the basic call processing provided by the R2 V4 switch.

ISDN—BRI data modules cannot be used as members of a Modem Pooling conversion resource.

Multiappearance Preselection and Preference

The Multiappearance Preselection and Preference feature works the same for an ISDN—BRI telephones as it does for any other voice terminal on the Generic 2 switch.

Off-Premises Data-Only Extensions

The ISDN—BRI feature is a line side only feature in Generic 2. ISDN—BRI terminals cannot be used as off-premise terminals.

Personal Central Office Line

The Personal Central Office Line feature is compatible with the ISDN—BRI feature. When available (based on network and local CO switch capabilities), ISDN Personal CO Lines can be provided to BRI stations.

PC Interface

With Generic 2.1, Issue 2.0, a new ISDN—BRI interface product, the PC/ISDN Platform, provides an ISDN—BRI configuration for the PC Interface feature.

connections between terminals using the PC Interface feature (either DCP or ISDN—BRI) and other ISDN—BRI terminals are supported either directly for BRI to BRI connections, or through the interworking function for BRI to DCP or BRI to analog connections.

The PC/ISDN Platform offers some characteristics that differ from other BRI station products or other PC Interface products.

- With the PC/ISDN Platform, up to four PC/ISDN Interface Cards can be installed in a single PC.
- Each separate PC/ISDN Interface Card requires its own separate port or ELL (Equipment Line Location), thus forming a separate station on the switch.
- Each separate card (station) requires its own separate and independent translation on the switch, even though the same PC may already appear as a station on the switch through administration for a different interface card.
- Station Administration:
  - — ISDN—BRI PC stations should be assigned using General Terminal Administration, Procedure 050, Word 1 and Word 2. There are no standard predefined terminal types for ISDN—BRI PC stations, and aliasing is not recommended.

— When the PC/ISDN Platform is installed in a stand alone configuration (no associated with 7500 series telephone), some necessary considerations are not readily apparent.

Option:

In Procedure 050, Word 1, Field 2 requires that certain options be identified. For a data only configuration, encode "6" should be used. While encode "0" specifies "Data Only" this is not applicable to a BRI data station as it actually means DCP Data Only. Encode "6" is used to provide for one or more call appearances with data capabilities even though no telephone is attached.

ISDN MIM:

As a stand alone data station (no telephone), or with a non-initializing voice calling device such as a handset or headset, the PC/ISDN Platform is a non-initializing station that does not support MIMs (Management Information Messages). This configuration should be translated in Procedure 050, Word 1, with Field 6 set to "0" ("None" means no initialization and no MIMs Support).

When the PC/ISDN Platform is installed with an associated 7500 series telephone, this configuration is translated as an initializing station that supports MIMs. This is because of the characteristics of the 7500 series telephones and has nothing to do with the PC/ISDN platform. A locally defined terminal type, using General Terminal Administration, should still be used to take advantage of the capabilities of the PC/ISDN Platform.

— Other specific characteristics of the locally defined terminal type will depend on the specific telephone manager applications software used with the PC.

— With the ability to install more than one PC/ISDN Interface Card in the same PC, and with each interface card being translated as a separate station, it is entirely possible for the same PC to appear on the switch as both a non-initializing station with no MIMs support and an initializing station that supports MIMs, at the same time.

Precedence Calling

The ISDN—BRI feature is compatible with the Precedence Calling feature. ISDN—BRI terminals can be used to place Precedence Calling calls and calls active on ISDN—BRI terminals can be preempted by calls of a higher precedence.

Priority Calling

The Priority Calling feature works the same for ISDN—BRI terminals as it does for other multiappearance terminals on the Generic 2 switch, except for data appearances. A Priority Calling call will not ring at an idle data appearance (including prime data lines) even if all other appearances are busy.

Queuing

The Queuing feature works for ISDN—BRI terminals a little differently than it does for DCP or analog terminals. Ordinarily, a call will queue (if necessary) on the first-choice trunk group in the appropriate muting pattern. For calls from ISDN—BRI terminals, the COS (Class of Service) may indicate ISDN facilities only. In this case the call will queue on the first ISDN trunk group with the needed bearer capability in the pattern. This may or may not be in the first-choice trunk group.

Ringing—Abbreviated and Delayed

The Ringing—Abbreviated and Delayed feature is fully compatible with ISDN—BRI. BRI telephones can be assigned this feature in the same way as other voice terminals.

Ringing Transfer

The Ringing Transfer feature is fully compatible with ISDN—BRI. BRI telephones can be assigned this feature in the same way as other voice terminals.

Tenant Services

ISDN—BRI terminals can be used within a Tenant Services arrangement in the same way as DCP terminals.

Transfer

All AT&T ISDN—BRI 7500 series telephones are equipped with a fixed Transfer feature button. For these telephones, the Transfer feature works the same as it does for a DCP voice terminal.

Trunk Verification — Voice Terminal

The Trunk Verification — Voice Terminal feature is fully compatible with the ISDN—BRI feature. That is, an ISDN—BRI telephone can be used for trunk verification. However, an appearance that is designated a data appearance cannot be used for any voice calling functions, including trunk verification.

Unattended Console Service — Call Answer from Any Voice Terminal

ISDN—BRI telephones can be used with the Unattended Console Service — Call Answer from any voice terminal feature. The ***interworking*** function supports calls from other than ISDN trunks.

Unattended Console Service — Preselected Call Routing

ISDN—BRI telephones can be used with the Unattended Console Service — Preselected Call Routing feature. The ***interworking*** function supports calls from other than ISDN trunks.

WCR (World Class Routing)

The World Class Routing feature provides network routing service for off-premises calls on DEFINITY Generic 2.2 switches. ISDN—BRI is a line side feature in Generic 2. BRI terminals access off-net features and services through the ***interworking*** function or via the ISDN—PRI feature. Once off-net service is accessed, the WCR feature provides the appropriate routing for network calls in the same way it supports other terminals.

Restricting Feature Use

The principal means of restriction for the ISDN—BRI feature is through the Restrictions — Voice Terminal Feature. This feature is used to assign the COS (Class of Service) restrictions to all DEFINITY Generic 2 terminals, including ISDN—BRI terminals.

Attendant Control of Voice Terminals can also be used to restrict access to or from ISDN—BRI terminals, however, certain cautions must be observed. The attendant console in Generic 2 is not an ISDN terminal. ISDN calls routed through the attendant may loose ISDN end-to-end connectivity. Also, with ISDN—BRI voice / data stations, the voice and data terminals share the same extension number. As a result, data calls as well as voice calls will route to the attendant. The attendant has no way of handling a data call.

Hardware Requirements

Universal Module

The ISDN—BRI feature requires the use of Universal or XE Modules. The XE Module is functionally equivalent to the Universal Module and is available in the Generic 2.1, Issue 3.0 and later time frame. The circuit packs for this feature are designed for use in the Universal and XE Modules and no equivalent circuit packs exist for traditional modules.

Circuit Packs

- TN556 ISDN—BRI Line Circuit

  Provides 12 ports for ISDN—BRI line appearances. Each port supports two B-channels and one D-channel.

Voice Terminals

The 7500 Series Telephones

- ISDN BMT (Basic Modular Telephone) 7505
- ISDN MDT (Modular Display Telephone) 7506
- ISDN IDT (Integrated Display Telephone) 7507

Terminal Adapters

Terminal adapters provide protocol conversion from standard non-ISDN interfaces (for example, RS-232D or X.25) to ISDN interfaces. Terminal adapters form the base of the

7500 series telephones. All BRI telephones used with the ISDN—BRI feature on Generic 2 require a terminal adapter. The following terminal adapters are available for use with Generic 2 switch:

- ADM-T (Asynchronous Data Module)

  The ADM-T is an asynchronous data module with an ISDN T interface for B-channel connectivity. It can be used with the 7505, 7506, and 7507 BRI terminals.

- VOM-T (Voice Only Module)

  The VOM-T is used for 7500 Series telephones that are not assigned an associated data terminal. It can be used with the 7505, 7506, or 7507 BRI telephones.

Note that while there are two *functional types* of terminal adapters, the ADM-T and the VOM-T, they are not all interchangeable. The terminal adapters that fit on the 7505 and the 7506, do not fit on the 7507 (which is a physically larger terminal).

Data Module

- ISDN—BRI 7500 Data Module

  In addition to the ADM-T mentioned above, the ISDN—BRI 7500 Data Module is available for use with stand alone (no associated voice terminal) data end point applications.

The ISDN—PC

With the ISDN—PC, an expansion card (circuit board), called the PC/ISDN Interface Card, provides the physical and firmware interface circuitry for BRI connectivity. The ISDN—PC provides several different configurations for a PC and/or one or more telephones or voice call devices. These arrangements are described in detail in the PC Interface feature.

Feature Administration

The ISDN—BRI feature is assigned on a per system basis, and within the switch, on a per module basis. ISDN—BRI is available only in Generic 2, and only on Universal Modules.

The BRI feature is administered using the DEFINITY Manager II.

ADMINISTRATION PROCEDURES INTEGRATED SERVICES DIGITAL NETWORK — BASIC RATE INTERFACE		
PROCEDURE	**WORD**	**PURPOSE**
000	1	Assigns extension number and COS (Class of Service) to an ELL (Equipment Line Location).
000	3	Assigns single terminal translations, including Hot Line, type of messaging to use for Dedicated Switch connections, audible or lamp type Automatic Message Waiting, and bearer capability class of service. (See also Procedure 354, Word 3.)
010	4	Assigns characteristics to an extension class of service, including ISDN muting. Word 4 is used to specify ISDN access and routing.
014	1	Used to assignor modify Bearer Capability Classes of Service (BCCOSs). For certain ISDN—BRI voice extensions, a BCCOS may need to be modified to receive international and certain other incoming voice calls by entering encode "3" in field 5.
050	1 & 2	Used for General Terminal Administration, to define new multiappearance terminals and Data Modules, including PC/ISDN Platform configurations.
051	1	Assigns Multiappearance Terminal and Data Module translations to an ELL (Equipment Line Location), including Keyboard (Terminal) Dialing for data terminals and BRI TEI (Terminal Equipment Identifier). BRI terminals should be assigned to even numbered circuits only (field 5).
051	2	Used to assign SPID for an ISDN—BRI terminal and the Service SPID (when activated) to the switch.

(Continued)

ADMINISTRATION PROCEDURES *(Continued)* INTEGRATED SERVICES DIGITAL NETWORK — BASIC RATE INTERFACE		
PROCEDURE	**WORD**	**PURPOSE**
052	1	Used to assign extension numbers and characteristics to an ELL (Equipment Line Location). For BRI terminals with a Data Appearance or Prime Data Line Field 10 (Line Type) is used to specify these special appearance. The following encodes apply: 3 = Prime data line 4 = Data appearance (non-prime). Field 12 (Home Terminal) is used to assign home terminal designation. There must be a home terminal for BRI data appearances and prime data lines.
052	2	Used to assign the extension and appearance characteristics to an ELL and appearance button.
059	1	Used to assign Abbreviate Dialing List A to an ELL (Equipment Line Location). Abbreviated Dialing List A assignment is a prequisite for Default Dialing.
059	2	Used to assign characters to item 1 of Abbreviated Dialing List A. Item 1 of List A is used as the destination address for Default Dialing.
059	4	Turns Default Dialing on or off for ISDN—BRI Data Terminals. This is done by adding a button without characters (fields 8 through 13 dashed).
262	1	Assigns timer parameters for D-channel signaling.
262	3	Assigns Hyperactivity Management (Flow Control).
275	3	Assigns system class of service features including Service SPID (Field 17). NOTE Service SPID should not be used for normal operation.
275	4	Assigns system class of service features including ISDN. Encode: 1 in Field 14
276	1	Assigns feature group class of service permissions including Service SPID (G2, 2.1 Issue 2.0). Encode: 1 in Field 11
NOTE: ISDN should be "turned on" using Procedure 275, Word 4, before the BRI terminals are translated. Otherwise the BRI terminals will not initialize until there is a hard processor swap or reboot.		

ISDN — PRI (Primary Rate Interface)

Description

The ISDN—PRI feature provides switch-to-switch ISDN capability to the DEFINITY Generic 2. The PRI feature, does not extend ISDN connectivity to the terminal level. It does, provide signaling and service advantages between switches. The ISDN—PRI feature provides external ISDN connections for ISDN—BRI terminals. The ISDN—PRI and BRI features together provide ISDN ***End-to-End Connectivity.*** Basic ISDN—PRI connections available are shown in Figure 66-1.

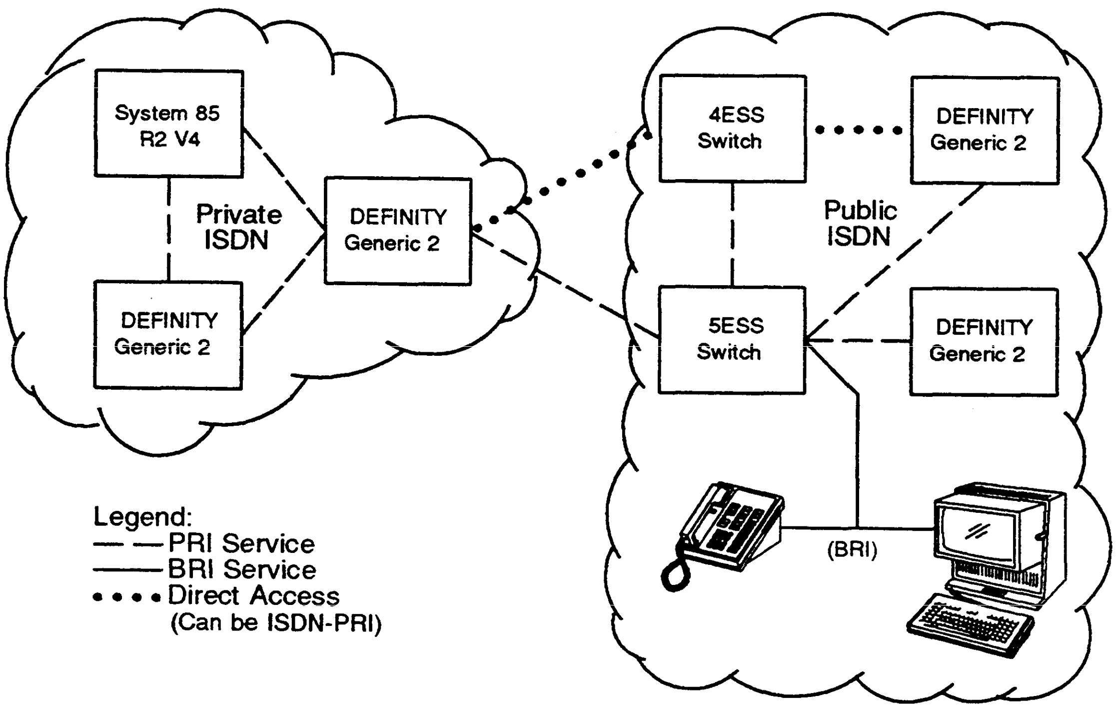


Figure 66-l. ISDN—PRI Connections

Feature History and Development

ISDN—PRI was first introduced on System 85, Release 2, Version 4. Enhancements added for DEFINITY Generic 2 include the following:

- Expanded and enhanced bearer capability
- Flow control
- Expanded IEs and Codesets
- Codeset conversion capability
- Administrable NSF (Network-Specific Facilities)

- Non-facility associated signaling
- D-channel backup.

Station Level Functions

With ISDN—PRI, individual stations work within the network provided by the local switch. Features and services available to each station are provided by the local switch and not directly by the network. ISDN—BRI stations and DMI MOS computer ports can have end-to-end ISDN connectivity to the station level; however, DCP (Digital Communications Protocol) stations, providing end-to-end digital connectivity, do not have end-to-end ISDN connectivity. Analog terminals have neither direct (end-to-end) digital nor ISDN connectivity.

Basic Configuration

The PRI (in the North American version) provides twenty-four, 64 Kbps channels. ‘These are normally arranged as 23 "B" (bearer) channels and one "D" (data) channel for common control signaling*. In ISDN terminology, this is called a **"23 B plus D"** configuration. ***Note that this is the North American version.*** In Europe, channel arrangement is different because the standard digital carriers currently used in Europe and North America are different. The North American arrangement corresponds to the North American DS1 format with common channel signaling, while the European version corresponds to the equivalent digital carrier used in Europe.

In the North American version, the main difference between ISDN and the standard versions of DS1 is signaling. ISDN uses MOS (Message Oriented Signaling), while DS1 traditionally uses a form of bit-oriented signaling.

Interworking

DEFINITY Generic 2 Originated Calls

Calls originating on a non-ISDN terminal of the DEFINITY Generic 2 switch (and non-ISDN calls coming into the DEFINITY Generic 2 switch) are handled in a standard manner by the conventional call processing software. These calls can use all DEFINITY Generic 2 features and services allowed by their class of service. When the switch is ready to pass calls to an ISDN outgoing trunk, ISDN call processing software is notified and ISDN routines are called into service.

The ISDN call processing converts DCP S-channel signals into the needed D channel messages and performs the ISDN functions and services, such as **Call-by-Call Service Selection and Channel Negotiation** (ISDN capabilities and services are described later in this section). The ISDN call processing software also converts information available to the

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* The NFAS (Non-Facility Associated Signaling) option provides an alternative to the 23B + D configuration. NFAS is discussed later.

switch, such as calling number and calling party name to ISDN messages when appropriate. It also converts the same kind of information from the distant switch (if available) to DCP "S" channel format and passes this information to the calling station (or called station on incoming calls) for use with the Display—Voice Terminal feature. When the ISDN connection has been set up, the end-to-end connection is made, and the call proceeds like a standard call as far as the user is concerned.

Incoming ISDN Calls

For incoming calls from ISDN facilities, the interworking process works in the same way as for outgoing calls, but in a reverse order. The incoming call is first processed by the ISDN call processing software. The standard switch software is notified by the ISDN layer that a call is ready for completion, and the standard call processing software sets up and services the switch side facilities (stations or non-ISDN trunk connections) in the same way as a conventional incoming call. All switch features and services that would normally be available (based on COS) are provided. Again, information such as calling number and calling party name is processed into the proper formats (DCP S channel messages for switch side stations and ISDN messages for the ISDN side of the call) and passed to the calling and called stations when appropriate.

Developments and Enhancements in DEFINITY Generic 2

Flow Control

The ISDN Flow Control enhancement in DEFINITY Generic 2 is a defensive mechanism based on the DS1 hyperactivity structure introduced in System 85, Release 2, Version 4. The concern is that excessive message processing over ISDN facilities (both PRI and BRI) could overload the call processing capacity of the switch. This particular situation is referred to as ***applications hyperactivity.***

The flow control mechanism monitors ISDN facilities and keeps track of the message flow rates by type of application [BRI facility, PRI facility, NFAS (Non-Facility Associated Signaling) etc.]. Actual message flow rates are checked against standards for each facility type. These standards are maintained and updated statistically by the switch as experience factors develop.

If a particular ISDN facility significantly exceeds the standards for its type over a designated period of time, that facility is flagged as a hyperactivity suspect If suspect status continues, an additional flag is raised and the offending facility is automatically removal from service and reported on a trouble audit.

There are no plans at this time to retrofit ISDN Flow Control to System 85, Release 2, Version 4 switches.

Codeset Conversion

The Need For Conversion

ISDN is an evolving protocol and evolution dictates change. Evolutionary changes have taken place in ISDN messaging between System 85, Release 2, Version 4 and Generic 2. New and enhanced ISDN capabilities and functions have resulted in an increase in the number and variety of ISDN messages that must be passed between switches and the number of IEs available for messaging purposes. For example, the number of ***User-to-User*** IEs (codeSet 7) approximately doubled between R2V4 and Generic 2. IEs that were contained in codeset 7 for System 85, Release 2, Version 4 have been moved to codeset 6. This change is not limited to the Generic 2 switch. The AT&T ISDN—PRI specification has changed making this change effective across the entire AT&T product line.

This migration of codepoints from codeset 7 to codeset 6 creates a communications problem between System 85, Release 2, Version 4 and Generic 2 switches. It is necessary to convert codepoints between codesets 6 and 7 on ISDN—PRI spans that connect System 85, Release 2, Version 4 and Generic 2 switches. This conversion is also needed on ISDN—PRI spare between Generic 2 switches and 4 ESS Toll Offices that use the 4E11 switch (the 4E11 uses the same ISDN codesets as System 85, Release 2, Version 4. Codepoints requiring this conversion are shown in Table 66-A

TABLE 66-A. Codepoints Requiring Codeset Conversion

Description	System 85, R 2, V 4		DEFINITY Generic 2	
	Code Set	IE Number	Code Set	IE Number
Display (Name)	7	40	6	40
Traveling Class Mark	7	8	6	8
Packet Mode IEs				
Link layer parameters	7	1	6	64
Packet layer parameters	7	2	6	62
Logical link identification	7	4	6	26

The Solution

Codeset conversion is accomplished through codeset/codepoint mapping to a group of tables in switch memory on the DEFINITY Generic 2 switch. This mapping is administered on a per interface basis using Procedure 280, Word 1 and Procedure 262, Word 3. Specifically:

- Messages from a System 85, Release 2, Version 4 (or 4E11) switch, that contain IEs from codeset 7 must have the reassigned IEs converted to codeset 6 IEs so that they can be correctly interpreted on a Generic 2 switch.

- Messages to a System 85, Release 2, Version 4 or 4E11 switch that contain codeset 6 IEs that appear in codeset 7 of the distant switch must also be modified so that the messages will be properly interpreted at the receiving switch.

Other Manufacturers Switches

The ***codeset/codepoint*** conversion capability is also useful on ISDN—PRI spans connecting to other manufacturers' switching equipment where codesets 6 and 7 (and very possibly specfic IEs in other codesets) will probably be structured differently. This application is encountered in networking arrangements between Generic 2 switches and Northern Telecom toll office switches (See 555-037-241, to be published). Eventually all manufacturers of telecommunications equipment can be expected to enter the ISDN market. Each manufacturer has considerable freedom to define codepoints, especially in codeset 7. If another manufacturer specifies a different IE with information that is recognized and usable (such as calling party name), codeset mapping allows the Generic 2 switch to process this IE to a codepoint that will be handled appropriately. As with the Version 4 switch, codeset mapping allows Generic 2 IEs in outgoing messages to be converted to a form that can be handled appropriately by the other manufacturer's switch.

Bearer Capability Expansion

Bearer capability is used on ISDN calls primarily to support data calling requirements. Bearer capability is a much less critical factor for voice calls. The scope and function of bearer capability has been expanded in Generic 2.

In System 85, Release 2, Version 4, bearer capability is assigned to ISDN facilities as a part of COS (Class of Service) and is used with the AAR and ARS features to select routes for ISDN calls. Bearer capability codes identify the types of ISDN calls that a specific ISDN facility can support. A bearer capability classes of service is used to identify the trunking services required by a specific ISDN calling facility.

In Generic 2, the power and versatility of ISDN messaging is used to determine bearer capability needs for call muting, both externally (over trunks) and internally (for example, Modem Pooling). Call type identification and resource requirements are based on the best available information, which is obtained as follows:

- ***Call Setup Messages:***

  Call control information in the form of ISDN call setup messages associated with each specific call is the primary source of information on protocol and call handling facility requirements. Call control setup messages (originating from ISDN facilities) contain IEs that indicate the type of call, protocol used, data rate, and other information needed to identify required resources (for example, the Bearer Capability IE and Low Level Compatibility IE).

- ***Optional Query:***

  Data modules, (both BRI and DCP) have the ability to respond to queries for information from the switch. For information that is needed but not available in a specific call setup message (for example, synchronous or asynchronous), this optional query ability is used.

- ***Default Values:***

  The last resort for determining resources needed for a specific call is the customer administered BCCOS (Bearer Capability Class of Service). A BCCOS is assigned to extensions (lines), trunks (including Host Access and Modem Pooling trunks), and routing patterns. This BCCOS provides default requirements and characteristics. Note that these default characteristics and requirements are associated with the end point or preference and not a specific call.

If call processing must, for any reason, use only the predefinned BCCOS values, the resulting Generic 2 handling will be consistent with call handling by a System 85, Release 2, Version 4 switch for the same call. In effect, the defaults for Generic 2 equate to the basic bearer capability handling provided by the System 85, Release 2, Version 4 switch. Table 66-B lists the default BCCOSs for Generic 2 and, to the extent possible, shows the compatable System 85, Release 2, Version 4 bearer capability codes.

TABLE 66-B. Default Bearer Capability Classes of Service

Default BCCOS DEFINITY G2	BCC System 85 R2V4	Type of Call Supported
0	(0)	Voice only*
1	2	Mode 2 Data
2	—	BRI Voice/Data
3	—	Unknown Digital
4	(0)	Unknown Analog
5	(0)	Voice Grade Data*
6	4	Mode 0 Data
7	1	Mode 1 Data
8	3	Mode 3 Data

* Bearer services available on System 85, Release 2, Version 4 but in a single BCC.

Part of the Bearer Capability feature function in Generic 2 is the specification of switch actions for specific routing preferences. These specific switch actions are administrable using Procedure 014, Word 1, Fields 4 through 13. For a given BCCOS one of three actions must be specified:

- Circuit switch the call
- Insert a Modem Pooling conversion resource
- Block the call.

Call routing over ISDN—PRI trunk groups (or any other type of trunk group) is controlled, to some extent by these switch action specifications. The muting features (AAR, ARS, or WCR) attempt to route a call over a preference where the switch action called for is ***circuit switch the call.*** If there are no trunks available that provide circuit switched service, an available trunk that calls for ***insert a modem pooling conversion resource*** will be used. If there are no trunks available that provide either of these actions, the call is blocked (unless another option such as raising the FRL or queuing the call is still available). The switch action ***block the call,*** will always deny use of a particular trunk group to calls with that BCCOS.

Administrable NSF (Network-Specific Facilities) Values

An administrable NSF value is a number between 1 and 511. The values from 1 through 511 are user definable through administration on DEFINITY Generic 2 switches. NSF values are assigned to routing preferences. These NSF values are associated in switch memory with specific network capabilities and services and used to generate the ISDN NSF IE.

The ***NSF IE*** is a part of the ISDN call setup message (codeset 0). This IE indicates to the NSO (Network Service Office) what network features and services are to be used for calls routed over specific preferences. These network features and services need not be ISDN unique. For example, WATS (Wide Area Telecommunications Service) is a service that can be designated by an NSF. However, the NSF IE is ISDN unique and apples only to ISDN—PRI preferences.

System 85, Release 2, Version 4

In System 85, Release 2, Version 4, NSF values are fixed in a table (not administrable). NSF values are assignable to AAR and ARS patterns, but their meaning (the network feature or service they specify) cannot be changed. For example, the NSF value of 352 specifies that SDN (Software Defined Network), a networking service offered by AT&T Communications, be used for the call. When 352 is assigned to an AAR preference that contains ISDN—PRI trunks, the call setup message that is generated when those trunks are used will indicate to the NSO that this is a SDN call. The System 85, Release 2, Version 4 switch administrator can assign the NSF value 352 to specific AAR preferences that are to be used for SDN calls but cannot change the meaning of this NSF value.

Generic 2

In DEFINITY Generic 2, NSF values are assigned to routing preferences in much the same way as they are in System 85, Release 2, Version 4. However, in Generic 2, the meaning of NSF values, or perhaps more specifically the NSF values themselves that are assigned to specific network features and services, are also user administrable. That is, the switch administrator can change the meaning of NSF value 352 from SDN to something else like ETN, and assign a different value such as 421 to the SDN service.

This capability gives the user increased flexibility in terms of specifying various network capabilities and services to be used by different preference. However, it also places more responsibility on the switch administrator for the performance of the switch and the availability of network services.

Initial NSF Values

When a new DEFINITY Generic 2 switch is shipped from the factory, it comes without ***NSF values*** assigned. If the customer specifies the NSF values that are wanted, TRACS (Translation Recovery, Additions, and Conversions System) can load these values on the initial tape. Otherwise, the switch administrator must establish these values before they can be used in a routing preference.

For a System 85, Release 2, Version 4 switch being upgraded to DEFINITY Generic 2, the Version 4 fixed NSF values that are in use will be perpetuated but become user administerable.

NSF Values For ACCUNET Switched Digital Service

There have been two fixed NSF values for ACCUNET Switched Digital service on System 85, R2 V4 switchs. These are either 357 or 999 depending on the vintage of the 4 ESS switch used at the supporting Network Service Office. The NSF value for ACCUNET Switched Digital Service is 999 for NSO (Network Service Office) switches prior to 4E13 and 357 for 4E13 and later switches. Most, if not all NSOs have been converted to 4E13 or later and the use of NSF 999 should no longer be necessary.

NSF Value Applications

The NSF Value is used with the applicable routing feature as part of the ***Preference*** specification. The NSF is also used for traffic measurement and analysis purposes and can be recorded on the CDR (Call Detail Record).

The NSF value assigned to a routing preference is used to generate the NSF IE which is then sent as part of the call setup message to identify to the ISDN (external), network capabilities and services that are to be used for a call. This application is the basis for Call-by-Call Service Selection.

Other Interexchange Carriers

The NSF values used by carrier other than AT&T, when these carriers offer ISDN services, may differ from those used in the AT&T network. The administrable NSF values will allow customers to modify their NSF value translations to accommodate values used by other carriers.

Network Capabilities and Services

Network capabilities and services that can be requested using the NSF IE are limited to those specifically offered by the serving network. For the AT&T Communications network, these are classified as either ***features*** or ***services.*** Currentl y available network features and services are listed in Table 66-C along with the applicable NSF values used in System 85, Release 2, Version 4.

TABLE 66-C. System 85, Release 2, Version 4, NSF Values

NSF Services Values Used System 85, Release 2, Version 4	
NSF Value	**Service**
	WATS (Out-WATS)*
33	Band 0
34	Band 1
35	Band 2
36	Band 3
|	
|	etc.
|	
286	Band 253
287	Band 254
288	Band 255
352	SDN Service
353	MEGACOM 800 Service
354	MEGACOM Service
355	In WATS (800 Service)
356	WATS Service (Maximal)
357	ACCUNET Service (If NSO is 4E13 or later)
359	International 800 Service
360	700 Service
361	Direct Access to 800 Service
362	ETN (Electronic Tandem Network)
363	Private Line Service

* Parameterized Service.

TABLE 66-C. System 85, Release 2, Version 4, NSF Values (Contd)

NSF Feature Values Used System 85, Release 2, Version 4	
NSF Value	**Feature**
	CPN (Calling Party Number)*
320	Preferred
322	Exclusive
	BN (Billing Number)*
321	Preferred
323	Exclusive
324	Operator
325	Prescribed Common Carrier Operator

Binary Versus Parameterized Features and Services

Network features and services are further categorized as either binary or parametrized. ***A binary feature or service,*** in ISDN terminology, is a feature or service with fixed characteristics. That is, a feature or service that is available in one form, and one form only. ***A parametrized feature or service*** is one with an optional range or scope available. The only ***parametrized*** feature or service currently available is WATS Service. This service is offered on a basis of "Bands" which define the range of service available. These "bands" are the ***parameters*** of WATS service.

Future Value Added

The real advantage in being able to define (or redefine) the NSF value comes from the fact that ISDN is a protocol that is still evolving. This means that new and improved network capabilities and services will be developed and offered as the networks evolve. The ability to define new NSF values and to redefine existing NSF values allows the user to keep pace with and take advantage of these new and improved developments in network capabilities and services.

Call-by-Call Service Selection

Call-by-call service selection is a capability of an ISDN that allows the same ISDN—PRI trunk group to be used for different network services on a call-by-call basis. Call-by-call Service Selection is implemented by combining the NSF (Network Specific Facilities) function with the network routing feature. The same ISDN—PRI trunk group is assigned to different routing preferences. The service to be used is specified by the NSF assigned to each preference. When a call routes through a specific preference, the NSF for that

* NSF Value available but feature not supported on the System 85, R2V4 switch. These feature were supported by the 4E11 but not by the 4E13.

preference is used to generate the NSF IE with is then sent to the network service node to identify the network service to be used for that call. In this way the same ISDN—PRI trunk group is used for a variety of different network services rather than having to dedicate one or more trunk groups to specific services.

Non-Facility Associated Signaling (NFAS)

Non-Facility Associated Signaling is an enhancement to the PRI trunking arrangement. With NFAS, one D-channel is used to provide call control and signaling for multiple PRIs (DS1 spans). In effect, non-facility associated signaling uses the excess signaling capacity usually available on a D channel to provide signaling for additional B channels.

D-Channel Groups

To allow identification of NFAS B-channels and the associated D-channel(s), the PRI facilities that use NFAS are assigned to D-channel groups. D-Channel Groups are setup in administration using Procedure 116, Word 1 and Procedure 262, Word 2. If any part of a PRI facility (circuit) is assigned to a D-channel group, all elements (channels) of that facility must be assigned to a D-channel group, however, not necessarily the same D-channel group. On any given PRI facility, some of the channels can be assigned to one D-channel group while others are assigned to a different D-channel group.

Channel 24 as a B-Channel

The signaling D-channel is channel 24 of the PRI facility (ANN35 circuit pack in traditional modules and TN767 and TN555 circuit packs in universal modules). With non-facility associated signaling, the 24th channel is not needed for signaling purposes on supported PRI spans (PRI spans that do not provide their own signaling). This allows the 24th channel on NFAS supported PRI spans to be used as B-channels, thus increasing the communications carrying capacity of some of the PRI facilities in each D-channel group. In DEFINITY Generic 2, Procedure 260 allows a specific PRI facility to be configured as either 23 B + D or as 24 B. PRI facilities that contain either a primary or backup D-channel are configured as 23 B + D while the other PRI facilities in a D-channel groups) are set up as 24 B.

Theoretically, one D-channel could provide call control and signaling support for as many as 20 PRI spans. This would provide as many as 479 B-channels in a single NFAS D-channel group or 478 B-channels (20 PRIs X 24 channels -2 D-channels = 478 B-channels) if you use D-channel backup). This provides a potential increase in traffic support capacity of about 4%. Depending on the level of messaging activity, the practical limit may be somewhat less than 20 PRI spans to an NFAS D-channel group. An additional cost avoidance benefit is achieved as each D-channel used carries its own separate charges.

D-Channel Backup

D-channel backup is another enhancement to PRI trunking arrangements. D-channel backup is used with non-facility associated signaling. With D-channel backup, a primary D-channel (D1) provides signaling for an NFAS D-channel group (two or more PRIs facilities). A second (redundant) D-channel (D2), located on a separate PRI facility of the same NFAS D-channel group is designated as backup for D1. The failure of the primary

D-channel (D1) causes an automatic transfer of call-control signaling to the backup D-channel (D2). When this happens, the backup becomes the active D-channel, and when the primary is returned to service it returns as the stand-by D-channel.

A PRI facility, as used here is the circuitry that supports an ISDN—PRI span. For a conventional module this is an ANN35 while for a universal module it is a TN767 and TN555.

Reliability

D-channel backup is used to increase reliability for the non-facility associated signaling function. As both D-channels must be channel 24 of their respective PRI facilities it is clear that they must be on separate PRI circuits. For further reliability, the separate circuitry should also be located in separate modules.

Configuration

At any time, only one of the D-channels in a D-channel backup arrangement is used to send level 3 ISDN signaling packets. The other (backup) D-channel is in a stand-by role and is active at level 2 only (see the ISO Model in Appendix G, Figure G-2).

It should be noted that NFAS is not dependent on D-channel backup. That is, NFAS can be used by itself without a D-channel backup. However, D-channel backup is dependent on NFAS. To use NFAS with D-channel backup in a trunking arrangement with less than three ISDN—PRI spans would not provide any significant savings. D-channel backup in an NFAS group of only two spans might make sense from a reliability standpoint but would not increase the number of available B-channels.

Applications for Non-Facility Associated Signaling and D-Channel Backup

Non-facility associated signaling and D-channel backup can be used in any switch-to-switch (DEFINITY Generic 2) arrangement where two or more ISDN—PRI spans are used. The increase in B-channel communications links will generally prove cost effective.

Separate Applications for ISDN—PRI

The following is a brief description of how other features use the PRI.

DMI (Digits/ Multiplexed Interface) MOS (Message-Oriented Signaling)

The DMI feature in the MOS version is a host computer application of the ISDN—PRI. With DMI—MOS, a host computer can be given its own direct ISDN interface. A more detailed discussion is provided under the DMI feature.

Look-Ahead Interflow

The Look-Ahead Interflow feature uses the messaging capability provided by ISDN—PRI spans to query the receiving switch before interflowing calls. The ISDN—PRI feature is required for the Look-Ahead Interflow feature.

User Operations

The use of ISDN—PRI facilities is a software driven function. For example, placing an outgoing call is handled by the AAR, ARS, or WCR feature. The selection of ISDN—PRI facilities is determined by the caller's class of service, FRL, and the network routing patterns invoked. Generally, there are no specific user operations that distinguish user access to ISDN—PRI from the use of the AAR, ARS, or WCR features. That is, a caller cannot select or influence the use of ISDN—PRI calling when placing an outgoing call, except when the DMI (Digital Multiplexed Interface) feature is involved.

Considerations

Compatibility

While enhancements introduced with DEFINITY Generic 2 are not being retrofitted to System 85, Release 2, Version 4 switches, the two versions are compatible over the same network. That is, a System 85, Release 2, Version 4 PRI switch can communicate over ISDN spans with a DEFINITY Generic 2 switch. In this arrangement, the least capable interface (the System 85, Release 2, Version 4) governs the ISDN capabilities and services available. Some specific new capabilities (NFAS and D-channel backup) cannot be used on spans between System 85, Release 2, Version 4 and DEFINITY Generic 2 switches while others (codeset conversion) must be used on these spans.

Legal Considerations

Local laws vary considerably and are constantly subject to change. Legislation concerning the non-voluntary disclosure of information has been proposed in some areas. It is possible that passing, recording, or displaying some information provided by ISDN messages may be, or may become contrary to local privacy regulations. It is the responsibility of local management (switch administrator, communications director, etc.) to ensure that available features and services are not used in such a way as to violate local laws and regulations.

Constraint of NFAS with D-channel Backup

- An NFAS D-channel group is limited to a maximum of 20 ISDN—PRI spans.
- A single ISDN—PRI span can be split between multiple NFAS D-channel groups.
- There can be up to 255 NFAS D-channel groups on a single switch.

- There can be up to 510 designated NFAS D-channels (primary and secondary D-channels in combination) on a single switch.
- The physical cofiguration of both ends of an ISDN—PRI span in an NFAS arrangement must be identical. That is, there must be the same number of physical PRI interfaces, and counterpart interfaces must have the same interface identifiers (Procedure 262, Word 1, Field 13).
- Due to real-time constraints, the number of calls (circuit and non-circuit) on a D-channel at any time must not exceed 500.

Maintenance and Testing

The ISDN—PRI can be used with either the traditional or the universal module type. For maintenance and testing operations, either a SN261C ADFTC (Analog / Digital Facilities Test Circuit) or a TN771B MTCP (Maintenance Test Circuit Pack) is required. The SN261C ADFTC is used with a traditional module. The TN771B MTCP is used with a universal module and is first available with the DEFINITY Generic 2.1 Issue 2.0 switch.

The TN771B MTCP can be retrofitted to earlier models of the DEFINITY Generic 2 switch, provided the universal bus interface is also upgraded to the UN154B.

Operator/Attendant Assisted Calls

Calls placed with operator (attendant) assistance can use ISDN—PRI facilities if the operator uses one of the networking features (AAR, ARS, or WCR) to place the call. If the attendant places the call using a direct trunk group selection method (such as trunk group dial access code or the Attendant Direct Trunk Group Selection feature), ISDN—PRI trunks cannot be used.

On DEFINITY Generic 2.2 switches, if the operator assistance DAC is part of the dialed number, this must be removed (through digit modification) before the call is sent out over an ISDN—PRI trunk. Attendant assistance information is part of a separate ISDN IE and should not be included in the ISDN Called Number IE.

Party Test

Party test is not allowed on ISDN—PRI trunks.

Interactions With Other Features

ACCUNET Service Interface

The ACCUNET Service Interface feature works with the ISDN—PRI feature through the ***interworking*** function. This interaction would occur when an ISDN—PRI call is tandemed through a switch to an ACCUNET Service Interface trunk. ISDN—PRI supports both 56 Kbps service and 64 Kbps Restricted service. However, terminals and data modules currently available on the DEFINITY Generic 2 are somewhat limited in their ability to utilize 64 Kbps restricted service (PC Interface feature). Calls using the ACCUNET Service Interface feature are limited to a data rate of 56 Kbps. ISDN calls that run at data rates as high as 64 Kbps may be unrestricted. For ISDN calls using data rates of 56 Kbps or less, **interworking** passes calls to the ACCUNET Service Interface with no difficulty. However, ISDN calls that use unrestricted data rates of more that 56 Kbps cannot be passed to ACCUNET Service Interface facilities and are blocked by the DEFINITY Generic 2 switch.

AAR (Automatic Alternate Routing)

Private network calls placed over ISDN facilities use the AAR feature for ISDN access on System 85, R2 V4, and DEFINITY Generic 2.1 By routing calls via AAR, the placing of an ISDN call is transparent to the user. Factors that affect AAR preferences selection over ISDN—PRI facilities include calling party COS (Class of Service), BCCOS (Bearer) Capability Class of Service) and FRL (Facilities Restriction Level), and trunk group and preference bearer capability.

The bearer capability factor works a little differently between System 85, Release 2, Version 4 and DEFINITY Generic 2. In System 85, R2 V4, there are five bearer capability classes with fixed predefined characteristics. In DEFINITY Generic 2, 256 BCCOSs (Bearer Capability Classes of Service) are possible, of which nine are predefined. All BCCOSs (including those that are predefine) are customer administrable. For detailed information, see the Bearer Capability feature.

ACD (Automatic Call Distribution)

The ISDN—PRI feature is compatible with the ACD feature. Calls to an ACD split can use ISDN—PRI facilities when available.

ARS (Automatic Route Selection)

On System 85, R2 V4, and DEFINITY Generic 2.1 switches, public network calls placed over ISDN facilities use the ARS feature for ISDN access. Using ARS, the placing of an ISDN call is transparent to the user. Factors involved in ARS preference selection include calling party FRL (Facilities Restriction Level), BCCOS (Bearer Capability Class of Service) and trunk group and preference bearer capability.

The Bearer Capability feature and BCCOS work with ARS in the same way as the do with AAR. See the AAR feature interaction and the Bearer Capability feature interaction for more detailed information.

Attendant Control of Trunk Group Access

Generally, the Attendant Control of Trunk Group Access feature will work normally for ISDN trunk groups. However, to place an ISDN call, the **AAR, ARS, or WCR feature must be used.** Therefore, attendants cannot use the Attendant Direct Trunk Group Selection feature to complete an ISDN call.

Attendant Direct Trunk Group Selection

For proper routing and to generate messages needed for ISDN calling, all ISDN calls must be placed using the AAR, ARS, or WCR feature. The Attendant Direct Trunk Group Selection feature invokes a trunk group access code and not a network routing pattern. Therefore, the Attendant Direct Trunk Group Selection feature cannot be used to extending an ISDN call.

Bearer Capability

The Bearer Capability feature was introduced with Generic 2. This feature provides 256 BCCOS (Bearer Capability Classes of Service) to specifically define the routing needs of a specific call and the call support capability of specific trunk groups. A BCCOS is assigned to each ISDN—PRI trunk group and is used by the network routing features to select trunk groups for call routing purposes.

CDR (Call Detail Recording)

The variable format call detail record can, as an administrable option, specify ISDN specific data items. The following are some of the possible ISDN data items that can be recorded:

- ISDN NS (Network Service) used for ISDN Call-By-Call Service Selection
- ISDN User-to-user information.
- ISDN cause value.

CAS (Centralized Attendant Service)

An ISDN—PRI trunk group can be used to connect a CAS main to a CAS branch location. However, RLT (Release Link Trunk) trunk types (57, 66) **cannot** be assigned to an ISDN—PRI trunk group. This is because RLT trunks require the use of in-band signaling while ISDN—PRI trunks use out-of-band (common channel) signaling.

Data Call Setup

The Data Call Setup feature is fully compatible with the ISDN—PRI feature. PRI facilities provide a full range of support to data communications. This includes data communications that involve ISDN facilities and, through the interworking function, data communications that involve non-ISDN facilities.

DMI (Digital Multiplexed Interface)

Beginning with System 85, Release 2, Version 4, the DMI feature is available in two forms: BOS (Bit-Oriented Signaling), and MOS (Message-Oriented Signaling). Both forms are compatible with the ISDN—PRI feature. However, there are differences in the way they function with ISDN—PRI and the features and services available.

DMI BOS

The BOS (Bit-Oriented Signaling) form of DMI relies on the ***interworking function*** of the DEFINITY Generic 2 switch when accessing (or being accessed by) ISDN—PRI trunks. A

DMI BOS call is processed on the switch side by conventional call-processing software, while the ISDN side of the call is handled by the ISDN call processing software. These calls are **not** ISDN calls end-to-end (from local host computer to remote computer). Figure 66–2 shows this type of arrangement.

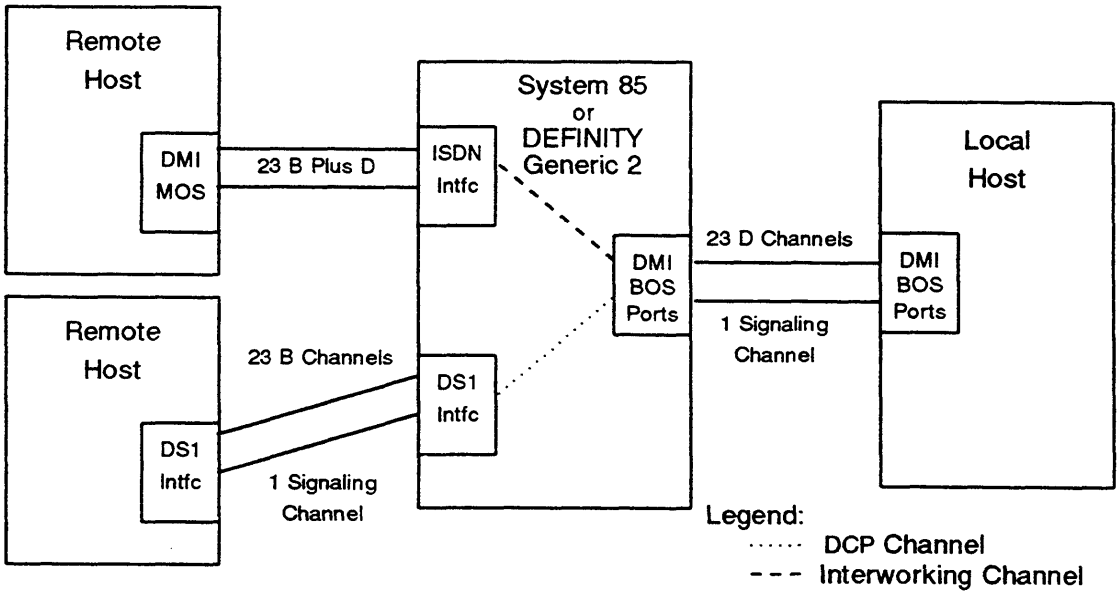

Figure 66-2. ISDN—DMI BOS Service Arrangement

DMI MOS

The MOS (Message-Oriented Signaling) form of the DMI feature is actually another form of ISDN—PRI With this interface, the DEFINITY Generic 2 switch performs a tandem switching function within the ISDN. The connection between the local and remote host is ISDN end-to-end (assuming that the remote computer also has a DMI MOS or equivalent interface). Interworking is not required unless the remote host is not supported by ISDN (or DMI MOS) interfaces. The full set of ISDN capabilities and services is available between the local and remote hosts. These include capabilities such as Call-by-Call Service Selection (driven by the host computer rather than the switch), Channel Negotiation (at the host computer level), and Tandemed User-to-User Information. Figure 66–3 shows this type or arrangement.

The two forms of the DMI feature are not mutually exclusive. That is, the same local host can connect to the same System 85, Release 2, Version 4 or DEFINITY Generic 2, switch with both a DMI MOS and a DMI BOS interface at the same time. The merits of such an arrangement must be determined locally on a case-by-case basis.

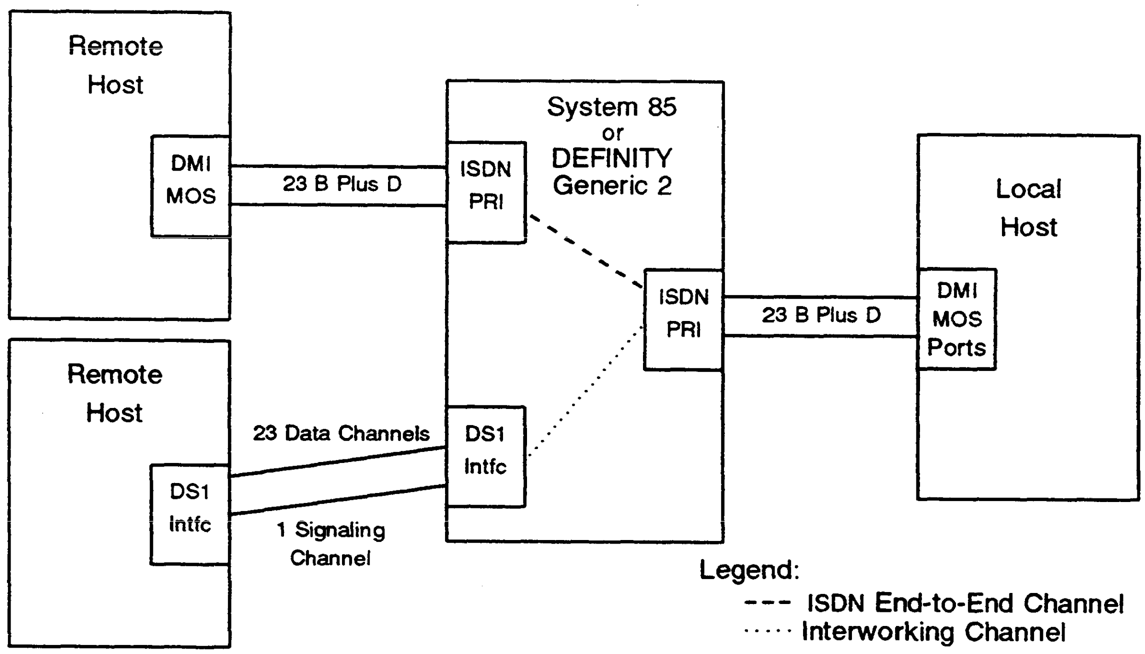

Figure 66-3. ISDN—DMI MOS (Message-Oriented Signaling) Service Arrangement

DS1 Interface

The ISDN—PRI is similar in many respects to the DS1 Interface and uses an AVD (Alternate Voice/Data) type carrier circuit like that used with the DS1 Interface feature. The signaling used with these two features is, however, different. The DS1 Interface feature uses a bit-oriented signaling protocol, while the ISDN—PRI feature uses message-oriented signaling. These two features are separate but compatible through the interworking function. That is, traffic that originated on one type of facility can be passed to the other type at a tandeming point. However, ISDN end-to-end connectivity and ISDN features and messaging is lost when this is done.

Display — Voice Terminal

The Display —Voice Terminal feature works with the ISDN—PRI feature through the interworking function. Message type information provided by ISDN is passed to the Display Voice Terminal feature software and made available to DEFINITY Generic 2 terminals with the display capability. Information displayed is not ISDN direct, but this fact is not readily apparent to the user.

DCS (Distributed Communications System)

A DCS arrangement can be set up using either ETN (Electronic Tandem Network) or Main/Satellite, trunking arrangements. The ISDN—PRI feature is compatible with ETN trunking arrangements but not with the Main/Satellite configuration.

For ISDN—PRI trunk groups that also serve as DCS trunk groups, Field 8 of Procedure 100, Word 3 should be set to "0." Otherwise, an FRL (Facilities Restriction Level) TCM (Traveling Class Mark) is not sent with each DCS call. Since a DCS trunk group is a private-network trunk group, setting the Optional IE Inhibited field to "0" does not result in additional tariff charges for the trunk group.

Extension Number Steering

This ***does not mean*** that a DCS with a Main / Satellite configuration cannot be connect to an ISDN. It does mean the ISDN facilities cannot be assigned as Main / Satellite trunks (Trunk Types 70 to 78) between the Main and Satellite switches. The Main / Satellite trunk types cannot use the ISDN signaling type 20. ***Extension Number Steering*** cannot handle ISDN message processing and the required facilities-oriented pattern searches. The network routing features (AAR, ARS, or WCR) with ***generalized route selection*** are required for the route and pattern discrimination needed by ISDN call processing.

ETN (Electronic Tandem Network) Trunking

In a DCS, ISDN—PRI trunking can be used between nodes that are arranged as an ETN. Conventional (non-ISDN) trunking must be used for any Main / Satellite connections. ISDN service can be provided to the Main (assuming the Main is a Release 2, Version 4 switch) but not between the Main and a Satellite switch (unless ETN is also provided between these switches using a parallel trunk group). Interworking at the Main will take care of call processing services for ISDN calls originating (or terminating) on a Satellite station. Figure 66-4 illustrates the various DCS / ISDN—PRI connectivity arrangements.

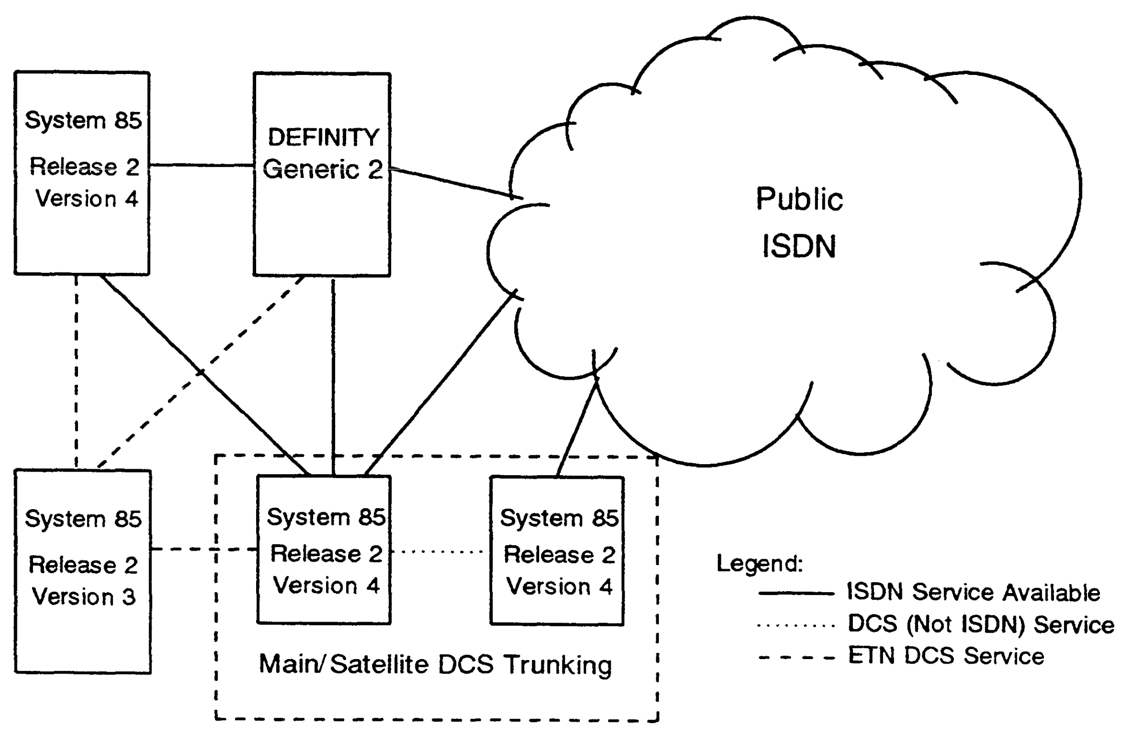

Figure 66-4. DCS / ISDN Network Connectivity

Optional ISDN IE Translation

For ISDN trunks, used in a DCS ETN arrangement, the "Optional ISDN Info Inhibited" bit should be set to "0" in Procedure 100, Word 3, Field 8. This translation allows the FRL (or TCM) of the call to be transmitted in the form of an ISDN message. Without this translation, the call is limited to the default FRL of the incoming trunk. Because these are private network connections, this translation does not increase the cost of these trunks.

ETN (Electronic Tandem Networking)

The ETN feature is essential to, and a required feature for, the use of the ISDN—PRI feature.

FX (Foreign Exchange) Access

The FX feature is compatible with the ISDN Interface feature. However, FX service is not currently being offered on a Call-By-Call Service Selection basis. This means that an ISDN T-1 carrier can be used to provide FX service on an all or nothing basis only. This is a constraint of the public network rules and not of the DEFINITY Generic 2 switch or ISDN. System 85 will support FX Service on a Call-By-Call Service Selection basis when it is available from the network.

Host Computer Access

The Host Computer Access feature can be used with the ISDN—PRI feature. The ***interworking*** function supports incoming calls through the ISDN—PRI to Host Computer Access ports. Host Computer Access ports, however, are DCP interfaced. Calls to Host Computer Access ports are (or can be) digital end-to-end, but they are not ***ISDN End-to-End*** connections. For ***ISDN End-to-End*** connectivity, the DMI MOS feature must be used.

ISN (Information Systems Network) Interface

The ISN Interface feature and the ISDN—PRI feature are compatible. ISN stations can access ISDN—PRI facilities (and ISDN—PRI calls can access ISN stations) through the interworking function.

ISDN—BRI (Basic Rate Interface)

The ISDN—BRI feature is a line side feature. For off-net ISDN services, BRI terminals use ISDN—PRI in the same way as DCP terminals, via interworking and the AAR, ARS, or WCR features.

Intercept Treatment

The Intercept Treatment feature works a little differently for ISDN calls than it does for other types of network calls. On other networks, Intercept Treatment is returned by the remote or intercepting switch with the trunk connections remaining in place until the originating party disconnects (goes on-hook).

On ISDN calls, intercept treatment is returned by the closest switch to the point of call origination that is capable of properly returning the type of intercept treatment being given, and the call is broken down to that point before intercept treatment is returned. Generally, Intercept treatment — Attendant and Intercept Treatment — Recorded Announcement are still returned by the remote switch. Intercept Treatment — Tone is returned by the local (point of origin) switch.

IXC (Interexchange Carrier) Access

The IXC Access feature works normally with the ISDN—PRI feature. Assuming the selected carrier provides ISDN service, ISDN calls will route normally. However, if the selected carrier does not provide ISDN service and the COS of the call indicates that ISDN facilities are ***required,*** the call cannot be completed.

Leave Word Calling

The Leave Word Calling feature is a DEFINITY Generic 2 switch feature and not an ISDN feature. Leave Word Calling will work within a DCS with ISDN connections (through the interworking function) but not for ISDN calls to or from the public network or a non-DCS private network.

Look-Ahead Interflow

For Look-Ahead Interflow calls (routed through ***either*** the private or public network) to succeed, ISDN—PRI connectivity is required from the sending switch to (and including) the receiving switch. When Look-Ahead Interflow is assigned to the Feature Group Class of Service (Procedure 276) at the sending (or tandeming) switch, interflow is always performed on a look-ahead basis when these ISDN facilities are available.

To increase the likelihood of "end-to-end" ISDN connectivity, Look-Ahead interflow calls always route as "ISDN Preferred."

Main/Satellite/Tributary

The Main / Satellite / Tributary networking feature is not compatible with the ISDN—PRI feature. That is, ISDN trunking arrangements cannot be used within a Main / Satellite or Tributary network. The Main (assuming it is a System 85, Release 2, Version 4 or DEFINITY Generic 2 switch) can use the ISDN—PRI feature to terminate trunks from outside the network (private or public network trunks), but ISDN service cannot be used within the Main Satellite or Main Tributary network. See also the interaction discussion for the DCS features.

Malicious Call Trace

The Malicious Call Trace feature uses the **calling number** IE (Information Element) from the ISDN message set, when available. This number is displayed at the controlling attendant console when the Malicious Call Trace feature is activated.

Modem Pooling

The Modem Pooling feature works the same on ISDN calls as it does for other types of network data calls. The switch at each end of the call must determine the need for a modem pooling conversion resource based on the administered characteristics of the station end point. The following ISDNPRI calling scenarios should receive Modem Pooling support:

- Incoming call over ISDN—PRI trunk group directed to a local analog data end point.
- Outgoing call over ISDN—PRI trunk group originated at a local analog data end point.
- Internal call from an analog line to a digital port administered as a trunk or from a digital line to an analog port administered as a trunk.

In DEFINITY Generic 2, when a data call that will require Modem Pooling support is originated from a DCP voice terminal using ***Voice Terminal Data Call Setup,*** and ***One Button Transfer,*** data preindication is required. This is not true for calls originated from ISDN—BRI voice terminals.

In System 85, Release 2, Version 4, a problem may occur if an analog interfaced data station places an ISDN call to a digital interface data end point on a remote switch. Because the call is coming in on an AVD type DS1 trunk facility, the receiving switch may not be able to recognize that a modem pooling conversion resource is needed.

Another problem could occur if data calls use the "**ISDN Facilities Preferred bearer code.**" If such calls start out on ISDN facilities and then are transferred to analog facilities at a tandem node, it is too late to insert a modem pooling conversion resource at the Originating switch.

Queuing

The Queuing feature works a little differently for certain ISDN calls. Ordinarily, a call will queue (if necessary) on the first-choice trunk group in the appropriate network muting pattern. For ISDN calls, if the COS (Class of Service) indicates ISDN facilities only, the call will queue on the first **ISDN trunk group** with the needed bearer capability in the pattern (see discussion under AAR, ARS and WCR interaction). This may not be in the first-choice trunk group.

Remote Access

The Remote Access feature should work normally with ISDN through the ***interworking*** function. The accessibility of ISDN facilities from a remote access trunk will depend on the permissions and characteristics assigned to the line appearance of that trunk.

ISDN—PRI trunks that are assigned trunk type 120 (Dynamic Trunk Type) cannot be used for the Remote Access feature.

Route Advance

The Route Advance feature must not be used for ISDN—PRI trunks. ISDN—PRI trunks can be assigned to Route Advance patterns; however, ISDN call routing must be accomplished using one of the network routing features. When either the AAR, ARS, or WCR feature is used to access the first trunk group in a Route Advance pattern, the Route Advance feature is suppressed.

Trunk Verification — Attendant

The trunk verification features work for ISDN trunks; however, special administration and user operations are required. Also, for the ISDN Dynamic trunk types, these trunks reflect a **default** trunk type based on the far end of the trunk connection.

Trunk Verification — Voice Terminal

The trunk verification features work for ISDN trunks; however, special administration and user operations are required. Also, for the ISDN Dynamic trunk types, these trunks reflect a **default** trunk type based on the far end of the trunk connection.

Touch-Tone Calling Senderized Operation

There is no direct interaction between Touch-Tone Calling Senderized Operation and ISDN—PRI. These features are compatible through the interworking function. That is, when calls tandem between touch-tone and ISDN facilities, interworking takes care of the necessary conversion. However, Touch-Tone Calling Senderized Operation is not used on ISDN facilities.

WCR (World Class Routing)

ISDN—PRI trunk groups can be used as WCR routing preferences. When this is done, the type of address and numbering plan ID for each ISDN preference must be specified to match what is expected by the serving ISDN office. This specification is made (along with the Network Service Value) for WCR preferences in Procedure 322, Word 1.

If the operator assistance code or international calling prefix is dialed, these need to be deleted from the dialed number when a PRI trunk group is selected for routing. The codes are represented elsewhere in ISDN messaging and should not be a part of the ISDN Called Number IE.

Restrictions

Access to ISDN—PRI trunks from DEFINITY Generic 2 terminals is controlled exclusively through a network routing feature routing pattern and the COS of the terminals. As part of the access control system used by AAR, ARS, and WCR, the FRL and Authorization Code features can be used to further control access to selected ISDN—PRI trunks.

Diversion type trunk restriction features such as Attendant Control of Trunk Group Access do restrict access to ISDN—PRI trunks; however, they should not be used. The diversion features remove routing control from the network muting features, and, as ISDN calls can only be routed via these features (AAR, ARS, or WCR), remover control from the network routing features effectively takes ISDN—PRI trunks out of service.

Hardware Requirements

Special Hardware Requirements

The ISDN—PRI feature uses a hardware configuration similar to the DS1 Interface feature.

In DEFINITY Generic 2, PRI is available on either the Traditional or universal module. Specific hardware needed depends on the type of module being used.

For Traditional Modules:

- DS1 /73 Series Port Carrier

  The DS1 /73 Series Port Carrier is designed to accommodate the circuit boards used for both the DS1 Interfaces and the 73 Series lime and trunk circuit boards. Each carrier supports two circuit boards in the DS1 configuration. For ISDN—PRI applications, the two circuit packs in a port earner can be arranged as follows:

  1. Both for ISDN—PRI
  2. One for ISDN—PRI and the other for conventional DS1 applications
  3. One for ISDN—PRI and the other for 73 Series applications.

- ANN35 ISDN Primary Rate Port

  The ANN35 is a single circuit pack that supports the 23 bearer channels and the D-channel called for in the ISDN—PRI standard. For the NFAS configuration, the ANN35 can also be configured as 24 B. It also provides termination facilities for the ISDN levels 1 and 2 protocol. Note that while physically similar, this circuit pack is functionally different from the standard DS1 interface circuit packs (ANN11C, 11D, and 11E).

- SN261C ADFTC (Analog/Digital Test Circuit)

  The ADPTC supports both automatic (self tests) and time available trunk testing.

- TN380D, Module Processor

  The TN380D (Module Processor) must be used in place of the TN380B or TN380C for ISDN—PRI applications. The TN380D provides for the message signaling format and larger processor instruction strings required by the ISDN feature.

- Clock Synchronization System Options

Digital trunking requires a clocking synchronization system. While this is not optional as such, the form that it takes is optional. Two clocking synchronization systems are available:

— TN463 SCS (System Clock Synchronizer)

The SCS (System Clock Synchronizer), TN463, provides Stratum 4 (Type II) clocking and synchronization. The SCS is available to all System 85 and DEFINITY Generic 2 switches with the ISDN—PRI feature. In a single-module system, the SCS resides in the module control carrier. In a multimodule system, the SCS resides in the TMS control earner.

— External Stratum 3 Clock Option

This option is available for System 85, Release 2, Version 3 and later switches and for DEFINITY Generic 2 switches. It provides a local, external, ***Stratum 3*** clock interface. This option consists of the following:

❑ TN2131, External Clock Interface Board

Replaces the TN463 SCS (System Clock Synchronizer) Board and mounts in the switch cabinet and slot that would normally house the SCS. This vanes depending on the switch vintage and configuration.

❑ Synchronization Clock, J58909A.

This unit is available only in a stand alone, duplex (duplicated) configuration, mounted in an AUDIX small cabinet.

This configuration is available in either an AC-powered or a DC-powered version.

❑ Connecting Cables

The Synchronization Clock system requires the use of three special connecting cables. The specific cables required depend on the switch configuration.

Multi-Module TMS Control or Single Module Traditional Module Control

H-600-260

H-600-274

H-600-293.

Single Module Universal Module Control

H-600-271

H-600-274

H-600-293.

For Universal Modules:

- TN767 DS1 Interface Board

  The TN767 board contains 24 port circuits that are combined to form a single, 1.544 Mbps DS1/T1 link. When used for standard ISDN—PRI service, a TN555 board is also required. The TN767 is also used for NFAS arrangements and can be configured as 24 B. When used in the 24 B configuration, the TN555 packet adjunct board is not required.

- TN555 DS1 Packet Adjunct Board

  The TN555 DS1 packet adjunct board provides packet handling service for the DS1 interface board to support ISDN—PRI messaging requirements. The TN555 DS1 packet adjunct board, in combination with the TN767 DS1 interface board, provide the functional equivalent of the ANN35 used with the traditional module.

- TN771B MTCP (Maintenance Test Circuit Pack)

  The MTCP supports both automatic (self test) and demand testing. The MTCP is equivalent to the ADFTC on the traditional module and is first available with Generic 2.1 Issue 2.0.

- UN154B Universal Bus Interface

  If the TN771B is used, the port carrier must be equipped with the UN154B Universal Bus Interface. This is an updated version from the UN154 used with initial releases of the universal module. Earlier Generic 2 switches can use the TN771B only if they are also upgraded with the UN154B.

Feature Administration

The PRI feature is assigned on a per system basis.

In Release 2, Version 4, this feature is administered using the MAAP (Maintenance and Administration Panel), SMT (System Management Terminal), VMAAP (Visual MAAP), or CSM (Centralized System Management).

In DEFINITY Generic 2, this feature is administered using the DEFINITY Manager II or the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES INTEGRATED SERVICES DIGITAL NETWORK—PRIMARY RATE INTERFACE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	3	Assigns single terminal translations, Bearer Capability Class of Service (Generic 2 only).	N/A
000	4	Assigns single terminal translations, including a code for the NPA-NXX- designator used with ISDN calling number identification. *(See* also Procedure 354, Word 3.)	Yes
010	4	Assigns features to an extension class of service, including the ISDN routing preference. 0 = Use any trunk (first available trunk) 1 = Use only ISDN trunks (if no ISDN trunks available, call fails) 2 = Use ISDN trunks if available (selects first available ISDN trunk, otherwise selects first available trunk). In System 85, Release 2, Version 4, Word 4 is used to assign bearer capability to the class of service. In DEFINITY Generic 2, this function is performed by the BCCOS assigned to the extension number in Procedure 000, Word 3.	Yes
014	1 & 2	Assigns or changes call characteristics values for BCCOS.	N/A
100	1	Assigns trunk group translations including trunk types. Any trunk type that can be assigned Signaling Type 20 (see Appendix F, Table F-A for System 85, Table F-B for DEFINITY Generic 2 switches) can be used.	No
100	2	For System 85, R2 V4 switches: Assigns digital trunk group characteristics such as data rates. For ISDN trunks, the following characteristics are applicable: ● Data Rate = 64 Kbps ("1" in Field 2) ● Asynchronous ● Full Duplex. For Generic 2 switches: Assigns the BCCOS in Field 2 (see Procedure 014).	No

(Continued)

ADMINISTRATION PROCEDURES *(Continued)* INTEGRATED SERVICES DIGITAL NETWORK—PRIMARY RATE INTERFACE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
100	3	Assigns trunk signaling types when the default signaling type is not acceptable. For trunk types other than 120, Field 2 is set to signaling type 20 for ISDN trunks. Also controls "Optional" ISDN IE transmission characteristics. Field 8 performs a "toggle" function. Encode "0" allows optional IEs to be sent while encode "1" blocks transmission of the optional IEs. The optional IEs are: ● Connected number ● User-to-user (System 85, R2 V4 only) ● Low-layer compatibility (Generic 2 only) ● Calling Party Number ● Display ● Logical link identification ● Traveling class mark ● Link layer parameters ● Packet layer parameters ● Look-ahead interflow.	No
101	1	Assigns trunk group translations (such as, use of battery reversal, signaling, CDR, etc.) and assigns ISDN trunk groups as AVD.	No
103	1	Assigns trunk group translations for network trunks (such as, Data Protection, FRL, etc.). For ISDN—PRI trunk groups (Field 14) should be set to "1."	No
108	1	Assigns the ISDN Terminating Test Line telephone digits.	No
116	1	Assigns trunks to DS1 channels. For ISDN applications, Field 10 describes the far end of the trunks for called party IE. 0 = PBX 1 = Host Computer (DMI) 2 = Network.	No
204	1	Assigns LDNs to attendant consoles and associated partitions. For ISDN, each attendant console must be assigned an LDN for use in the outgoing call message when an ISDN call is placed from the attendant console.	No

(Continued)

ADMINISTRATION PROCEDURES *(Continued)*
INTEGRATED SERVICES DIGITAL NETWORK—PRIMARY RATE INTERFACE

PROCEDURE	WORD	PURPOSE	SMT
210	2	Assigns an LDN and NPA-NXX-X designator to an attendant console for use with the ISDN Calling Number Display function.	Yes
260	1	Assigns DS1 circuits to equipment locations and assigns signaling requirements and transmission type. For the ISDN application the following specific encodes apply: ● PRI (24B or 23B+D): Field 7 — Encode 0 = 23B+D — Encode 1 = 24B. ● Application: Field 14 = "5"	No
262	1	Administers ISDN specific options such as: ● Interface type - Note that opposite ends of the same span must be different. ● ISDN facility test type ● ISDN level 2 protocol parameters	No
262	3	Assigns Codeset Mapping and D-channel Hyperactivity Management to a specific ISDN—PRI circuit board (Generic 2 only).	N/A
275	1	Assigns system class of service features, including Tandem Tie Trunk Connections and trunk-to-trunk calling.	Yes
275	4	Assigns additional system class-of-service features including ISDN active. For ISDN, Field 14 must be set to "1."	Yes
279	1	For Generic 2 switches, defines NSF (Network-Specific Facilities) code values used with the ISDN call setup message (NSF IE). NSF values are fixed (not administrable in System 85, Release 2, Version 4.	N/A
290	1	Displays DS1 trunk interface and SCS circuit assignments ELLs (equipment locations) and identification information. To search for ISDN specific trunk assignments, encode 25 or 26 is entered in the Port Type field.	Yes

(Continued)

ADMINISTRATION PROCEDURES *(Continued)* INTEGRATED SERVICES DIGITAL NETWORK—PRIMARY RATE INTERFACE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
309	1	For System 85 and Generic 2.1, assigns trunk groups to ARS plans, patterns, and preferences. Identifies specific ARS routings for ISDN applications.	Yes
		For Generic 2.2, Procedure 309 does not exist.	
309	5	For System 85 and Generic 2.1, assigns ISDN parameters to ARS routes with ISDN applications. For System 85 R2 V4, specific parameters are: ● ISDN trunk type assignment (for Dynamic Trunk Type 120) ● NSF (Network-Specific Facilities) ● Bearer capability code: Code 0 Voice or voice grade data Code 1 Mode "1" data (56 Kbps) Code 2 Mode "2" data (64 Kbps) Code 3 Mode "3" data Code 4 Mode "0" data.	Yes*
		For Generic 2.1 the ***bearer capability parameters*** are replaced by the BCCOS in Field 6.	
		For Generic 2.2, this procedure does not exist.	
321	1	For System 85, R2 V4 and Generic 2.1, assigns trunk groups to AAR plans, patterns, and preferences. Identifies specific AAR routings for ISDN applications.	Yes
		For Generic 2.2, this procedure defines parameters for digit sending and formating requirements for non-ISDN calls.	
* Display only procedure for SMT.			

(Continued)

ADMINISTRATION PROCEDURES *(Continued)* INTEGRATED SERVICES DIGITAL NETWORK—PRIMARY RATE INTERFACE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
321	5	For System 85 and Generic 2.1, assigns ISDN parameters to AAR routings with ISDN applications. For System 85, R 2, V 4 specific parameters are: ● ISDN trunk type assignment (for Dynamic Trunk Type 120) ● Network specific facilities ● Bearer capacities: Code 0 Voice or voice grade data Code 1 Mode "1" data (56 Kbps) Code 2 Mode "2" data (64 Kbps) Code 3 Mode "3" data Code 4 Mode "0" data.	Yes*
		For Generic 2.1 ***bearer capability parameters*** are replaced by the BCCOS in Field 5.	
		For Generic 2.2, there is no Word 5.	
322	1	For Generic 2.2 only, defines ISDN Sending Index characteristics used by the WCR feature when an ISDN—PRI trunk is selected for call routing.	N/A
354	3	Assigns the prefix digits (NPA-NXX-) to the extension translation codes assigned in Procedure 000, Words 3 and 4. These are used to provide ISDN Calling Number Display information (for digits before the local extension number).	No
* Display only procedure for SMT.			

Special Administration For Generic 2

The following administration may be required on a DEFINITY Generic 2 switch in selected cases. This administration is never applicable to System 85, Release 2, Version 4. The requirement in DEFINITY Generic 2 depends on the needs of the specific ISDN—PRI span, and each span must be separately evaluated to determine its needs.

Codeset Conversion

This administration is required only on the DEFINITY Generic 2 end of a span. It is required on spans connecting the DEFINITY Generic 2 switch with any of the following:

Switch Type	**Nature of Need**
System 85, R2 V4	Required
4 ESS, 4E11	Required
Other Manufacturers' PRI Switches	May be required

ADMINISTRATION PROCEDURES
ISDN—PRI CODESET CONVERSION

PROCEDURE	WORD	PURPOSE
262	3	Assigns codeset mapping number to an ISDN—PRI circuit equipment location.
280	1	Used to setup codeset conversion map tables*.

* Enmasse conversions can be mapped using a dash (—) for the codepoint identifier. When this is done, the codepoints (IE numbers) do not change. For the codeset conversion between R2 V4 and Generic 2, some of the codepoint numbers must change, therefore, mass conversion cannot be used.

Non-Facility Associated Signaling and D-Channel Backup

Non-facility associated signaling and D-channel backup are optional capabilities of the DEFINITY Generic 2 switch. These options can be used on any (two or more) ISDN—PRI spans between two DEFINITY Generic 2 switches or between a DEFINITY Generic 2 switch and a 4 ESS(E13) or System 75, Release 1, Version 4 switch.

ADMINISTRATION PROCEDURES ISDN—PRI
NON-FACILITY ASSOCIATED SIGNALING AND D-CHANNEL BACKUP

PROCEDURE	WORD	PURPOSE
116	1	Assigns B-channels to D-channel groups.
260	1	Configures PRI circuits (ANN35 or TN767 and TN555) as either 23B + D, or 24B. This allows the 24th channel (when not needed as a D-channel) to be used as a B-channel.
262	1	Assigns interface identifier number to an ISDN—PRI circuit. These must match on both ends of span.
262	2	Assigns primary and backup D-channels to D-channel groups.
262	3	Assigns Flow Control (Hyperactivity Management) to D-channels and Mapping for Codeset Conversion.
627	1	Used to manually switch call control and signaling to the backup D-channel and to busy out the failed D-channel.

Intercept Treatment

Description

The Intercept Treatment feature provides consistent treatment for calls that cannot complete in the normal manner. The forms of treatment provided are:

Intercept Treatment—Tone

Provides intercept tone for internal or private-network calls where calling parties can be expected to understand the meaning of intercept tone. The tone alternates rapidly between a high and low pitch.

Provides reorder tone (fast busy signal) for calls originated from the public network.

Intercept Treatment—Tone is the default form of Intercept Treatment when other options are not administered.

Intercept Treatment—Attendant

Allows the attendants to provide information and assistance to caller. When an intercepted call is directed to an attendant, the Attendant Display identifies the call as intercepted (INTC). Using a dial access code, an attendant can divert attendant-seeking calls, including Attendant Intercept calls, to a recorded announcement. This ability can reduce the attendant workload during busy periods, and provdide alternate treatment when an attendant is not on duty.

Intercept Treatment—Recorded Announcement

Provides a set of recorded announcements for internal, DID, and private network calls that cannot complete as dialed. The switch administrator selects and records the messages.

A second recorded announcement can be provided for internal, DID, and incoming private network calls to disconnected extensions. The switch administrator selects the recorded message to be provided.

Intercept Treatment—Recorded Announcement With Time-Out to Attendant

Provides a set of recorded announcements for internal, DID, or private access calls that cannot be completed as dialed. Calls that do not disconnect after the announcement route to the attendant. The switch administrator selects the messages.

Flexibility is available to allow different types of Intercept Treatment to be returned to callers for different types of denied calls. The customer can select the desired type of Intercept Treatment for different kinds of calls except that Intercept Tone should be prevented for DID calls from the public network. Also, a variety of recorded announcements is available to allow the announcement's message to suit the denied call.

Feature History and Development

A basic form of the Intercept Treatment feature was first available for System 85 with Release 1.

With the Release 2, Version 1 Soft package, the flexibility of this feature was increased. Beginning with the R2 V1 software, System 85 and Generic 2 are able to return different types of Intercept Treatment for different types of denied calls.

Causes and Sources

To provide different types of Intercept Treatment for different calls, Procedure 289 separates the conditions for intercept treatment into a 3 x 3 array. The horizontal dimension specifies three ***sources*** for intercepted calls while the vertical dimension specifies three ***causes*** of Intercept Treatment. This array is depicted in Table 67-A.

TABLE 67-A. Recommended Intercept Treatment by Cause and Call Source

		Source of Call		
		Public Network Calls	Private Network Calls	Local Station Calls
Causes	Calls to Vacant DACs	Attendant*	Tone	Tone
	Calls to Restricted Features or Trunks	Attendant*	Recorded Announcement W/ Time Out to Attendant	Tone
	Calls to Recently Disconnected Extensions	Recorded Announcement W/ Time Out to Attendant	Recorded Announcement W/ Time Out to Attendant	Recorded Announcement

* Attendant diversion to recorded announcement is used when an attendant is not on duty.

The following list provides a more detailed expansion of conditions that will cause a call to receive intercept treatment.

- Calls to Vacant Dial Access Codes including:
  - — Time out

    Either a digit is not dialed within 10 seconds after receiving dial tone,

    or

    a pause of more than 10 seconds occurs between dialed digits.
  - — An invalid first digit is dialed.
  - — An invalid trunk-group or feature access code is dialed.

— An unassigned extension number is dialed.

— A call is attempted across partition boundaries in a partitioned switch when Interpartion Calling is not allow.

- Calls to Restricted Features or Trunks including:

  — A call is attempted that is denied either by the class of service (Procedure 010, Word 3) or by an Attendant Control of Voice Terminals restriction.

  — A user attempts to activate a feature that is not authorized to the terminal.

  — A user dials the access code for a trunk group that has dial access restriction (Procedure 100, Word 1) assigned.

  — An invalid digit string is dialed while attempting to access the WCR (World Class Routing) feature.

  — An account code is not dialed (or an invalid account code is dialed) when one is required by FEAC (Forced Entry of Account Codes).

  — The characteristics of a call (such as FRL) do not allow the call to access any preference in the selected AAR, ARS or WCR routing pattern.

- Calls to Recently Disconnected Extensions.

User Operations

The following are the user operating procedures for this feature.

To Divert Attendant Calls to a Recorded Announcement:

1. Press an idle loop button. [PA lamp goes out, and loop lamp lights.]
2. Press **[START]** . [Dial tone]
3. Dial the Activate Announcement access code. [Confirmation tone]
4. Press **[RELEASE]** . [PA lamp lights, and loop lamp goes out.]

To Deactivate the Diversion of Attendant Calls to a Recorded Announcement:

1. Press an idle loop button. [PA lamp goes out, and loop lamp lights.]
2. Press **[START]** . [Dial tone]
3. Dial the Cancel Announcement access code. [Confirmation tone]
4. Press **[RELEASE]** . [PA lamp lights, and loop lamp goes out.]

Considerations

Mutually Exclusive Options

Attendant Intercept and Recorded Announcement Intercept (both optional) cannot be used together. However, a set of recorded announcements which time-out to Attendant Intercept is available.

Announcement Set Limits

The use of multiple recorded announcements could require several announcement sets.

The 13A system provides up to eight separate messages per announcement set. The maximum duration of a recorded announcement on the 13A announcement system is 24 seconds.

The basic KS-65270, L12 system provides one message per announcement set. The maximum duration of a recorded announcement on the basic KS-65270 digital announcer is 32 seconds. Memory expansion kits can be added to increase total message duration by 128 seconds. A 3-channel adder kit is also available to expand the system to provide 4 separate messages.

The KS-65272 4-channel digital announcer provides the same capabilities as the KS-65270, L12 with 3-channel adder kit. Four messages with a duration of 16 seconds each (total of 64 seconds) are provided by the basic system. Memory expansion is also available for the KS-65272, in 64 second increments up to a total of 512 seconds (128 seconds per channel).

Intercept Queuing Delays

When an intercepted call is routed to a recorded announcement that is already playing, the new call must wait until the announcement is finished before being connected. These calls cannot be connected to a recorded announcement in progress.

Announcement Barge-In for Attendant-Seeking Calls

When an attendant diverts all attendant-seeking calls to a recorded announcement, these calls will barge into an announcement that is already playing. Since these attendant calls could be emergency calls and their corresponding announcements quite lengthy, allowing the calls to barge in is deemed preferable to placing the calls in an announcement queue.

Recorded Announcement Limit

Based on time-slot and TMS-blockage limitations, as many as 255 callers per module can listen to the same recorded announcement at the same time. In practice, the limit is considerably lower.

The Intercept Treatment software contains a 2-second task that adds an announcement to the time slot of every calling party waiting to hear the announcement.

Intercept Tone and Tandemed Incoming Calls

Normally, according to the dialed area code and office code, the public network delivers incoming calls to the correct switch within a private network. In this event, the receiving private-network switch knows which incoming calls originated in the public network and can return reorder tone as the preferred form of Intercept Treatment for these calls. However, there can be situations where the receiving private-network switch will tandem an incoming call from the public network to another switch in the private network. When this happens, the subsequent receiving switch may not be aware that the original source of the call was the public network. Then, if the final switch applies Intercept Treatment to the call, the Intercept Tone option could be used.

As an example, consider an incoming public-network call to a ported extension in a portability subnetwork without DCS. In this situation, the public network will deliver incoming calls to the extension's original node. In turn, the receiving node will tandem the incoming call (without the benefit of DCS messaging) to the extension's new node. If Intercept Treatment should apply to the call, the new node will recognize the call as a private-network call, and, when the default form of Intercept Treatment is assigned, return Intercept Tone to deny the call.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

AAR (Automatic Alternate Routing)

The Intercept Treatment Feature works with the AAR feature to provide appropriate responses when a call attempted using AAR is denied (for reasons other than blockage). The causes for intercept treatment on AAR calls include following conditions:

- The FRL of the call is not high enough to access any preference in the selected routing pattern.
- No preference that supports callers BCCOS (Bearer Capability Class of Service) is available.
- Call times out.
- No preferences are assigned to the selected routing pattern.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

ARS (Automatic Route Selection)

The Intercept Treatment Feature works with the ARS feature to provide appropriate responses when a call attempted using ARS is denied (for reasons other than blockage). The causes for intercept treatment on ARS calls include following conditions:

- The FRL of the call is not high enough to access any preference in the selected routing pattern.
- No preference that supports callers BCCOS (Bearer Capability Class of Service) is available.
- Call times out.
- No preferences are assigned to the selected routing pattern.
- All available preferences are toll routes and the caller dialed the non-toll DAC or the calling extension is toll restricted.
- An account code was not dialed and the only preferences available have FEAC in effect.

CDR (Call Detail Recording)

If an incoming trunk call receives Intercept Treatment and is routed to the attendant, the call record looks as if the attendant had been dialed directly. This interaction occurs regardless of whether recording of ineffective attempts is active or not.

Dial Access to Attendant

The Attendant Diversion to Recorded Announcement function overrides the Dial Access to Attendant feature. When an attendant activates Attendant Diversion to Recorded Announcement, a local voice terminal user cannot reach the attendant queue or a selected attendant. Instead, these calls are diverted to the recorded announcement.

DCS (Distributed Communications System)

When an incoming call from another switch in the DCS subnetwork arrives at the local System 85 or Generic 2, the Intercept Treatment feature considers the source of the call to be ***private,*** not a local terminal.

ISDN—PRI (Primary Rate Interface)

The Intercept Treatment feature works a little differently for ISDN calls than it does for other types of network calls. On other networks, Intercept Treatment is returned by the remote or intercepting switch with the trunk connections remaining in place until the originating party disconnects (goes on-hook).

On ISDN calls, intercept treatment is returned by the closest switch to the point of call origination that is capable of properly returning the type of intercept treatment being given, and the call is broken down to that point before intercept treatment is returned. Generally, Intercept Treatment — Attendant and Intercept Treatment — Recorded Announcement are still returned by the remote switch. Intercept Treatment — Tone is returned by the local (point of origin) switch.

Look-Ahead Interflow

For Look-Ahead Interflow Calls where the sending (or tandeming) switch routes the call to the receiving switch over ISDN—PRI facilities, no type of Intercept Treatment (such as, intercept tone, reorder tone, or recorded announcement) is given to the calling party. Instead of returning Intercept Treatment, the receiving switch rejects the call with a D-channel message, and the sending switch relies on its own vector to provide a suitable alternate treatment.

For Look-Ahead Interflow Calls where the sending (or tandeming) switch routes the call to the receiving switch over ***non*** -ISDN—PRI facilities, some form of Intercept Treatment may be returned to the original calling party. Without an ISDN—PRI facility, the receiving switch has no way of rejecting the call. Moreover, the final switch does not know the trunk type of the original incoming trunk.

Main/Satellite/Tributary

At a satellite location, the ETA (Extended Trunk Access) function of the Main / Satellite feature overrides one cause of Intercept Treatment. When ETA is assigned, calls to vacant dial access codes do not receive Intercept Treatment from the satellite. Instead, the satellite outpulses the collected digits to the main over the designated trunk group. For these calls, if the main determines that Intercept Treatment is required, the main will return intercept across the trunk to the calling party at the satellite. Otherwise, the main will handle the call based on its complete numbering plan.

The two other forms of Intercept Treatment (calls to recently disconnected stations and calls to restricted features or trunks) operate normally at a satellite location.

Multiple Listed Directory Numbers

The Attendant Diversion to Recorded Announcement function overrides the Multiple Listed Directory Numbers feature. When an attendant activates Attendant Diversion to Recorded Announcement, a public-network caller cannot reach the attendant queue. Instead, these calls are diverted to the recorded announcement.

Precedence Calling

Incoming Precedence Calling calls that receive intercept treatment are given a special form of Attendant Intercept that routes the call to an available attendant or, if no attendant is immediately available, places the call in the attendant priority queue. In this case, intercept is based on criteria established for the Precedence Calling feature and not the criteria set for the Intercept Treatment feature.

Tenant Services

The Intercept Treatment feature is administered on a per-system basis. If the Intercept Treatment feature ***is programmed*** in Procedure 289, Word 1, partitioning denials for internal calls always cause the switch to return the type of intercept administered for internal calls to vacant access codes (usually, intercept tone). If the Intercept Treatment feature ***is not programmed*** in Procedure 289, Word 1, the switch always returns the default types of intercept treatment intercept tone for internal calls and reorder tone for calls from the public network.

Intercept Treatment—Attendant

In a partitioned System 85 or Generic 2, this function of the Intercept Treatment feature routes intercepted calls to an associated attendant partition. Incoming trunk calls placed to ***invalid extension numbers*** are routed to an attendant partition assigned to the trunk group. (If an attendant partition has not been assigned to the trunk group in Procedure 270, Word 5, the intercepted call routes to Attendant Partition 0.) Incoming trunk calls placed to ***valid extension numbers*** that are restricted from receiving the calls (for example by DID Restriction, Inward Restriction, or Termination Restriction) are routed to an attendant partition assigned to called extension's partition.

Intercept Treatment—Recorded Announcement

The recorded announcement messages for the Intercept Treatment feature are system-wide messages. There is no capability to assign messages with different content for different partitions.

Unattended Console Service—Call Answer Any Voice Terminal

If an attendant activates Attendant Diversion to Recorded Announcement and then an attendant activates Unattended Console Service, calls placed to the attendant queue do not enter the CAAVT queue. Instead, these callers will receive the recorded announcement. If this operation is not desired, the attendant should deactivate Attendant Diversion to Recorded Announcement before activating Unattended Console Service.

Unattended Console Service—Preselected Call Routing

If an attendant activates Attendant Diversion to Recorded Announcement and then an attendant activates Unattended Console Service, calls placed to the attendant queue do not complete to the preselected voice terminal. Instead, these callers will receive the recorded announcement. If this operation is not desired, the attendant should deactivate Attendant Diversion to Recorded Announcement before activating Unattended Console Service.

WCR (World Class Routing)

The Intercept Treatment Feature works with the WCR feature to provide appropriate responses when a call attempted using WCR is denied (for reasons other than blockage). The causes for intercept treatment on WCR calls include following conditions:

- The FRL of the call is not high enough to access any preference in the selected routing pattern.
- No preference that supports callers BCCOS (Bearer Capability Class of Service) is available.
- Call times out.
- No preferences are assigned to the selected routing pattern.
- All available preferences are toll routes and the caller dialed the non-toll DAC or the calling extension is toll restricted.

- An account code was not dialed and the only preferences available have FEAC in effect.
- The number dialed after the WCR network DAC cannot be identified in the network translations.

Hardware Requirements

The Intercept Treatment feature requires the following additional or special hardware.

For Traditional Modules:

- An auxiliary trunk circuit of an SN231 circuit pack for each announcement trunk (four circuits per SN231).

For Universal Modules:

- An auxiliary trunk circuit of a TN763C circuit pack for each announcement trunk (four circuits per TN763C).

Regardless of Module Type:

To provide the recorded announcements:

- 13A announcement system(s) (eight channels per 13A). A 36A voice coupler, with a 2012D power transformer, is required for each 13A announcement trunk.

  or

- KS-65270, L12 digital announcer(s) (single-channel announcement set). One circuit of an SN228B, SN229, or TN742 (eight circuits per pack), or one circuit of a TN746 (16 circuits per pack) must be provided for each KS-65270 to support remote announcement recording.

  or

- KS-65272 4-channel digital announcer to provide recorded announcements. One circuit of an SN228B, SN229, or TN742 (eight circuits per pack), or one circuit of a TN746 (16 circuits per pack) must be provided for each KS-65272 to support remote announcement recording.
- An auxiliary trunk circuit of an SN231 or TN763C circuit pack is needed for each announcement trunk (four circuits per SN231 or TN763C).

Feature Administration

Assignment of the Intercept Treatment feature is on a per-call type and on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — INTERCEPT TREATMENT

PROCEDURE	WORD	PURPOSE	SMT
000	1	When an extension is removed from service, it goes into the "Recent Disconnect" state for the system default interval. Calls to that extension receive the intercept treatment assigned to that state. A second removal makes the extension unassigned.	Yes
003	1	Displays and changes the remaining interval for an extension in the recent disconnect state. Also can be used to take an extension out of the recent disconnect state.	Yes
100	1	Assigns the trunk type of a recorded announcement trunk group. The applicable trunk-type encode includes: 52 Recorded announcement interface.	No
101	1	Administers the characteristics of a recorded announcement trunk group.	No
150	1	Assigns a trunk for each separate recorded announcement to the recorded announcement trunk group. All announcements can be in the same trunk group.	No
204	1	Administers console messages for the three classes of intercept calls: vacant dial access code (encode 2305), restricted feature or facility (encode 2306), or recently disconnected extension (encode 2307). Also administers a message for attendant diversion to recorded announcement (encode 2308).	No
275	4	Sets the system default for the recent disconnect interval.	Yes
289	1	Assigns the type of Intercept Treatment and specifies the recorded announcement number (from 1 to 15) to each type of call requiring Intercept Treatment.	Yes
350	1	Assigns the first digit of the feature dial access codes (if required).	No
350	2	Assigns the feature dial access codes for diverting attendant calls to recorded announcement. The applicable encodes are as follows: 80 Activate 81 Cancel.	No

Intercom — Automatic

Description

The Automatic Intercom feature provides the equivalent of a dedicated talking path between two multiappearance voice terminals. One of six different ringing patterns can be used to alert the called terminal when the calling terminal goes off-hook. This choice of ringing is made on a per-system basis and is used for both Automatic and Dial Intercom.

This feature reduces dialing time. The called party also knows who is calling.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Initiate an Automatic Intercom Call:

1. Press the appropriate **[AUTO ICOM]** button. [AUTO ICOM red status lamp lights.]
2. Go off-hook. [AUTO ICOM green status lamp lights, and ringback tone is heard. At the called terminal, the AUTO ICOM green status lamp flashes, and distinctive ringing is heard. All other intercom appearances in the intercom group will have a lighted green status lamp.]

To Answer an Automatic Intercom Call:

1. Press the appropriate **[AUTO ICOM]** button.
2. Go off-hook. [Distinctive ringing is removed at the called terminal, and ringback tone is removed at the calling terminal.]

To Bridge Onto an Automatic Intercom Call:

1. Press the appropriate **[MANUAL ICOM]** button. [MAN ICOM red status lamp lights.]
2. Go off-hook.

Considerations

Feature Parameters

Automatic Intercom is used only within the local switch. As many as 300 intercom (Automatic, Manual, or combined) groups are allowed in the switch.

Each intercom group can include a maximum of 16 voice terminals. Only 2 of the 16 terminals can be Automatic Intercoms. The remaining 14, if assigned, must be assigned as Manual Intercoms.

AUTO ICOM Buttons

The multiappearance voice terminal provides AUTO ICOM buttons for selecting the Automatic Intercom appearances.

Button Restrictions

On a single multiappearance voice terminal, either an AUTO ICOM or a MANUAL ICOM button can be assigned to a single intercom group, but not both.

Restricted Features

Features such as Call Coverage, Hold, Hunting, Transfer, etc., cannot be used on Automatic Intercom lines.

Selectable Ringing Patterns

When administering Procedure 061, the six selectable ringing patterns are:

- 0 — Ring 1 (This is about 2 seconds of modulated ringing repeated every 5 seconds.)
- 1 — Ring 2 (This is one short ring, then a 2-second modulated ring. This pattern is repeated every 5 seconds.)
- 2 — Ring 3 (This is two short rings, then a 2-second modulated ring. This pattern is repeated every 5 seconds.)
- 3 — Ring 4 (This is one short modulated ring that is not repeated.)
- 4 — Buzz 1 (This is a 1-second unmodulated tone that is repeated every 5 seconds.)
- 5 — Buzz 2 (This is a short unmodulated tone that is not repeated.)

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Bridged Call

Bridging onto an existing Automatic Intercom call is allowed. A maximum of three multiappearance voice terminals may be connected to the call at any one time. Additional parties attempting to bridge onto the connection receive reorder tone.

Extension Number Portability

The equipment location of an extension number must be removed from all intercoms, if assigned, before the extension number can be ported to another node.

Manual Signaling

When Automatic Intercom ringing is in progress, it prevents Manual Signaling.

Multiappearance Preselection and Preference

At the originating voice terminal, the Multiappearance Preselection and Preference feature cannot automatically select an intercom appearance. However, at the receiving voice terminal, the Multiappearance Preselection and Preference feature can select the intercom appearance receiving the call.

Privacy—Manual Exclusion

The Manual Exclusion feature cannot be used to prevent intercom group members from bridging onto an Automatic Intercom line.

Ringing—Ringing Cutoff

Ringing Cutoff denies all intercom ringing for the given terminal.

Tenant Services

There are no tests in Procedure 056, Word 1 to ensure that the two Automatic Intercom members of a nondial intercom group belong to the same extension partition. It is the responsibility of the system manager to ensure that intercom groups do not cross partition boundaries.

Hardware Requirements

Multiappearance voice terminals are required for this feature.

Feature Administration

Assignment of the Intercom feature is on a per-terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES INTERCOM—AUTOMATIC			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
056	1	Administers intercom type and features for a voice terminal.	Yes
061	1	Specifies the six possible intercom ringing patterns.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS INTERCOM—AUTOMATIC	
PATH NAME	**PURPOSE**
terminal-change group intercom	Displays or prints an intercom report.
terminal-change system parameters (select the Signaling option)	Specifies intercom ringing type (see Procedure 061 for types).
terminal-change terminal buttons	Assigns the AUTO ICOM button (use "nondial intercom") to a voice terminal.

Intercom — Dial

Description

The Dial Intercom feature allows multiappearance voice terminal users to gain rapid access to 27 other multiappearance voice terminal users in the same intercom group. There is a choice of six different ringing patterns that the called party can receive when reached. This choice of ringing is made on a per-system basis, and is used for both Dial and Automatic Intercom. The selected pattern lets the called party know that someone within the intercom group is calling. This feature also reduces dialing time.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Initiate a Dial Intercom Call:

1. Press **[DIAL ICOM]** . [DIAL ICOM red status lamp lights.]
2. Go off-hook. [Dial tone is heard, and every terminal in the intercom group will have a lit DIAL ICOM green status lamp.]
3. Dial a 1- or 2-digit code to reach the desired party. [Dial tone is removed, and ringback tone is heard. At the called terminal, the DIAL ICOM green status lamp flashes, and ringing is heard.]

To Answer a Dial Intercom Call:

1. Press **[DIAL ICOM]** . [DIAL ICOM red status lamp lights.]
2. Go off-hook. [Ringing is removed. Calling and called terminal DIAL ICOM red status lamps remain lighted.] (If one of the two links is idle, all other dial intercom appearances in this dial intercom group will have a dark status lamp. If both links become busy, all dial intercom appearances in this intercom group will have a lighted status lamp.)

Considerations

Feature Parameters

The Dial Intercom feature can be used only within the local switch. Up to 280 Dial Intercom groups are allowed per switch. Each group can have up to 28 members.

Mixing Codes

Within an intercom group, a mixture of 1-digit and 2-digit codes is allowed. However, any number used as the first digit of a 2-digit code cannot also be used as a 1-digit code within the same intercom group. Furthermore, no more than two distinct digits can be used as the first digit of a 2-digit code. (A legal dialing arrangement would be: 0, 1, 3, 4, 6, 7, 8, 9, 20, 21, 22, 23, 24, 50, 51, 52, 53, 54, 55, and 56. This group contains 20 members using 1- and 2-digit codes.)

Dial Intercom Lamps

The multiappearance voice terminal provides DIAL ICOM buttons for the dial intercom appearances. When both Dial Intercom links are busy, the Dial Intercom status lamp is dark. When only one link is busy, only the voice terminals involved have a lit status lamp. Incoming Dial Intercom calls make the Dial Intercom status lamp flash at a fast steady rate.

Dial Intercom Links

Only two separate conversations (Dial Intercom links) may occur at the same time within a Dial Intercom group. If both Dial Intercom links are being used, a user attempting to access Dial Intercom will receive reorder tone. If a Dial Intercom voice terminal is busy, a user attempting to connect to that terminal receives a busy tone. If a user attempts to dial an unassigned or invalid Dial Intercom code, they receive intercept tone.

DROP Button and Dial Intercom Appearances

Currently, the user of a Dial Intercom appearance who is using this appearance for a 2-party call cannot press the DROP button to disconnect the call and to receive new dial tone on the same appearance.

Bridging Dial Intercom Calls

Other Dial Intercom voice terminals cannot bridge onto an existing Dial Intercom call. Features such as Call Coverage, Hold, Hunting, Transfer, etc., cannot be used on Dial Intercom lines.

Selectable Ringing Patterns

When administering Procedure 061, the six selectable ringing patterns are:

- 0 — Ring 1 (This is about 2 seconds of modulated ringing repeated every 5 seconds.)
- 1 — Ring 2 (This is one short ring, then a 2-second modulated ring. This pattern is repeated every 5 seconds.)
- 2 — Ring 3 (This is two short rings, then a 2-second modulated ring. This pattern is repeated every 5 seconds.)
- 3 — Ring 4 (This is one short modulated ring that is not repeated.)

- 4 — Buzz 1 (This is a 1-second unmodulated tone that is repeated every 5 seconds.)
- 5 — Buzz 2 (This is a short unmodulated tone that is not repeated.)

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Bridged Call

Bridging onto a Dial Intercom connection is not allowed.

Extension Number Portability

The equipment location of an extension number must be removed from all intercoms, if assigned, before the extension number can be ported to another node.

Last Extension Dialed

The Last Extension Dialed feature will not store or redial a Dial Intercom number.

Last Number Dialed

The Last Number Dialed feature will store and redial a Dial Intercom number.

Manual Signaling

When Dial Intercom ringing is in progress, it prevents Manual Signaling.

Multiappearance Preselection and Preference

At the originating voice terminal, the Multiappearance Preselection and Preference feature cannot automatically select an intercom appearance. However, at the receiving voice terminal, the Multiappearance Preselection and Preference feature can select the intercom appearance receiving the call.

Privacy—Manual Exclusion

Since bridging is denied for Dial Intercom lines, the Manual Exclusion feature does not function on Dial Intercom lines.

Ringing—Abbreviated and Delayed Ringing

This feature cannot be assigned to Dial Intercom groups.

Ringing—Ringing Cutoff

Ringing Cutoff denies all intercom ringing for the given terminal.

Tenant Services

There are no tests in Procedure 056, Word 1 to ensure that the members of a Dial Intercom group belong to the same extension partition. It is the responsibility of the system manager to ensure that intercom groups do not cross partition boundaries.

Hardware Requirements

Multiappearance voice terminals are required for this feature.

Feature Administration

Assignment of the Dial Intercom feature is on a per-terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — INTERCOM—DIAL			
PROCEDURE	WORD	PURPOSE	SMT
056	1	Administers the intercom type and features for a voice terminal.	Yes
061	1	Administers the six possible intercom alerting patterns. All the alerting patterns have visual alerting.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — INTERCOM—DIAL	
PATH NAME	PURPOSE
terminal-change group intercom	Displays or prints an intercom report.
terminal-change system parameters (select the Signaling option)	Specifies intercom alerting type (see Procedure 061 for patterns).
terminal-change terminal buttons	Assigns the DIAL ICOM button to a multiappearance voice terminal.

Intercom — Manual

Description

The Manual Intercom feature allows a multiappearance terminal user to access other multiappearance terminals assigned to the same intercom group. At any ont time, up to three users in the group can be connected to a single call.

A manual intercom signal is heard at the called party's terminal each time the caller presses the SIGNAL button. The manual intercom signal is one short beep.

This feature is useful when involved in a 2-way connection and it is desirable to have a third party, like a secretary, bridge on to the conversation.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Initiate a Manual Intercom Call:

1. Press **[MANUAL ICOM]** . [MANUAL ICOM red status lamp lights.]
2. Go off-hook. [MANUAL ICOM green status lamp lights. All other intercom buttons in the group have a lit green status lamp.]
3. Press the appropriate **[SIGNAL]** button. [SIGNAL green status lamp lights while the button is pressed. The called terminal receives one short beep each time the SIGNAL button is pressed.]

To Answer a Manual Intercom Call:

1. When the beep is heard, go off-hook.
2. Press **[MANUAL ICOM]** . [MANUAL ICOM red status lamp lights.]

To Bridge Onto a Manual Intercom Call:

1. Go off-hook.
2. Press **[MANUAL ICOM]** **.** [MANUAL ICOM red status lamp lights.]

Considerations

Feature Parameters

As many as 300 intercom (Automatic, Manual, or combined) groups are allowed on the switch. More than one intercom group may appear on a multiappearance terminal. A separate button is required for each intercom group assigned to the terminal.

Each intercom group can include a maximum of 16 terminals. A separate Manual Signaling button is required to alert each terminal.

Button Restriction

On a multiappearance voice terminal, either a MANUAL ICOM or an AUTO ICOM button can be assigned to an intercom group, but not both.

Assigning Manual Signaling

Manual Signaling should be assigned to multiappearance terminals that have Manual Intercom assigned.

Features Used With Manual Intercom

Manual Signaling and Bridged Call are the only voice terminal features that can be used with Intercom—Manual. All others (such as, Call Coverage, Hold, Hunting, etc.) cannot be used.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Bridged Call

Bridging onto an existing Manual Intercom call is permitted. A maximum of three multiappearance voice terminals may be connected to a Manual Intercom call at any one time. Additional calls trying to bridge onto the call receive reorder tone.

Extension Number Portability

The equipment location of an extension number must be removed from all intercoms, if assigned, before the extension number can be ported to another node.

Multiappearance Preselection and Preference

At the originating voice terminal, the Multiappearance Preselection and Preference feature cannot automatically select an intercom appearance. However, at the receiving voice terminal, the Multiappearance Preselection and Preference feature can select the intercom appearance receiving the call.

Privacy—Manual Exclusion

Privacy—Manual Exclusion cannot be used to prevent an intercom group member from bridging onto an intercom call.

Ringing—Ringing Cutoff

Ringing Cutoff denies Manual Intercom ringing for the given terminal.

Tenant Services

There are no tests in Procedure 056, Word 1 to ensure that the Manual Intercom members of a nondial intercom group belong to the same extension partition. It is the responsibility of the system manager to ensure that intercom groups do not cross partition boundaries.

Hardware Requirements

Multiappearance voice terminals are required for this feature.

Feature Administration

Assignment of the Manual Intercom feature is on a per-terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can fully administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — INTERCOM—MANUAL			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
053	1	Administers a signaling and a signaled terminal for Manual Signaling.	Yes
056	1	Administers the intercom type (nondial). Also, enter dashes in Field 10 and 0 (for Manual Signaling) in Field 11. This procedure also displays the number of terminals in the intercom group.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — INTERCOM—MANUAL	
PATH NAME	**PURPOSE**
terminal-change group intercom	Displays or prints an intercom report.
terminal-change terminal buttons	Assigns the MAN ICOM button (use "nondial intercom") to a voice terminal.

Interexchange Carrier Access

Description

The IXC (Interexchange Carrier) Access feature allows customers to specify the particular interexchange carrier vendor (such as AT&T, MCI, Sprint, etc.) they wish to use for calls to a given dialing destination. The IXC Access feature uses the capabilities of the network routing features (AAR [Automatic Alternate Routing], ARS [Automatic Route Selection], or WCR [World Class Routing]) to routing calls to selected long distance service venders. On System 85 and DEFINITY Generic 2.1 switches, this feature is controlled entirely through administration. On Generic 2.2 switches, IXC Access may be entirely controlled through administration, or may optionally allow caller participation.

Feature History and Development

The IXC Access feature was first available for System 85 in Release 2, Version 3.

With DEFINITY Generic 2.2, this feature is enhanced with the addition of administerable options which can allow the caller to dial an IXC Access code to influence selection of an interexchange carrier to be used for the call.

Initiating Calls

On System 85 and Generic 2.1 switches, the ARS feature routes calls over the most preferred off-network route. The caller dials only the ARS dial access code followed by the public network number. The switch chooses the outgoing trunk group and (when appropriate) uses subnetwork trunking to make any modifications to the dialed number and then sends the digits necessary to complete the call.

For DEFINITY Generic 2.2 switches, networking services are provided by the WCR feature. The WCR feature can be administered to duplicate the operations of the AAR and ARS features used with earlier switches. However, the WCR feature can also be administered to allow the caller to dial the IXC access code to select the carrier to be used. Generic 2.2 allows calls to specified carriers to be routed over private network facilities if all switches in the network are Generic 2.

Receiving Calls

If the destination switch has DID (Direct Inward Dialing) capability, the call is routed directly to the called terminal. Otherwise the public network number is the LDN (Listed Directory Number) of that switch, and an attendant at the destination switch must complete the call to the desired terminal.

For a destination switch without DID, if the destination switch has a Remote Access feature, it may be possible for overflowed network calls to terminate directly to the called terminal by using Remote Access. With proper digit manipulation, the called number can be altered to provide access to the Remote Access feature on the destination switch and from there to the desired extension.

User Operations

The following is the user operating procedure for this feature.

To Place a Call to a Given Destination Using Automatic IXC Selection:

1. Go off-hook. [Dial tone]
2. Dial the appropriate dial access code (AAR, ARS, or WCR). [Second dial tone]
3. Dial the appropriate address digits for the call you want to place. [Ringback tone]

To Place a Call to a Given Destination Using Dial Access to a Specific IXC:

1. Go off-hook. [Dial tone]
2. Dial the appropriate WCR network dial access code. [Second dial tone]
3. Dial the desired IXC Access code, followed by appropriate address digits for the call you want to place. [Ringback tone]

Considerations

Real Time Impact

From a the callers perspective, there is no time impact when IXC selection is automatic. However, increasing the number of digit sent for subnetwork trunking requires the holding of an originating Register for a longer period of time. When the caller enters an IXC Access code, dialing time is increased slightly.

Interactions With Other Features

Abbreviated Dialing

Calls placed using the Abbreviated Dialing feature are provided the same support by the IXC Access feature as calls place by dialing from the terminal.

AAR (Automatic Alternate Routing) and ARS (Alternate Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, the subnetwork trunking function of the AAR and ARS features is used to implement the IXC selection automatically. An IXC code (10XXX) cannot be dialed by a station user, and the IXC digits are not passed over the ETN network.

ISDN—PRI (Primary Rate Interface)

The IXC Access feature works normally with the ISDN—PRI feature. Assuming the selected earner provides ISDN service, ISDN calls will route normally. However, if the selected carrier does not provide ISDN service and the COS of the call indicates that ISDN facilities are required, the call does not complete.

Last Number Dialed

Redial by the LND (Last Number Dialed) feature utilizes all of the network routing digits (including IXC Access digits if required for the newly chosen preference) as though the call were manually dialed. The LND feature will also store and redial the network access code.

Look-Ahead Interflow

The Look-Ahead Interflow feature is compatible with the IXC access feature. The AAR, ARS, and WCR subnetwork trunking functions can modify the digit formats as necessary for vector-group list items so that Look-Ahead Interflow calls can route over IXC Access trunk groups.

Therefore, from a sending switch, calls can interflow over the public network using ISDN—PRI trunk groups assigned as IXC Access trunk groups. When this is done, these interflow calls will succeed if every public-network switch between the sending switch and the destination telephone number is ISDN—-PRI-capable.

WCR (World Class Routing)

On System 85, and DEFINITY Generic 2.1 switches, use of the IXC Access feature is determined automaticaly by the ARS feature and cannot be influenced directly by the caller. With WCR (on DEFINITY Generic 2.2 switches), use of the IXC Access feature can be automatic (as with the ARS feature. However, with WCR, IXC codes can be included in the network dialing plan. This allows the caller to dial an IXC Access code to select a specific carrier for a specific call. Additional options allow the administrator to decide if these calls can be routed in special ways (such as directly to the earner or via tail-end hop off) or if they should be sent to the regional telephone operating company for routing.

Restricting Feature Use

The IXC Access feature is used to provide access to toll facilities. An individual extension can be restricted from access to this feature by the Restriction features that restrict toll access. They can also be restricted through the use of the Facilities Restriction Level feature.

Hardware Requirements

None.

Feature Administration

Assignment of the IXC Access feature is on a per-switch basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the FM (Facilities Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
INTEREXCHANGE CARRIER ACCESS

PROCEDURE	WORD	PURPOSE	SMT
309	1	For System 85 and Generic 2.1 switches, administers routing pattern parameters for ARS including trunk group number, FRL, warning tone, home NPA at distant end of route, dial 1 for toll requirements, toll table index, number of beginning digits deleted, dc signal ignore, and IXC Identifier. To forward calls off-net, all three ARS plans must be defined using this procedure. For Generic 2.2 switches, this Procedure does not exist.	Yes
309	3	For System 85 and Generic 2.1 switches, defines the four digit groupings and dialing format (touch-tone or rotary). For Generic 2.2 switches, this procedure does not exist.	Yes
309	4	For System 85 and Generic 2.1 switches, defines the inserted digits a trunk group in an ARS pattern requires for subnetwork trunking. For Generic 2.2 switches, this Procedure does not exist.	Yes
314	l & 2	For Generic 2.2 switches, defines network parameters and dial plans used with the WCR feature for digit analysis and routing, including IXC codes.	N/A

(Continued)

ADMINISTRATION PROCEDURES INTEREXCHANGE CARRIER ACCESS (Continued)			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
321	1	For System 85 and Generic 2.1 switches, administers routing pattern pattern parameters for AAR including: trunk group number, FRL, warning tone, off net (to a DDD or IDDD location) number of beginning digits deleted, dc signal ignore, IXC identifier, and whether network has 0XXX extension numbers or not. For Generic 2.2 switches, defines digit sending index characteristics including DAC, IXC code, Toll Prefix, and End of Dialing character for trunk groups in WCR patterns.	Yes
321	2	For System 85 and Generic 2.1 switches, defines the digit grouping and dialing format (touch-tone or rotary) for a trunk group in an AAR pattern that uses subnetwork trunking. For Generic 2.2 switches, defines the digit sending index used to provide digit grouping instructions, including mode (touch-tone or rotary) for trunk groups in WCR patterns.	Yes
321	3	For System 85 and Generic 2.1 switches, defines the digits to insert for a trunk group in an AAR pattern that uses subnetwork trunking. For Generic 2.2 switches, there is no Word 3.	Yes

The following are the applicable FM path names used with the AP 16.

FM SCREENS — INTEREXCHANGE CARRIER ACCESS	
PATH NAME	**PURPOSE**
facilities-mgmt routing route-selection patterns (for ARS)	Displays and changes attributes associated with the trunk groups that make up an ARS pattern.
facilities-mgmt routing alternate-routing patterns (for AAR)	Displays and changes attributes associated with an AAR pattern.

Notes:

Interpartition Access

Description

The IPA (Interpartition Access) feature provides greater calling flexibility in a partitioned System 85 or DEFINITY Generic 2. Using IPA, a voice terminal user in one extension partition can call a voice terminal user in another extension partition, provided ***both*** partitions belong to the same partition group.

Attendant and Trunk Group Access

In the Tenant Services environment with IPA, several partitioning relationships remain unchanged. These relationships include:

- Extension partitions and attendant partitions
- Extension partitions and trunk groups
- Attendant partitions and trunks groups.

In this way, most of the partitioning can be preserved while extension calling across partition boundaries is allowed.

Full Interpartition Access

In some locations, the PUC (Public Utilities Commission) allows Interpartition Access for an entire partitioned switch. When this is allowed, a savings in public-network access trunks can be derived.

Configured in this way, the partitioned switch could have as many as 1000 extension partitions and only one partition group. Voice terminal users on this switch can call any other user within the switch (regardless of extension partition). Meanwhile, each extension partition would still have separate (or shared) trunk groups and attendants.

Attendant Direct (Dividing Attendant Responsibilities)

Other System 85s or DEFINITY Generic 2s can be configured to provide an Attendant Direct configuration. This configuration provides a functional division of the attendant workforce.

Perhaps a company has three major divisions (e.g., sales, service, and production) in one location, and an attendant is desired for each division. This company's voice terminals can be subdivided into three extension partitions. Each extension partition is assigned to a different attendant partition. And, each attendant partition contains a different attendant. Also, three LDNs (Listed Directory Numbers) complete calls to the three separate attendant partitions.

In effect, this company's switch has been subdivided into three functional components. However, since sales, service, and production also share interests, the three extension partitions are also allowed to freely communicate by assigning these partitions to a common partition group.

Hybrid IPA Configuration

The "hybrid" IPA configuration combines the two previous configurations. Under this scenario, a System 85 or DEFINITY Generic 2 is fully partitioned, but one (or several) tenants want to have the Attendant Direct functionality, or several tenants share a community of interest. Some tenants could have an Attendant Direct configuration while others could choose more flexible extension calling.

> **NOTE:** The IPA (Interpartition Access) feature is used in conjunction with the Tenant Services feature to allow calling between extension partitions. Refer to the Tenant Services chapter of this manual for a description of the Tenant Services feature.

Table 72-A shows the ways that extension partitions can be assigned to partition groups. Besides the voice terminal in Extension Partition 0, this configuration contains four tenants.

- The extension in Extension Partition 0 can call or be called by any extension in the switch.
- Tenant A's extensions belong to Extension Partition 1, and Tenant B's Extensions belong to Extension Partition 2. However, both extension partitions belong to Partition Group 1. Therefore, an extension in either of these extension partitions can call or be called by any other extension in the same partition group.
- Tenant C'S extensions belong to Extension Partition 3, and Extension Partition 3 is the only partition in Partition Group 2. This relationship retains strict partitioning. An extension in Extension Partition 3 can call or be called by any other extension in the same extension partition.
- The configuration for Tenant D is an example of how Attendant Direct can be administered. Tenant D's extensions belong to Extension Partitions 4 and 5. However, all of Tenant D's extensions belong to Partition Group 3. Although Extension Partitions 4 and 5 can have separate attendants, extensions in both partitions can still call each other because these extensions belong to the same partition group.

TABLE 72-A. Extension Partitions and Partition Groups

Tenant	Extension Number	Extension Partition	Partition Group
	1000	0	—
A	2001	1	1
A	2002	1	1
A	2003	1	1
B	3001	2	1
B	3002	2	1
C	4001	3	2
C	4002	3	2
D	5001	4	3
D	5002	4	3
D	5003	5	3
D	5004	5	3

Feature History and Development

This feature was first available for System 85 in Release 2, Version 4.

User Operations

The InterPartition Access feature has no special user operations. To call a voice terminal user within the partition group, the calling party dials an extension number in the usual manner.

Considerations

Switch Capacities

As many as 500 partition groups can be provided in a partitioned System 85 or DEFINITY Generic 2.

Each extension partition can only be assigned to one partition group.

As many as 1000 extension partitions can be assigned to the same partition group.

Legal Considerations

Laws governing the use of Interpartition Access vary in different locations and are subject to change. It is the responsibility of the System Manager to understand and comply with the applicable regulations.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of IPA. When IPA is assigned, these interactions take precedence over the corresponding Tenant Services interactions. However, the Tenant Services feature impacts the operation of many features that are not included in this list. For these features, the Tenant Services interactions will apply.

Abbreviated Dialing

Abbreviated Dialing list items are not checked for legality at the time they are entered. Illegal list items are accepted into the list, but calls to these illegal numbers are denied when the digits are outpulsed by the switch. When a user attempts to use Abbreviated Dialing to place a call over another partition's trunk group or to an extension in another partition group, the switch returns intercept treatment to the calling party.

AUDIX (Audio Information Exchange)

A partitioned System 85 or DEFINITY Generic 2 is not aware of AUDIX user permissions. When a voice terminal user dials the AUDIX extension number, the switch follows the usual rules for terminal-to-terminal calling to reach the AUDIX system. Therefore, the AUDIX extension number must either belong to the user's partition group or to Extension Partition 0.

After reaching the AUDIX system, messages can be left for, created by, or retrieved by any subscriber (regardless of the partition group the subscriber belongs to).

The Call Transfer To (and From) AUDIX functions are partitioned. A voice terminal user (in an extension partition other than Extension Partition 0) can transfer these calls to extension numbers in the same partition group or to extensions in Extension Partition 0. When the user tries to transfer these calls to any other partition group (using an extension number), the switch returns intercept treatment to the transferring party.

A voice terminal user in Extension Partition 0 is allowed to transfer these calls (using an extension number) to any voice terminal in the switch.

Authorization Codes

For switch security, strict partitioning is applied to Authorization Codes. Authorization Codes can only be assigned to an extension partition (not to a partition group), and these codes cannot be shared by the other extensions in a different extension partition.

ACD (Automatic Call Distribution)

To protect the privacy of ACD agents, service observing is only allowed from within the same extension partition or Extension Partition 0. An observer in a different extension partition (but the same partition group) as an ACD agent cannot observe the agent using service observing.

To protect the privacy of ACD agents, agent override is only allowed from within the same extension partition or Extension Partition 0. An observer in a different extension partition (but the same partition group) as an ACD agent cannot observe the agent using agent override.

Automatic Callback

A voice terminal user (in a partition other than Extension Partition 0) is allowed to activate Automatic Callback toward voice terminals in the same partition group or in Extension Partition 0. When the user tries to activate Automatic Callback toward a voice terminal in any other partition group, the switch returns intercept treatment.

A voice terminal user in Extension Partition 0 is allowed to activate Automatic Callback toward any voice terminal in the switch.

Call Forwarding—Busy and Don't Answer

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition group or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition group.

Call Forwarding—Don't Answer

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition group or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition group.

Call Forwarding—Follow Me

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition group or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition group.

Call Pickup

There are no tests in Procedure 000, Word 2 to ensure that a Call Pickup group is only assigned to extensions residing in the same partition group. The system manager should ensure that every member of each Call Pickup group belongs to the same partition group.

When Call Pickup groups have been assigned to overlap partition-group boundaries, the call-processing software provides partitioning for the feature. If a Call Pickup group member in one partition group tries to pickup a call to another group member residing in a different partition group, intercept treatment is returned by the switch.

Call Vectoring

A voice terminal user (in a partition other than Extension Partition 0 is allowed to call (using an extension number) a VDN assigned to an extension partition in the same partition group or assigned to Extension Partition 0. If the user tries to call a VDN assigned to another group's extension partition, the switch will return intercept treatment.

The "route to" vector command can route calls to an extension partition in the VDN's partition group or to Extension Partition 0. If a "route to" command is programmed to route calls to another extension partition, the switch will treat a final effective "route to" step as a "stop" step. Otherwise, the "route to" step is ignored, and vector processing continues with the next sequential step.

Call Waiting

A partitioned System 85 or DEFINITY Generic 2 allows Call Waiting and provides 1-burst tone for terminal-to-terminal calls inside a partition group. The switch also allows Call Waiting and provides 1-burst tone for terminal-to-terminal calls between an extension partition and Extension Partition 0.

Otherwise, terminal-to-terminal calls are not allowed to cress partition-group boundaries. In order to call a voice terminal in any other partition group, a voice terminal user must dial the appropriate 7-digit number which routes the call over a CO (Central Office) trunk. When this is done, the switch allows Call Waiting and provides 2-burst tone for the called voice terminal.

The switch allows Call Waiting for incoming calls from the public network, and provides 2-burst tone for these calls.

Conference—Three Party

A voice terminal user (in a partition other than Extension Partition 0) is allowed to establish 3-party conferences that include participants that the user is otherwise allowed to call. If the user tries to add a conferee to the conference by dialing an extension number in another partition group, the switch returns intercept treatment to the user.

A voice terminal user in Extension Partition 0 is allowed to establish 3-party conferences with any voice terminal or over any trunk in the switch.

Hot Line

Hot Line numbers are not checked for legality at the time they are entered. Illegal Hot Line numbers are denied when the digits are outpulsed by the switch. When a user attempts to place a Hot Line call over another partition's trunk group or to an extension in another partition group, the switch returns intercept treatment to the calling party.

Last Number Dialed

A voice terminal user (in an extension other than Extension Partition 0) can use the Last Number Dialed button to redial calls within the user's partition group, to Extension

Partition 0, to the associated attendant partition, to Attendant Partition 0, or over an associated trunk group. When the user tries to use this button to redial an illegal call, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can use the Last Number Dialed button to redial calls to any extension partition, to any attendant partition, or over any trunk group in the switch.

Leave Word Calling

Leave Word Calling messages can only be left for an extension in the same partition group or for an extension in Extension Partition 0. When Leave Word Calling is blocked by partitioning, the switch returns intercept treatment to the calling party.

Demand Print

The Demand Print function of the Leave Word Calling feature is also partitioned as part of the Tenant Services feature.

When a demand printout is requested from a voice terminal (in a partition other than Extension Partition 0), System 85 or DEFINITY Generic 2 software checks to determine whether the extension used to retrieve the messages and the extension for which the messages were left are in the same partition group. If not, the switch returns intercept treatment to the party requesting the demand printout.

From a voice terminal in Extension Partition 0, a demand printout can be requested for messages left to any extension in the switch.

Given the above operation, there are two approaches to implementing the Demand Print function on a partitioned switch. One approach is to provide a voice terminal and an associated printer for each partition group. The other approach is to provide a voice terminal and an associated printer to Extension Partition 0, and share this facility with all of the partition groups.

> **NOTE:** Access to a printer cannot be shared by several partition groups and denied to the other partition groups.

Override

A voice terminal user (in a partition other than Extension Partition 0) is allowed to place Overnde calls to extensions in the same partition group or in Extension Partition 0. When the user tries to place an Override call to an extension in any other partition group, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can place an Override call to any other extension in the switch.

Priority Calling

A voice terminal user (in an extension partition other than Extension Partition 0) is allowed to place priority calls within the user's partition group or to Extension Partition 0. When the user tries to activate Priority Calling toward an extension in any other partition group, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can place a priority call to any other extension in the switch.

Whenever Priority Calling is allowed, the switch provides 3-burst ringing or 3-burst waiting tone for the called voice terminal.

Transfer

A voice terminal user (in an extension partition other than Extension Partition 0) can transfer calls to extension numbers within the same partition group or to extensions in Extension Partition 0. When the user tries to transfer a call to any other partition group (using an extension number), the switch returns intercept treatment to the transferring party.

A voice terminal user in Extension Partition 0 is allowed to transfer calls (using an extension number) to any voice terminal in the switch.

Hardware Requirements

None.

Feature Administration

Assignment of the Interpartition Access feature is on a per-extension partition basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

The customer can administer this feature using the SMT (System Management Terminal).

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — INTERPARTITION ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
270	1	Assigns an extension partition to a partition group.	Yes

AT&T

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DEFINITY® Communications System Generic 2
Feature Descriptions, Vol. 2
Features L-W and Appendixes

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Operation of this equipment in a residential area is likely to cause interference, in which case the user at his/her own expense will be required to take whatever measures may be required to correct the interference.

Network Registration Number. This equipment is registered with the FCC under FCC network registration number AS593M-13283-MFE.

Answer-Supervision Signaling. Allowing this equipment to be operated in such a manner as to not provide proper answer-supervision signaling is in violation of Part 66 rules. This equipment returns answer-supervision signals to the public switched network when:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user.

This equipment returns answer-supervision on all DID calls forwarded back to the public switched telephone network. Permissible exceptions are:

- A call is unanswered
- A busy tone is received
- A reorder tone is received

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This document was prepared by the AT&T Technical Publications Department, Denver CO.

Last Extension Dialed

Description

This feature automatically redials the last 3-, 4-, or 5-digit extension number dialed from a multiappearance voice terminal. This function is useful when redialing a busy extension. Also, by freeing the user from repeatedly locating an extension number in the directory, the feature saves time and effort for the user.

Feature History and Development

This feature was first available for System 85 in Release 1. Five-digit extension numbers were made available to System 85 in Release 2, Version 1. When this was done, the Last Extension Dialed feature was made compatible with 5-digit extensions.

The Last Extension Dialed feature was replaced and enhanced by the Last Number Dialed feature in the Issue 1.2 software of Release 2, Version 3.

User Operations

The following is the user operating procedure for this feature.

To Repeat the Last Extension Dialed:

1. Go off-hook. [Dial tone]
2. Press **[LAST EXT DIALED]** . [Ringback tone]

Considerations

Access Codes

The Last Extension Dialed feature ignores feature-button presses and access codes dialed using Abbreviated Dialing buttons. Instead, the switch redials the last extension number dialed before these operations were performed.

Manually dialed access codes are not redialed. This feature redials only extension numbers.

Last Extension Dialed but No Effect

When a hard processor swap or a power failure occurs, the last extension number dialed from each voice terminal is lost from memory. Therefore, pressing the LAST EXT DIALED button after a hard swap or a tape reload (to initialize the System 85) and before dialing another extension number has no effect.

The Last Extension Dialed feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 features affect or are affected by the operation of this feature.

AUDIX (Audio Information Exchange)

The Last Extension Dialed feature will redial an associated extension number to access the AUDIX system.

Automatic Callback

Automatic Callback can be used in conjunction with Last Extension Dialed. This is useful when activation of the Automatic Callback feature toward the last extension dialed is desired. (Dial the Automatic Callback DAC or press the feature button before pressing the LXD feature button.)

Bridged Call

The extension number that the Last Extension Dialed feature stores and redials is kept on a per-equipment location basis. Therefore, even though a voice terminal user shares an appearance with another user, the stored extension number cannot be overwritten by a user sharing the appearance.

Call Forwarding—Busy and Don't Answer

Call Forwarding can be used in conjunction with Last Extension Dialed. This is useful when activation of the Call Forwarding—Busy and Don't Answer feature toward the last extension dialed is desired. (Dial the Call Forwarding—Busy and Don't Answer DAC or press the feature button before pressing the LXD feature button.)

Call Forwarding—Don't Answer

Call Forwarding can be used in conjunction with Last Extension Dialed. This is useful when activation of the Call Forwarding—Don't Answer feature toward the last extension dialed is desired. (Dial the Call Forwarding—Don't Answer DAC or press the feature button before pressing the LXD feature button.)

Call Forwarding—Follow Me

Call Forwarding can be used in conjunction with Last Extension Dialed. This is useful when activation of the Call Forwarding—Follow Me Answer feature toward the last extension dialed is desired. (Dial the Call Forwarding—Follow Me DAC or press the feature button before pressing the LXD feature button.)

Data Communications Access

The Last Extension Dialed feature cannot be used to redial calls to Data Communications Access trunk groups that use extension number steering.

Display—Voice Terminal

When using the Last Extension Dialed feature to place a call, the called extension appears on the display unit If the names database is loaded for the called extension, the associated name immediately replaces the number on the display.

DCS (Distributed Communications System)

Last Extension Dialed is not strictly a DCS transparency. However, the Last Extension Dialed feature does operate across a cluster of switches that use a 4- or 5-digit numbering plan.

Intercom-Dial

The Last Extension Dialed feature will not redial a Dial Intercom number.

Off-Premises Terminal

Calls using the Multidigit Steering feature cannot use the Last Extension Dialed feature (this includes calls directed to a trunk port off-premises terminal or to a Data Communications Access port).

Hardware Requirements

Multiappearance voice terminals are required for this feature.

Feature Administration

Assignment of the Last Extension Dialed feature is on a per-voice terminal basis. The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature. The Last Extension Dialed feature can also be administered using the Manager IV.

The following is the applicable MAAP and SMT procedure.

MAAP AND SMT PROCEDURE — LAST EXTENSION DIALED			
PROCEDURE	WORD	PURPOSE	SMT
054	2	Assigns the LAST EXT DIALED button to a voice terminal. The applicable encode is as follows: 1 Last Extension Dialed.	Yes

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — LAST EXTENSION DIALED	
PATH NAME	**PURPOSE**
terminal-change terminal buttons	Assigns the LAST EXT DIALED button to a voice terminal.

Last Number Dialed

Description

This feature enables a voice terminal user to easily redial the last internal or external call. This function is primarily used for (one button) redialing a busy number. It also frees the user from repeatedly locating telephone numbers in the directory.

The LND (Last Number Dialed) feature is usable in all cases where the LXD (Last Extension Dialed) button was used, and provides additional functionality.

Dialed Number Storage

The LND feature uses a register to record manually dialed digits. Even after cut-through, the LND storage register continues to record dialed digits until the 10-second interdigital time-out occurs. Any numbers that are not dialed before the 10-second time-out, are not recorded in LND memory. Thus, if the complete phone number is not dialed before the 10-second time-out and the LND feature is attempted, the number redialed is either incomplete (results in a time-out) or invalid (returns intercept tone).

Feature History and Development

This feature was first available for System 85 in the Issue 1.2 software of Release 2, Version 3. The Last Number Dialed feature replaces and enhances the last Extension Dialed feature.

User Operations

The following are the user operating procedures for this feature.

To Repeat the Last Number Dialed:

1. Go off-hook. [Dial tone]
2. Press the **[LND]** button,

   or

   Dial the LND dial access code. [Ringback tone]

To Review the Last Number Dialed on a Display Voice Terminal:

1. Remain on-hook.
2. Press the **[LND]** button. [Display shows the LND number.]

Considerations

Maximum Digit Storage

The LND feature can store a maximum of 20 manually dialed digits.

Feature DACs (Dial Access Codes)

Except for the networking features (AAR, ARS, and WCR), if a call is begun with a feature DAC, only the digits following the second dial tone (after the DAC has been accepted) are stored. If the Last Number Dialed feature is used to redial such calls, the DAC must be re-entered first.

Attendant Console

The LND feature cannot be assigned to an attendant console. However, calls placed to an attendant console from a voice terminal (that has LND capability) are recorded by the LND.

Redial Delay Timer

The administrable Redial Delay Timer provides a timing sequence between the trunk-group access coded the outpulsing of the destination number. This delay ensures that the distant switch is ready to receive the destination number. Once the trunk is seized by the distant switch and second dial tone is heard, ***cut-through*** is established.

Shared Appearances

The LND feature is assigned on a per-terminal basis. This ensures terminal users who share appearances that their LND memory will not be overwritten.

Hard or Soft Processor Swaps

When a hard processor swap or a power failure occurs, the last number dialed from each voice terminal is lost from memory. Therefore, pressing the LND button after a hard swap, tape reload, or disk reload (to initialize the switch) and before dialing another number has no effect.

The Last Number Dialed feature operates normally during a soft processor swap.

Applications and Limitations

The LND feature can be used to store and redial the following:

- External CO (Central Office), tie trunk, AAR (Automatic Alternate Routing), ARS (Automatic Route Selection), WCR (World Class Routing), or DCS (Distributed Communication Service) calls
- The destination part of a feature access call (digits following the second dial tone)

- Public-network destination for Call Forwarding-Off Net
- Station-to-attendant calls
- Station-to-station calls
- Trunk-group access codes.

The LND feature will not store or redial the following:

- AUDIX control sequences
- Authorization Codes
- Except for the networking features (AAR, ARS, and WCR), feature dial access codes. When a feature DAC (other than a network DAC) is used to initiate a call, the destination portion (the address digits) is saved. However, the feature dial access codes could destroy valid destination numbers and are, therefore, not retained.
- Digits dialed after full cut-through (after the 10-second interdigital time-out) or after dialing the "#" end of dialing digit
- Addressing received over a Remote Access facility
- CDR Account Codes
- Keyboard Dialing digits
- Trunk-test dial sequences.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

General

Except for the LND capability of redialing external calls, the LND feature interactions are similar to the LXD feature interactions. For many features' activations, only the destination number following the second dial tone is stored. (The initial feature access code will not be stored.) However, the LND feature will completely redial ARS calls, AAR calls, WCR calls, trunk-group access calls, and Precedence Calling calls.

Abbreviated Dialing

While feature dial access codes can generally be dialed using the Abbreviated Dialing feature, the LND (Last Number Dialed) feature dial access code (Encode 59) is specifically blocked. This access code can be programmed into Abbreviated Dialing lists and buttons. However, when Abbreviated Dialing is used to outpulse the digits of the LND access code, the call is blocked by the switch (intercept tone is returned). This operation avoids the possibility of recursive feature activation.

The LND feature records and redials station-to-station calls placed with the Abbreviated Dialing feature. However, other types of calls, including outgoing trunk calls and

Attendant Dial Access calls placed using the Abbreviated Dialing feature are ***not*** recorded and cannot be redialed using the LND feature. Except for extension numbers, the LND (Last Number Dialed) feature does not store and redial the digits stored behind an Automatic Dialing button. Instead, the previous manually dialed digits are stored. The Automatic Dialing call can be redialed by pressing the Automatic Dialing button again.

Except for extension numbers, the LND feature does not store and redial Abbreviated Dialing list-access calls. If a voice terminal user presses the List Access button or dials a List-Access code, the subsequently dialed digits are ignored by the LND feature. Instead, the LND feature will redial the previous manually dialed digits.

The LND feature can be used in conjunction with the Abbreviated Dialing feature. For example, a user could manually dial a trunk-group access code and then press an Automatic Dialing button to call a public-network telephone. If the user receives busy tone, this call can be retried by pressing the LND button and then pressing the Automatic Dialing button. When the LND feature is used with Abbreviated Dialing the LND feature only stores and redials digits that ***precede*** Abbreviated Dialing access.

APLT (Advanced Private Line Termination)

The LND feature can be used to redial outward network calls. If a distant network authorization code was dialed as part of the destination address during initial call setup, the number of dialed digits cannot exceed 20, including the network authorization code.

AUDIX (Audio Information Exchange)

The LND feature redials calls to an associated extension number or VDN (Vector Directory Number) of an AUDIX system. However, the LND feature does not redial the AUDIX user's extension number, password, or the previously dialed digits to access AUDIX.

Authorization Codes

For switch security, the LND feature does not stem or redial Authorization Codes.

AAR (Automatic Alternate Routing)

On System 85 and DEFINITY Generic 2.1 switches, when the LND feature is used to redial an AAR call, the LND feature sends all of the originally dialed digits (up to a maximum of 20 digits), including the AAR access code, to the AAR feature for routing. If an AAR function, such as subnetwork trunking, or IXC (Interexchange Carrier) Access, needs to modify the digit stream, the AAR feature performs the necessary modifications (internally) each time the number is redialed.

For switch security, the LND feature does not store or redial Authorization Codes.

ACD (Automatic Call Distribution)

The LND (Last Number Dialed) feature stores and redials calls to an associated extension number of an ACD split.

ACD agents can use the LND feature to minimize redialing calls to other ACD agents, to other departments, or to other locations outside the switch.

Since the ASSIST button is an Automatic Dialing button, the LND feature does not redial assistance calls to the split supervisor. Instead, the LND feature will redial the last digits that were manually dialed before the assistance call.

Automatic Callback

Automatic Callback can be used in conjunction with LND. This is useful when activation of the Automatic Callback feature toward the last number dialed is desired. (Dial the Automatic Callback DAC or press the feature button before pressing the LND feature button.)

ARS (Automatic Route Selection)

For System 85 and DEFINITY Generic 2.1 switches, when the LND feature is used to redial an ARS call, the LND feature sends all of the originally dialed digits (up to a maximum of 20 digits), including the ARS access code, to the ARS feature for routing. If an ARS function such as 10- to 7-digit conversion, ARS subnetwork trunking, or IXC (Interexchange Carrier) Access, needs to modify the digit stream, the ARS feature performs the necessary modifications (internally) each time the number is redialed.

For switch security, the LND feature does not store or redial Authorization Codes.

Call Forwarding—Busy and Don't Answer

Call Forwarding can be used in conjunction with Last Number Dialed. This is useful when activation of the Call Forwarding—Busy and Don't Answer feature toward the last number dialed is desired. (Dial the Call Forwarding—Busy and Don't Answer DAC or press the feature button before pressing the LND feature button.)

Call Forwarding—Don't Answer

Call Forwarding can be used in conjunction with Last Number Dialed. This is useful when activation of the Call Forwarding—Don't Answer feature toward the last number dialed is desired. (Dial the Call Forwarding—Don't Answer DAC or press the feature button before pressing the LND feature button.)

Call Forwarding— Follow Me

Call Forwarding can be used in conjunction with Last Number Dialed. This is useful when activation of the Call Forwarding—Follow Me Answer feature toward the last number dialed is desired. (Dial the Call Forwarding—Follow Me DAC or press the feature button before pressing the LND feature button.)

Data Call Setup

The LND feature does not redial keyboard dialing sequences because these sequences use Abbreviated Dialing to outpulse the digits.

DCA (Data Communications Access)

The LND feature completely stores and redials DCA calls to a host computer.

Digital Service (DS1) Interface

The LND feature completely stores and redials the digits dialed during a DS1 call. The LND feature stems the DS1 access code and the following destination number.

Display—Voice Terminal

When using the Last Number Dialed feature to place a call, the called number appears on the display unit. If the names database is loaded for a called extension, the associated name immediately replaces the number on the display.

If a user presses the LND button while the voice terminal is on-hook, the display shows the stored number for the button as "LND=stored number."

DCS (Distributed Communications System)

Last Number Dialed is not strictly a DCS transparency. However, the Last Number Dialed feature does operate across a cluster of switches that use a 4- or 5-digit numbering plan.

Extension Number Portability

The LND feature stores and redials 4- or 5-digit extension numbers dialed to voice terminals in a portability subnetwork.

Hold

The LND feature operates while a call is on "hard hold" (Call Hold or Multifunction Hold) or "soft hold" (Conference—3-Party / Transfer). The number of the last call (the call placed after the previous call was put on hold) is stored in LND memory.

HCA (Host Computer Access)

The LND feature completely stores and redials HCA calls to a host computer.

Intercom—Dial

The LND feature stores and redials intercom numbers dialed for an Intercom-Dial call.

ISDN—BRI (Basic Rate Interface)

ISDN—BRI voice terminals can be assigned a LND feature button and can use the LND feature. However, BRI voice terminals have a fixed ***Redial*** button that performs essentially the same function within the BRI set itself. One difference between the Redial button and the LND feature is when Station-to-Station calls are made using Abbreviated Dialing. These calls are recorded and can be redialed by the LND feature, while they are not when the Redial button is used.

IXC (Interexchange Carrier Access)

Redial by the LND (Last Number Dialed) feature sends all of the networking feature (AAR, ARS, or WCR) digits dialed (including network dial access code and IXC code if originally dialed) as though the call were manually redialed. With the AAR and ARS features, the IXC code cannot bw part of the dialed number. With the WCR feature, the IXC code may or may not be part of the dialed number.

IPA (Interpartition Access)

A voice terminal user (in an extension other than Extension Partition 0) can use the Last Number Dialed button to redial calls within the user's partition group, to Extension Partition 0, to the associated attendant partition, to Attendant Partition 0, or over an associated trunk group. When the user tries to use this button to redial an illegal call, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can use the Last Number Dialed button to redial calls to any extension partition, to any attendant partition or over any trunk group in the switch.

LWC (Leave Word Calling)

On activation of LWC, the extension dialed after hearing the second dial tone is stored in LND memory.

Override

On activation of Override, the extension dialed after hearing the second dial tone is stored in LND memory.

Personal CO Lines

The Last Number Dialed feature can be used to redial calls over Personal CO Lines that are administered as "rotary out."

The Last Number Dialed feature cannot be used to redial calls over Personal CO Lines that are administered as "touch-tone out." In this case, the serving CO scans Personal CO lines to detect an off-hook signal. After the CO recognizes an off-hook signal, the CO returns dial tone to the calling party. System 85 or Generic 2 software is not involved in this call-origination process. So, during the process, the switch does not invoke an originating register. Without this register, the switch cannot outpulse the stored digits.

Precedence Calling

The LND feature completely stores and redials the digits dialed during a Precedence Calling sequence.

Priority Calling

On activation of Priority Calling, the extension dialed after hearing the second dial tone is stored in LND memory.

Queuing

The Queuing feature handles external calls placed with the Last Number Dialed feature in the same way that calls placed with manually dialed digits are handled.

Remote Access

The LND feature does not store and redial the digits dialed during a Remote Access call.

Route Advance

The LND feature will store and redial a Route Advance call. The original trunk-group access code and the destination number are stored by LND.

Tenant Services

A voice terminal user (in an extension other than Extension Partition 0) can use the Last Number Dialed button to redial calls within the user's partition, to Extension Partition 0, to the associated attendant partition, to Attendant Partition 0, or over an associated trunk group. When the user tries to use this button to redial an illegal call, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can use the Last Number Dialed button to redial calls to any extension partition, to any attendant partition, or over any trunk group in the switch.

Through Dialing

The Last Number Dialed feature stores and redials the digits dialed by a voice terminal user during a Through Dialing connection. However, for a Through Dialing call, an attendant initially accesses a trunk group for the voice terminal user. Therefore, when digits are redialed using the Last Number Dialed feature, the trunk-group access digit(s) would not be included as outpulsed digits.

Trunk Verification—Voice Terminal

The LND feature does not store or redial trunk-test dial sequences. Once a trunk has been verified, dialing the digit sequence would not be useful.

Unattended Console Service—Call Answer From Any Voice Terminal

The LND feature does not store or redial the CAAVT access code at a voice terminal that responds to the general night-service alert. Instead, the previously stored digits will remain in LND memory.

WCR (World Class Routing)

The LND feature works with the WCR feature in the same way as it did with the AAR and ARS features on Generic 2.1 switches. When the LND feature is used to redial a network call, the LND feature sends all of the originally dialed digits (up to a maximum of 20 digits) including the network dial access code, to the WCR feature for routing. If a networking function, such as digit modification, or digit formating and sending is used to modify the digit stream, the WCR feature performs the necessary modifications (internally) each time the digits are redialed.

Hardware Requirements

None.

Feature Administration

The LND feature is assigned on a per-switch, per-trunk group, and per-voice terminal basis.

The per-trunk group option is the LND redial delay interval. This interval is between 0 and 5 seconds in 1/10-second increments (with a default of 0 seconds), and is assigned in Procedure 101, Word 2. The field that assigns the redial delay to a particular CO or tie trunk group can range from 0 to 50 with 1 equal to 0.1 seconds and 50 equal to 5.0 seconds. This option is needed only when users access a trunk group using the trunk group DAC. It is not used for AAR, ARS or WCR calls.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

ADMINISTRATION PROCEDURES — LAST NUMBER DIALED

PROCEDURE	WORD	PURPOSE	SMT
054	2	Assigns the LND feature button to a multiappearance voice terminal.	Yes
101	2	Assigns the LND Redial Delay Timer to a trunk group.	Yes
350	1	Assigns the first digit of the LND dial access code (if required).	No
350	2	Assigns the LND dial access code. The applicable encode is: 59 LND.	No

Notes:

Leave Word Calling

Description

The LWC (Leave Word Calling) feature allows internal callers (internal to the local switch or the Distributed Communications System) to leave messages for internal principals (the called parties) without the assistance of a secretary or Message Center agent.

Message Storage Options

Leave Word Calling messages can be stored on the local switch, an AP, or on an AUDIX Adjunct. The storage destination is an administered option. Table 75-A shows a comparison of the differences between the various message storage options available.

TABLE 75-A. Leave Word Calling Storage Option Attributes

Storage	AP		Switch		AUDIX	
Terminal Type	With Display	Without Display	With Display	Without Display	With Display	Without Dispiay
Return Calls	Yes	No	Yes	No	Yes*	Yes*
Hard Copy	Yes	Yes	No	No	No	No
DCIU Link Dependent	Yes	Yes	No	No	Yes	Yes
Message Capacity	38,000 and up	38,000 and up	6,000	6,000	15,770 and up	15,770 and up
Off Hours Access	Yes	Yes	Yes	No	Yes	Yes
Off Premises Access	N/A	Yes	N/A	Yes	N/A	Yes
Message Waiting Notification	Yes	Yes	Yes	Yes	Yes	Yes
Unified Messaging	Yes	Yes	Yes†	Yes†	Yes	Yes
LWC Cancel Capability	Yes	Yes	Yes	Yes	No	No

* This function requires the R1 V2 version of the AUDIX-L or -M system.
† See the Feature Interaction between LWC and Unified Messaging.

Each storage option offers some advantages under specific conditions. For example, Leave Word Calling on the switch is an economical means of message handling on smaller System 85s or DEFINITY Generic 2s where an AP or AUDIX adjunct would not otherwise be necessary. Message storage on either an AP or AUDIX adjunct provides significantly greater message storage capacity than does storage on the switch. For users equipped with display capable terminals, message storage on the switch or AP provides visual access to messages. The AUDIX option provides audible message retrieval capability, without going through a message center agent, for users on any voice terminal (with or without display capability). Storage on the AP provides a hard copy printout capability through the demand print function.

Storage On An AP

For Leave Word Calling messages stored on an AP, flexible administration is available on the AP that defines the conditions under which a message is considered "retrieved." The persons who are allowed to access Leave Word Calling messages are subdivided into three categories:

- The principal
- Message Center agents
- The Leave Word Calling administrator and other covering users.

For each of these categories, the customer can define on a ***per-system basis*** if accessing a message is considered retrieving the message. For example, the customer can designate that the principal and Message Center agents retrieve messages. Whereas, the Leave Word Calling administrator and other users access messages.

Storage On The Switch

For Leave Word Calling messages stored on the switch, there are two fixed categories:

- The principal

  The principal is allowed to retrieve messages.

- The Leave Word Calling administrator and other covering users

  The administrator and other covering users can only access messages.

Storage On AUDIX Adjunct

When messages are stored on the AUDIX Adjunct, they are not automatically removed. When a message is first entered, it is held in a ***new mail*** mailbox along with other messages not yet accessed. Once an AUDIX message has been accessed, it is moved to an ***old mail*** mailbox. It remains in the old mail mailbox until deliberate action is taken to delete it.

Conditions For Use

A calling party can activate Leave Word Calling under any of the following conditions.

- During the ringing of an unanswered call
- While receiving a busy tone
- When someone other than the called party answers
- During the Caller Response Interval before the call routes to coverage (see Call Coverage)

- After the covering user answers
- By the covering user, leaving a message to call the covering user
- By the covering user, leaving a message to call the calling party (using the Coverage Callback button).

Message Cancellation

A Leave Word Calling message (except those stored on AUDIX) can be canceled by the originator of the message at any time before the message has been retrieved. Messages stored on AUDIX cannot be canceled or retrieved by anyone except the addressee.

Message Access and Retrieval

An automatic message waiting lamp on the called voice terminal alerts the principal to a waiting message. For the Leave Word Calling feature to be fully effective, all personally assigned terminals in the switch should have a message waiting lamp. A message left for a principal without a message waiting lamp might never be retrieved, or be retrieved after it was too late for the needed response, simply because the principal has no way of knowing about it.

After all messages for a principal have been retrieved, the message waiting lamp goes out However, users may be able to ***access*** messages without necessarily ***retrieving*** messages.

Message Security

Message Center Service

Message security can be ensured when a principal calls a Message Center agent to access messages. The principal can be required to recite a password to the agent. The agent verifies the password before reporting the messages.

Demand Print

Message security can also be provided for demand printouts. The principal can be required to dial a password when requesting a printout. If the password option for demand printouts is assigned, a password is required for both types of demand printing: Demand Printout and Demand Print of Delivered Messages. The password can be up to eight characters in length.

With AUDIX

Message security is provided for AUDIX messages by requiring a password during the log-on procedure.

Display Lock/Unlock Option

The Lock/Unlock option reduces possible unauthorized retrieval of messages from a voice terminal with display capabilities (see the Display Voice Terminal feature). Dialing the Lock Message Retrival access code "locks" the voice terminal to message retrieval (or cancellation). When the voice terminal is locked, pressing the Message Retrieval button (or attempting to cancel a message) doesn't work. Dialing the Unlock Message Retrieval access code returns the Message Retrieval button to operation.

Feature History and Development

The Leave Word Calling feature was first available for System 85 in Release 1, Version 1. subsequent enhancements have included:

- Display—Voice Terminal feature. First available with Release 1.
- Leave Word Calling on the AP. First available with Release 1, Version 1.
- EDC (Electronic Document Communications). First available with Release 1, Version 3.
- Leave Word Calling on the Switch. First available with Release 2, Version 1.
- DCS Transparency for LWC. First available with Release 2, Version 1 for messages on the AP, and Release 2, Version 2 for messages on the switch.
- AUDIX. First available with Release 2, Version 2.
- Demand Print. First available with Release 2, Version 2.
- Unified Messaging. First available with Release 2, Version 2.
- Demand Print of Delivered Messages. First available with Release 2, Version 3, and retrofitted to Release 2, Version 2.

User Operations

The following are the user operating procedures for this feature.

To Activate Leave Word Calling

From a terminal with a LWC button:

1. After placing your call, press the **[LWC]** button. [Confirmation tone]
2. Go on-hook.

From a multiappearance terminal without a LWC button:

1. Go on-hook
2. Go off-hook. [Dial tone]
3. Dial the Activate LWC access code. [Second dial tone]
4. Dial the extension number where the LWC message is to be delivered. [Confirmation tone]
5. Go on-hook.

Using a single-appearance terminal without a RECALL button:

1. Momentarily press the switchhook. [Recall dial tone]
2. Dial the Activate LWC access code.
3. Dial the extension number where the LWC message is to be delivered. [Confirmation tone]
4. Go on-hook.

From a single-appearance terminal with a RECALL button:

1. Press the **[RECALL]** button. [Recall dial tone]
2. Dial the Activate LWC access code.
3. Dial the extension number where the LWC message is to be delivered. [Confirmation tone]
4. Go on-hook.

Leave Word Calling Without Placing an Initial Call:

Any of the preceding can be accomplished without placing a call to the other terminal first.

1. Press the **[LWC]** button,

   or

   Dial the LWC access code.
2. Dial the extension number where the LWC message is to be delivered. [Confirmation tone]
3. Go on-hook.

To Cancel a Leave Word Calling Message:

1. The message originator should go off-hook (see NOTE). [Dial tone]
2. Dial the LWC—Cancel access code

   or

   Press the **[LWC CANCEL]** button (if provided).
3. Dial the principal's (message destination) extension number. [Confirmation tone]
4. Go on-hook.

NOTE: Cancellation must be done from the same extension that was used to leave the message originally. Otherwise the switch will not be able to properly identify the message to be canceled.

Accessing Leave Word Calling Messages

To access messages through Message Center agent:

1. Call the Message Center extension number.
2. Request any messages.
3. The Message Center agent reports the messages verbally. Also, the agent can tell the principal whether EDC (Electronic Document Communications) or AUDIX messages have arrived.

To access messages using Demand Printout:

1. Dial the Demand Printing access code,

   or

   Press an Abbreviated Dialing button with this access code as the stored number.
2. Dial the extension number of the principal whose messages are being accessed.

   The printed report also notifies the principal if EDC and AUDIX messages have arrived.

To access delivered messages through Demand printout:

1. Dial the Demand Print of Delivered Messages access

   or

   Press an Abbreviated Dialing button with this access code as the stored number).
2. Dial the extension number of the principal whose messages are being accessed.

Retrieving Your Own Messages Via the Display—Voice Terminal Feature

Message Retrieval Without Scrolling:

1. Press **[MESSAGE RETRIEVAL]** . [Display shows Retrieval in Progress, Messages for (Name).]
2. Press **[NEXT MESSAGE]** (to display each message).

To delete displayed message from file:

Press **[DELETE MESSAGE]** . [Display shows DELETED.]

To return a call during Message Retrieval:

1. Select an idle appearance. (Dial tone)
2. Press **[RETURN CALL]** . (Extension number must appear in the displayed message.)

Message Retrieval With Scrolling:

1. Press **[MESSAGE RETRIEVAL]** . [Display shows Retrieval in Progress, the Messages for (Name).]
2. Press **[NEXT MESSAGE]** (to display each message).

   If message is continued, display shows continuation mark ">" in last field.
3. Press **[SCROLL]** (to display additional segments of displayed message).

   If SCROLL is pressed when there is no next segment, the display will show the next message (same as NEXT MESSAGE button).

Message Access By Covering User

To access principal's message file as a covering user:

1. Press an idle appearance button.
2. Press **[COVER MSG RETRIEVAL]** . [Display shows WHOSE MESSAGES?]
3. Dial principal's extension number (when display shows: MESSAGES FOR [Name]).
4. Press **[NEXT MESSAGE]** .

To access principal's message file during call with principal:

1. Press **[HOLD]** .
2. Press an idle appearance button.
3. Press **[COVER MSG RETRIEVAL]** . [Display shows WHOSE MESSAGES?]
4. Dial principal's extension number.
5. Press the held appearance button.

To return call for principal to displayed message originator:

NOTE: Originator's extension number (the number to which the call is being returned) must appear in the current message.

1. Press **[TRANSFER]** . [Dial tone]
2. Press **[RETURN CALL]** .
3. Press **[TRANSFER]** .

NOTE: When Leave Word Calling messages are stored on the switch, these messages must be accessed using a display module.

Securing the Message Retrieval Function on the Display

To lock the Message Retrieval function off the display:

1. Go off-hook on an idle line appearance. [Dial tone]
2. Dial the LOCK access code. [Confirmation tone]

   (Now, this terminal cannot be used to perform Message Retrieval or Cancellation functions.)

To unlock the terminal (return Message Retrieval):

1. Go off-hook on an idle line appearance. [Dial tone]
2. Dial the UNLOCK access code. [Confirmation tone]

Accessing Messages from AUDIX:

1. Log into AUDIX.
2. Scan the Incoming Mailbox.

   The AUDIX system reports the new messages and notifies subscribers of the presence of EDC and Message Center messages.

NOTE: When Leave Word Calling messages are stored on the AUDIX system, these messages are accessed by scanning the Incoming Mailbox.

Considerations

Confirmation Tone

With the Leave Word Calling feature, confirmation tone has a limited meaning. It means that the switch has accepted the Leave Word Calling message for the dialed extension. It does not mean that the message will (or even can) be delivered. The same is true for Leave Word Calling Cancellation. That is, confirmation tone means that the switch has accepted an instruction to cancel the last message left by the extension used to enter that instruction. It does not mean that the message has not already been accessed or retrieved. It also does not mean that the desired message will be canceled if it was originally left from a different extension.

Message Waiting Lamps

The Leave Word Calling feature works best when as many extensions as possible have a Message Waiting Lamp assigned. The switch will accept a message for any assigned extension. However, it cannot alert an extension user that a message is waiting unless a message waiting lamp is available. If message waiting lamps are not assigned to all extensions, some other system, such as regularly calling the Message Center for messages, must be used to ensure that messages are retrieved. As many as 10,500 automatic message waiting lamps can be assigned to voice terminals within a single switch.

With Multiple APs

When multiple APs are connected to one System 85 or DEFINITY Generic 2, messages are stored on the AP serving the called principal.

Redundant Messages

On the AP: Messages left more than once from the same extension number to the same principal are not stored repeatedly on the AP. Only the most recent message is stored along with the number of times the calling party left the message.

On the Switch: Messages left more than once from the same extension number to the same principal are stored a maximum of nine times. If the calling party attempts LWC more than nine times, the most recent nine messages are retained.

Global Message Access

Persons authorized to access messages for anyone on the switch do not have to be included in the coverage path (see Call Coverage feature) for a principal to access that principal's messages.

LWC Messages on Redirected Calls

Messages are always stored for the originally dialed voice terminal, even if the call has been redirected to another voice terminal by Call Forwading—Follow Me, Hunting, Call Pickup, or Call Coverage.

Fixed Standard Messages

The content and format of Leave Word Calling messages are predefined in switch software. The switch administrator or a user cannot change the standard information within or the format of these messages.

Switch Message Capacity

When the storage destination for LWC is the switch, up to 6,000 messages can be stored at any one time in the switch memory. An individual principal can have as many as 16 messages at any one time. When either of these limits is reached, the switch denies activation of LWC by returning intercept tone.

LWC in a DCS Environment

On the AP: In System 85 and DEFINITY Generic 2 switches, the calling partys' names and network numbers are stored on the called party's local AP. This mode of operation is preferable and recommended, since it provides transparency. When the originating node is a System 85, Release 2, Version 2 or later switch, the callers' names are also associated with calling number messages, but these names will only be used by the distant switch (not the distant AP).

On the Switch: DCS transparency of LWC on the switch is available when every switch in the DCS is System 85, Release 2, Version 2 or later. These switches send, transfer, and receive "name-messages" on internodal calls. These messages provide the same information that a local AP would. That is, the caller's name is associated with the calling extension and is provided to the called party on message retrieval. However, transparency is lost if all nodes involved (including tandeming nodes) are not Release 2, Version 2 or later. Earlier System 85 switches (and DIMENSION System switches or System 75 switches earlier than Release 1, Version 2) cannot handle name messages. When one of these switches is in the path, the name message is lost.

On AUDIX: Full DCS transparency is not available for messages stored on AUDIX Only the extension number of the calling party is passed to the AUDIX adjunct. The name is not provided on internodal calls. If DCS transparency is required, AUDIX subscribers can use the AUDIX system for Voice Mailbox and Call Answering while using Leave Word calling on the AP.

In Mixed Networks: When a DCS network is composed of different kinds of switches (such as, DEFINITY Switch, System 85, System 75, and DIMENSION System), or different versions of the System 85, Release 2 switch, transparency for Leave Word Calling will vary based on the specific switches used.

Hard and Soft Processor Swaps

The Lock / Unlock option for message retrieval on display sets is stored in a status portion of switch memory. When a hard processor swap occurs, every display voice terminal will be in the locked mode after the hard swap is finished.

The Leave Word Calling feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

AUDIX (Audio Information Exchange)

If a principal's LWC messages are stored on the AUDIX system, the originator of a LWC message cannot cancel the message. When the message originator tries to cancel the message, the switch will accept the request with confirmation tone. However, the LWC message is not removed from the principal's AUDIX mailbox. During message retrieval for LWC messages stored on the switch, the display will notify the user whenever there are also AUDIX messages to be retrieved. If this notification contains the AUDIX associated extension number [or VDN (Vector Directory Number)] the user can press the RETURN CALL button to easily access the AUDIX system.

Call Coverage

Leave Word Calling complements the Call Coverage feature by allowing the calling party to request a return call during or before the coverage process. Leave Word Calling messages are directed to the principal originally called even when calls redirect via Call Coverage. The only exception is calls redirected to an attendant. Leave Word Calling is not allowed when a call is redirected to an attendant.

Coverage Message Retrieval

Coverage Message Retrieval is denied to any covering user not in the principal's coverage path, except for "Global Retrievers."

LWC on the Switch

Leave Word Calling can be provided on a switch that has no AP. Messages are stored on the switch Global retrievers can retrieve messages for anyone in the switch. This ability is useful for message center agents who may support users for whom they are not covering users (Note that message retrieval cannot be assigned to an attendant console. A separate display voice terminal or data terminal must be used for this purpose.) Global retrievers need only one COVERAGE MSG RETRIEVAL button. The same button works for global retrieval, as well as for coverage-path retrieval. All messages accessible by Coverage Message Retrieval can be accessed globally.

Call Forwarding—Busy and Don't Answer

Within the local switch, Leave Word Calling messages are directed to the principal originally called even when calls redirect via Call Forwarding—Busy and Don't Answer. The only exception to this is calls redirected to the attendant. Leave Word Calling is not allowed when a call is redirected to the attendant.

Call Forwarding— Don't Answer

Within the local switch, Leave Word Calling messages are directed to the principal originally called even when calls redirect via Call Forwarding—Don't Answer. The only exception to this is calls redirected to the attendant. Leave Word Calling is not allowed when a call is redirected to the attendant.

Call Forwarding— Follow Me

In a non-DCS environment, Leave Word Calling messages are directed to the prinapal originally called even when calls redirect via Call Forwarding—Follow Me. The only exception to this is calls directed to an attendant. Leave Word Calling is not allowed when a call is redirected to an attendant.

In a DCS Environment

In a DCS environment without APs, if a call is forwarded between nodes, activation of Leave Word Calling is denied toward the originally called voice terminal.

Call Pickup

Leave Word Calling messages are always addressed to the principal originally called, even when a call redirects via Call Pickup.

Conference—Three Party

In a DCS environment without APs, after an internode 3-way conference is established, activation of Leave Word Calling is denied.

DCS (Distributed Communications System)

In the DCS with all System 85, Release 2, Version 2, or later switches, transparency is provided between switches with APs. However, with Version 2 and 3 switches, messages can only be accessed from the node on which the message is stored. The message indication sent by the switch (for example, light the MESSAGE lamp) can only be sent to a voice terminal within the node on which a message is stored; the message indication cannot be sent to another node in the DCS. Also, if the attendant extends a call to another node, activation of Leave Word Calling is denied.

In the DCS environment, where one or more switches do not have APs, LWC can still be provided. In this case, the LWC messages are stored on the switch or the AUDIX adjunct rather than on the AP. However, the caller's name is not always included as part of the LWC message. When the caller's name is not available, the LWC message includes only the caller's extension number.

For LWC on the switch only, if all the switches involved are Release 2, Version 2, or later, the assigned name for the extension is also included in the message.

For DCSs consisting only of System 85, Release 2, Version 4 and DEFINITY Generic 2 switches, using the ES (Enhanced Services) message set, centralized messaging provides DCS transparency for Leave Word Calling.

In mixed DCSs (where other types of switches may be involved in the connection) DCS transparency can be lost depending on what type of switch is used.

LWC on the Switch and Global Retrieval

Global retrieval of LWC messages is not a DCS transparency. Within a DCS cluster, each node can have a global retriever who can access messages stored within the local switch's memory. However, global retrievers cannot access LWC messages outside the local node.

DCS and Call Coverage

A covering user on a node where LWC messages are stored on the switch cannot activate coverage callback when the caller is on a remote node. The covering user may activate LWC.

Hunting

Leave Word Calling messages are directed to the principal originally called even when calls redirect via the Hunting feature. The only exception to this is calls redirected to an attendant. Leave Word Calling is not allowed when a call is redirected to an attendant.

ISDN—PRI (Primary Rate Interface)

The Leave Word Calling feature is a switch feature and not an ISDN feature in System 85 R2 V4 and DEFINITY Generic 2. Leave Word Calling will work within a DCS with ISDN connections (through the interworking function) but not for ISDN calls to or from the public network or a non-DCS private network

IPA (Interpartition Access)

Leave Word Calling messages can only be left for an extension in the same partition group or for an extension in Extension Partition 0. When Leave Word Calling is blocked by partitioning, the switch returns intercept treatment to the calling party.

Demand Print

The Demand Print function of the Leave Word Calling feature is also partitioned as part of the Tenant Services feature.

When a demand printout is requested from a voice terminal (in a partition other than Extension Partition 0), System 85 or DEFINITY Generic 2 software checks to determine whether the extension used to retrieve the messages and the extension for which the messages were left are in the same partition group. If not, the switch returns intercept treatment to the party requesting the demand printout.

From a voice terminal in Extension Partition 0, a demand printout can be requested for messages left to any extension in the switch.

Given the preceding operation, there are two approaches to implementing the Demand Print function on a partitioned switch. One approach is to provide a voice terminal and an associatedi printer for each partition group. The other approach is to provide a voice terminal and an associated printer to Extension Partition 0, and share this facility with all of the partition groups.

> **NOTE:** Access to a printer cannot be shared by several partition groups and denied to the other partition groups.

Last Number Dialed

When a user activates LWC, the LWC access code is not stored and redialed by the LND (Last Number Dialed) feature. The extension number dialed after the second dial tone is stored in LND memory.

LWC on the Switch and Call Coverage

Global retrievers need only one COVERAGE MSG RETRIEVAL button. The same button works for global retrieval, as well as for coverage-path retrieval. All messages accessible by Coverage Message Retrieval can be accessed globally.

LWC Without an AP and Traffic Reports

Traffic routines count the number of messages accessed by principals and the number accessed by covering users. Messages accessed globally by the authorized users are counted as coverage message retrievals.

Tenant Services

Leave Word Calling messages can only be left for an extension in the same extension partition or for an extension in Extension Partition 0. When Leave Word Calling is blocked by partitioning, the switch returns intercept treatment to the calling party.

Demand Print

The Demand Print function of the Leave Word Calling feature is also petitioned as part of the Tenant Services feature.

When a demand printout is requested from a voice terminal (in a partition other than Extension Partition 0), System 85 or DEFINITY Generic 2 software checks to determine whether the extension used to retrieve the messages and the extension for which the messages were left are in the same extension partition. If not, the switch returns intercept treatment to the party questing the demand printout.

From a voice terminal in Extension Partition 0, a demand printout can be requested for messages left to any extension in the switch.

Given the preceding operation, there are two approaches to implementing the Demand Print function on a partitioned switch. One approach is to provide a voice terminal and an associated printer for each extension partition. The other approach is to provide a voice terminal and an associated printer to Extension Partition 0, and share this facility with all of the extension partitions.

> **NOTE:** Access to a printer cannot be shared by several tenants and denied to the other tenants.

Transfer

In a DCS environment without APs, after an internode call is transferred, activation of Leave Word Calling is denied.

Unified Messaging

The Unified Messaging feature is provided between all message storage services. However, when LWC messages are stored on the switch, LWC message notification is not provided to the AP messaging services [EDC (Electronic Document Communications) or Message Center].

Restricting Feature Use

Covering users can be restricted from retrieving the principal's messages. This is done through the voice terminal translation using Procedure 000, Word 2.

Hardware Requirements

The Leave Word Calling feature uses the following additional or special hardware.

For Traditional Modules:

- An SN228B or SN229 Automatic Message Waiting Indicator interface circuit for each analog voice terminal (eight circuits per circuit pack)

For Universal Modules:

- A TN742 or TN746 Automatic Message Waiting Indicator interface circuit for each analog voice terminal (eight circuits per TN742 and 16 circuits per TN746)

Regardless of the Module Type:

- Voice terminals equipped with an automatic message waiting lamp
- Digital voice terminals with Voice Terminal Display capabilities, such as the 7405D voice terminal with 40-character D401A display module (optional for message retrieval), the 7407D IDT (Integrated Display Terminal), the 7506 or 7507 BRI (Basic Rate Interface) voice terminal, or the CALLMASTER voice terminal.
- The 510D or 515 BCT voice terminals (optional for message retrieval)
- An AP with Message Center software (unless LWC is used without an AP)

  NOTE: For demand printing of delivered messages, the AP must be 1D.12 or later.

- An AUDIX adjunct to store LWC messages (optional)
- TELETYPE® 443, 445, 450, or 460 printer(s) if the demand print option is used.

Feature Administration

Assignment of Leave Word Calling is on a line class-of-service basis. Assignment of the message-storage destination is on a per-voice terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

Leave Word Calling can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — LEAVE WORD CALLING			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to an extension.	Yes
000	2	Specifies the destination (switch, AUDIX adjunct, or AP) for LWC message storage for a voice terminal and designates Call Coverage message retrieval permission.	Yes
010	2	Assigns Leave Word Calling to a voice terminal class of service.	Yes
012	1	Assigns a name to an extension or trunk group.	Yes
012	2	Administers the name or identification for the extension or trunk group administered in Procedure 012, Word 1.	Yes
012	3	Compacts the Name Data Base software table to free as much memory space as possible to add additional names.	Yes
051	1	Assigns Leave Word Calling global retrieval and Lock / Unlock to a terminal.	Yes
054	1	Assigns the Leave Word Calling—Activate button to a multiappearance voice terminal. The applicable encode is as follows: 22 Leave Word Calling—Activate.	Yes
054	2	Assigns the Leave Word Calling—Cancel button to a multiappearance voice terminal. The applicable encode is as follows: 0 Leave Word Calling—Cancel.	Yes
063	1	Assigns an automatic message waiting lamp to a voice terminal.	Yes
256	1	Assigns the DCIU data link characteristics for Leave Word calling on AP.*	No
256	2	Administers the level 2 timers and counters for the DCIU link.	No
256	3	Administers the level 3 timers and counters for the DCIU link.	No

* A DCIU link is required for LWC on the AP or the AUDIX. See Appendix H for a detailed discussion of the DCIU and its administration.

(Continued)

ADMINISTRATION PROCEDURES
LEAVE WORD CALLING *(Continued)*

PROCEDURE	WORD	PURPOSE	SMT
257	1	Administers the components and priority status of the DCIU channel.	No
257	2	Administers the switch port for the DCIU channel.	No
257	5	Reserves a DCIU port for the LWC application (R2 V4 and DEFINITY Generic 2).	No
258	1	Copies the translation changes made using Procedures 256 and 257 to working tables.	No
258	2	Refreshes the DCIU temporary translation tables before using Procedures 256 and 257.	No
261	1	Provides the translation between an internal AUDIX or AP number and the network AUDIX or AP number.	Yes
275	1	Assigns DCIU to the system class of service.	Yes
275	3	Assigns Demand Print passwords to the system class of service.	Yes
350	1	Assigns the first digit of the feature dial access codes (if required).	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 10 Demand Printing 14 Demand Print of Delivered Messages 66 Activate Leave Word Calling 67 Cancel Leave Word Calling 68 Lock Message Retrieval 69 Unlock Message Retrieval.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — LEAVE WORD CALLING	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns Leave Word Calling to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change names compact	Compacts the Name Data Base software table to free as much memory space as possible to add additional names.
terminal-change names extension-names	Assigns a name or identification to an extension number.
terminal-change names trunk-group-names	Assigns a name or identification to a trunk group.
terminal-change terminal buttons	Assigns the Leave Word Calling—Activate and Cancel buttons to a voice terminal. Also, use this screen to assign the display feature buttons to a 7405D with display module or 7407D IDT. This screen is also used to assign the Automatic Message Waiting lamp to a voice terminal.
terminal-change terminal equipment	Assigns the LOCK and UNLOCK option to a voice terminal with display capability.

The following is the applicable FM path name used with the AP 16.

FM SCREEN — LEAVE WORD CALLING	
PATH NAME	**PURPOSE**
facilities-mgmt dciu link-assignments	Generates a report of the current DCIU link assignments.

Line Lockout

Description

The Line Lockout feature places a voice terminal in a lockout condition when the user goes off-hook and does not dial within 10 seconds or pauses for more than 10 seconds while dialing. This feature frees switching facilities for other calls. When the user's extension number is taken out of service, the switch returns intercept tone for 10 seconds as a warning to the user. Going on-hook for 0.3 seconds restores service.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

None.

Considerations

Busy Tone

Busy tone is heard by a voice terminal user when calling another voice terminal in the lockout state.

Returning a Voice Terminal to Service

The user may think the voice terminal is out of service when intercept tone is received, but there is probably nothing wrong with the voice terminal. The voice terminal can be returned to service by going on-hook then off-hook again.

Disconnected Party

When a user fails to go on-hook after the other party has disconnected, the user will be placed in a lockout condition after 10 seconds. No intercept tone is returned.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

An attendant call is not allowed to wait on a voice terminal that has been locked out.

Automatic Callback

Automatic Callback can be activated toward a voice terminal in the lockout condition. When the voice terminal in the lockout condition goes on-hook, the callback process begins.

To activate Automatic Callback, a voice terminal user goes off-hook, dials the Automatic Callback access code, dials the number of the busy voice terminal, and goes on-hook. If the user fails to go on-hook within 6 seconds, line lockout warning occurs. The warning is intercept tone for a duration of 10 seconds.

Busy Verification of Lines

Busy verification of a locked-out voice terminal makes an immediate talking connection to the locked-out voice terminal. No barge-in tone is heard. The ANS lamp lights at the console.

Call Forwarding—Busy and Don't Answer

A call directed toward a locked-out voice terminal with this feature active forwards to the designated terminal.

Call Forwarding—Don't Answer

A call directed toward a locked-out voice terminal with this feature active does not forward. Instead, the switch will return busy tone to the calling party.

Call Forwarding—Follow Me

A call directed toward a locked-out voice terminal with this feature active forwards to the designated terminal.

Call Waiting

A call is not allowed to wait on a voice terminal in a locked out state.

Override

Busy tone is returned to a caller using override toward a voice terminal in lockout.

Priority Calling

Busy tone is returned to a caller attempting a priority call toward a voice terminal in lockout.

Hardware Requirements

None.

Feature Administration

The Line Lockout feature is always provided. Feature administration is not required.

Line/Feature Status Indication

Description

This feature provides a visual indication of a call's status and/or a feature's status (activated or deactivated) for each button on a multiappearance voice terminal. The visual indication is provided by lamps located beside the button they monitor. This feature reduces the likelihood of inadvertent call interruption and serves as a reminder of what features are active.

All appearance buttons and some feature buttons have green status lamps. For appearance buttons, the green status lamp flashes during ringing, lights steadily when busy, and winks or flutters during a hold. For feature buttons, the green status lamp indicates when a feature is active or inactive.

All appearance buttons also have red status lamps. This lamp lights to indicate the appearance the user is connected to or will connect to when going off-hook. This lamp is idle if a feature is assigned to the button. Only one red status lamp may be lighted on one voice terminal at any one time.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

None.

Interactions With Other Features

None.

Hardware Requirements

The Line/Feature Status Indication feaure requires the use of multiappearance voice terminals.

Feature Administration

The Line/Feature Status Indication feature is provided automatically when an appearance or feature button is assigned to a multiappearance voice terminal.

Notes:

Look-Ahead Interflow

Description

The Look-Ahead Interflow feature is used in conjunction with ACDs that ***also*** use the Call Vectoring and the ISDN (Integrated Services Dial Network)/PRI (Primary Rate Interface) features.

With this arrangement, vector programming (at the switch that initially received an ACD call) first decides whether the interflow operation is necessary. After the need to interflow is determined, the interflow destinations are specified by one or more "route to" steps. However, before an ACD call is diverted to the VDN of another switch, the "sending switch" attempts to setup an ISDN call to the "receiving switch" over the appropriate ISDN—PRI trunk group. This process allows the receiving switch to decide whether it can adequately handle the diverted call. The receiving switch makes this decision according to vector programming within its own vector assigned to the same VDN specified in the "route to" step at the sending Switch.

If the receiving switch can handle the interflow call, the receiving switch accepts the call with a D Channel message, and the sending switch sends the call over a B Channel. If the receiving switch cannot handle the call, the receiving switch refuses the call with a different message. At this time, the sending switch either attempts to setup an ISDN call to another switch (according to subsequent "route to" steps) or executes the alternative action programmed within its own local vector.

Feature History and Development

The Look-Ahead Interflow feature is first available for System 85 Release 2, Version 4, Issue 1.3 and in DEFINITY Generic 2. Beginning with Issue 2.0 of Release 2, Version 4 System 85 and Issue 2.0 of DEFINITY Generic 2.1, a Look-Ahead Interflow call can route over any available preference in a routing pattern. Before this enhancement, a Look-Ahead Interflow call could only route over the first preference in a network routing pattern.

Required Features

The following is a list of the features that are required for Look-Ahead Interflow and a description of the conditions under which they are required.

- Call Vectoring

  Vector processing makes the decision to interflow at a sending switch. Likewise, vector processing makes the decision to accept or reject each Look-Ahead Interflow call at a receiving switch. Therefore, the Call Vectoring feature ***must be assigned*** at both switches.

- ISDN—PRI (Primary Rate Interface)

  In order to query the receiving switch, the network routing feature (AAR, ARS, or WCR) at the sending switch routes Look-Ahead Interflow calls over an ISDN—PRI trunk group in a routing pattern. Meanwhile, in order for the receiving switch to respond to this query, this switch must also be ISDN—PRI capable. Therefore, the ISDN—PRI feature ***must be assigned*** at every switch involved in a Look-Ahead Interflow call, and these switches must be interconnected with ISDN facilities.

- Look-Ahead Interflow

  The Look-Ahead Interflow feature (at the sending switch) queries a receiving switch before routing each Look-Ahead Interflow call. Therefore, the Look-Ahead Interflow feature ***must be assigned*** at the sending switch.

  The Look-Ahead Interflow feature can be invoked (at the receiving switch) to divert previously accepted calls outside the switch on a Look-Ahead basis. For the receiving switch to take the role of the ***second*** sending switch, also called the "tandeming switch," Look-Ahead Interflow ***must be assigned*** at the tandeming switch.

  For the answering agent at the receiving switch to hear 3-burst zip tone or to receive DNIS (Dialed Number Identification Service) from the sending switch, Look-Ahead Interflow ***must be assigned*** at the receiving switch.

- AAR (Automatic Alternate Routing)

  For System 85 and DEFINITY Generic 2.1 switches, the AAR feature is required to route ISDN—PRI calls over intertandem tie trunks in the private network. Since the trunk type of an ISDN—PRI private-network trunk group must be an ETN trunk type, the AAR feature is also required to tandem ISDN—PRI calls over these intertandem tie trunks. Therefore, to route Look-Ahead Interflow calls over the private network, the Standard Networking field (in Procedure 276) and the AAR feature ***must be assigned*** at the sending (or tandeming) switch.

- ARS (Automatic Route Selection)

  For System 85 and DEFINITY Generic 2.1 switches, the ARS feature is requiredd to route ISDN—PRI calls over the public network.

- WCR (World Class Routing)

  For DEFINITY Generic 2.2, the WCR feature is required to route ISDN—PRI calls over the public or private network.

Related Feature

The Look-Ahead Interflow feature is usually used with the ACD feature. Calls directed to an ACD split at the sending switch are interflowed (diverted) to an ACD split at the receiving switch if the set of conditions specified at both switches are met. These conditions may include time of day and day of week, number of staffed or available agents, or the amount of time the oldest call in queue has waited.

Look-Ahead Interflow Call Flow

Figure 78-1 shows a Look-Ahead Interflow configuration and describes a simplified call flow through the configuration. The three large boxes represent a sending switch, an intervening switch, and a receiving switch. (The intervening switch is dashed because this switch is not involved when there is a ***direct*** ISDN—PRI trunk group from the sending switch to the receiving switch.)

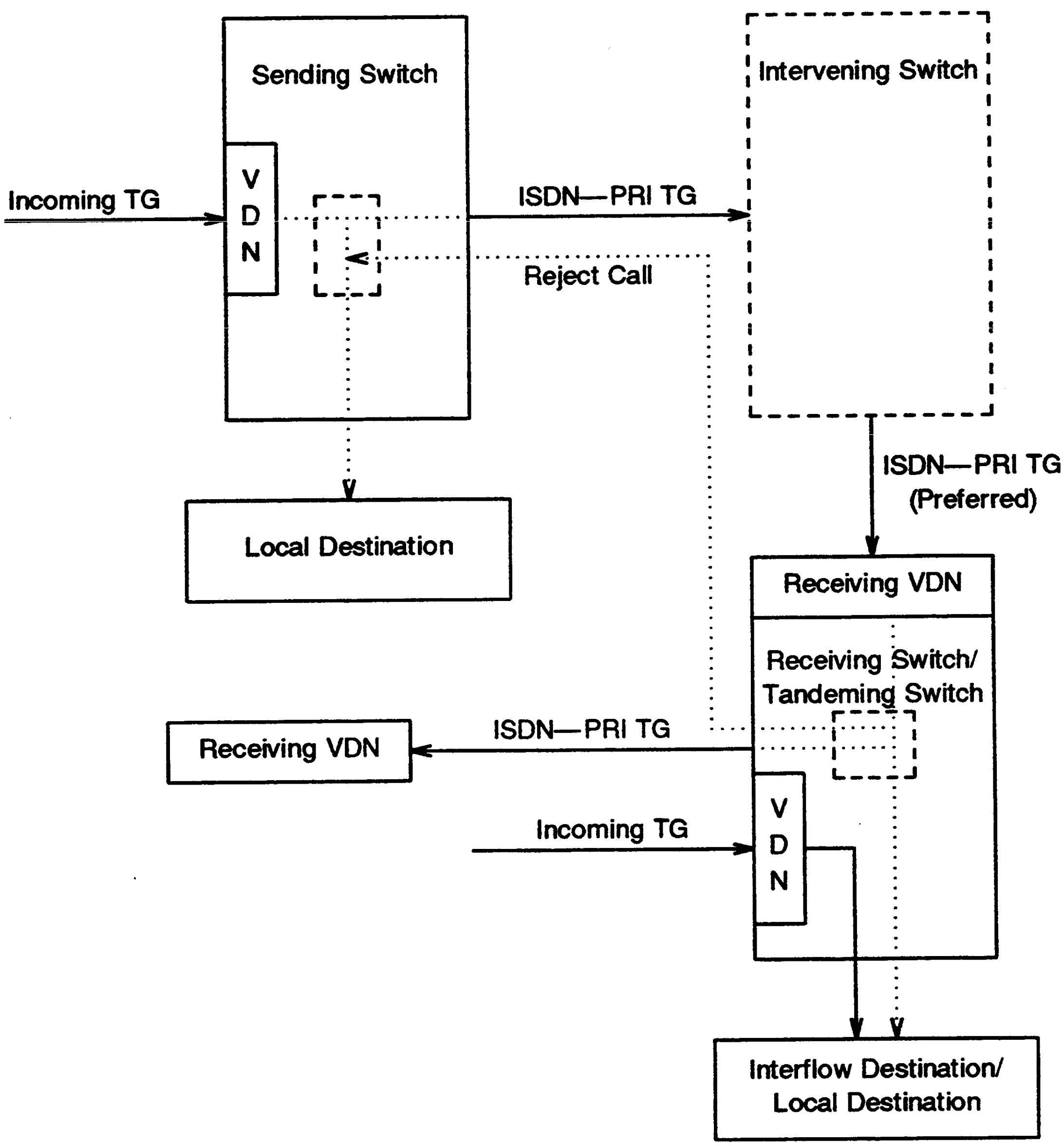

Figure 78-1. Look-Ahead Interflow Configuration

Smaller dashed boxes and dotted lines appear inside the sending switch and the receiving switch. The small dashed boxes represent the combination of Call Vectoring software and Look-Ahead Interflow Software at the sending and receiving switches. (Actually Look-Ahead Interflow software is optional at the receiving switch.) The dotted lines represent some of the possible paths that Look-Ahead Interflow calls can follow through the sending and receiving switches (based on the decisions that each working vector makes according to the current calling conditions).

To understand how Look-Ahead Interflow works, let's track the possible routes that an incoming call to a VDN (at the sending switch) can take. At the sending switch, an incoming call terminates to the VDN. When this happens, the Call Vectoring software (at the sending switch) gives the call to the VDN's vector for processing. Normally, this vector will deliver the call to a local destination (often, an ACD split). However, to handle periods of high calling volume, the vector can be programmed to check alternate destinations.

Alternate Destinations With Call Vectoring

Alternate destinations can (like the primary destination) be local to the sending switch. In fact, the vector at the sending switch could deliver excess incoming calls to:

- Another local ACD split (including an AUDIX) with a "check backup split" or "queue to main split" step (a "check backup split" step is preferred if the alternate destination is an ACD split staffed by human agents, because the answering agent hears 2-burst zip tone indicating that the call has been intraflowed)
- The local attendant queue with a "route to" step
- The local CAS queue with a "route to" step (if the sending switch is a main location in a CAS arrangement)
- The CAS queue at the main with a "route to" step (if the sending switch is a branch location in a CAS arrangement)
- A local extension number with a "route ID" step.

Sometimes, however, the vector programmer wants the vector to send excess calls ***outside*** the local switch. With Call Vectoring (and ***without*** Look-Ahead Interflow), this can also be done (on a call-by-call basis) with a "route to" step. The "route to" step passes the destination digits to the network routing feature, AAR (Automatic Alternate Routing), ARS (Automatic Route Selection), Main / Satellite, or WCR (World Class Routing) software to route the call. When AAR, ARS, or WCR are used, these calls route over an available preference (trunk group) in the corresponding network routing pattern.

However, the shortcoming with this type of interflow is that the network software routes the call even if the destination is too busy to handle it.

Querying the Receiving Switch

This is the point where Look-Ahead Interflow enhances the capabilities of a regular Call Vectoring "route to" step. With Look-Ahead Interflow, **all** preferences (in the network pattern that corresponds to the destination digits) ***should*** be ISDN—PRI trunk groups. With ISDN—PRI "end-to-end," the Look-Ahead Interflow software can take advantage of the D-channel messaging capabilities. After the network routing software at the sending switch selects an available ISDN trunk group, the ISDN software reserves a B Channel and queries the receiving switch.

> **NOTE:** If the destination at the receiving switch is the attendant queue, Look-Ahead Interflow calls are unconditionally accepted. The software that controls the attendant queue has no mechanism to reject incoming calls.

To "query" the receiving switch, the sending switch sends an ISDN—PRI CALL SETUP message, which includes a Look-Ahead Interflow IE (Information Element), across a D channel to the "receiving VDN" at the receiving switch.

Responding to the Query

When the receiving switch gets the CALL SETUP message, the ISDN—PRI software at the receiving switch carries out channel negotiation with the sending switch. Then the Call Vectoring software refers to the receiving VDN's vector to decide whether this switch can accept the call. In this description, the term "accept" applies only to vector processing. A call can be accepted (at the vector processing level) at one or more switches before answer supervision (CONNECT message) is returned to the sending switch. As partially shown in Figure 78-1, the results of this vector processing (on a call-by-call basis) show that the receiving switch can accomplish the following:

- Accept the Look-Ahead Interflow call, with the intent of delivering the call to a local destination by using:
  - A "queue to main split" step (PROGRESS message)
  - A "check backup split" step (PROGRESS message)
  - A "delay (wait) with silence" step (PROGRESS message)
  - A "delay (wait) with ringback" step (ALERT message)
  - A "delay (wait) with music" step (CONNECT message)
  - An "announcement" step (CONNECT message)
  - A "forced disconnect ***with*** announcement" step. (CONNECT message)
- After accepting the Look-Ahead Interflow call, decide to apply intraflow to the call (divert the call to a different local destination) by using:
  - A "queue to main split" step
  - A "check backup split" step
  - A "route to" step.

- After accepting the Look-Ahead Interflow call, decide to apply interflow to the call (divert the call to a distant destination on either a Look-Ahead* or a non-Look-Ahead basis) with a "route to" step.
- Reject the call using:
  - A "forced busy" step (DISCONNECT message)

If a "route to" step is the first step in the receiving vector, the receiving switch attempts to route the Look-Ahead Interflow call without accepting it. The receiving switch returns a PROGRESS message to the sending switch if the "route to" step is successful and the call routes over a non-ISDN trunk group. If the "route to" step is successful and an ISDN trunk group is selected, any ISDN messages received by the sending switch originate from a switch downstream of the receiving switch.

If an unsuccessful "route to" step is the final effective step in a receiving (or tandeming) vector that has not yet accepted the call, the switch does not retry the "route to" step. Instead, the switch returns a DISCONNECT message to the sending switch and the sending vector invokes its own failure disposition. Otherwise, the receiving (or tandeming) vector continues vector processing with the next step.

Responding to an Acceptance From the Receiving Switch

After the receiving vector executes a "queue to main split", "check backup split", "delay (wait) with silence", "delay (wait) with ringback", "delay (wait) with music", "announcement", "forced disconnect with announcement" step to accept the interflow call, the sending switch proceeds with ISDN signaling. Then, the sending switch treats the Look-Ahead "route to" step in its own vector as a successful step. So, the sending switch removes the incoming call (to the sending switch's VDN) from the queue where it may reside and discontinues vector processing for the call. Then, the sending switch passes the call back to the ISDN software which sends the call (over the reserved B Channel) to the receiving switch.

If an ACD agent at the sending switch becomes available ***during*** the look-ahead interflow call-setup process, the look-ahead interflow attempt must complete and be rejected before the call is distributed to the agent.

Responding to a Rejection From the Receiving Switch

After receiving the vector executes a "forced busy" step to reject the interflow call, the sending switch treats the Look-Ahead "route to" step in its own vector like an unsuccessful "route to" step. So, while the initial incoming call to the sending switch's VDN can ***remain*** in queue, vector processing at the sending switch either retries the unsuccessful "route to" step (if the final effective step) or continues with the next sequential step.

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* To divert the call on a Look-Ahead basis, the receiving switch must have Look-Ahead Interflow assigned.

Intervening Switches

Up to this point, this description has ***assumed*** ISDN—PRI connectivity from the sending switch through to the receiving switch. Certainly, this assumption is always valid in a private ETN network where ISDN—PRI trunk groups ***directly connect*** the sending and receiving switches. However, in many networking arrangements, one or more private or public-network switches ***come between*** the sending and receiving switches.

When this is the case, an intervening switch continues routing a Look-Ahead Interflow call according to its ***normal*** routing software [for example, AAR or WCR for an ETN tandem, or DNHR (Dynamic Nonhierarchical Routing) for a 4 ESS in the AT&T Switched Network].

However, the CALL SETUP messages for Look-Ahead Interflow calls always contain instructions to route these calls on an ***ISDN-Preferred*** basis. Therefore, as the intervening switch continues routing a Look-Ahead Interflow call, the intervening switch will always (when trunks within the trunk group are permitted ***and*** available) tandem the call to an ISDN—PRI trunk group first. Otherwise (without permitted and available ISDN facilities), the intervening switch will begin routing the call over a non-ISDN facility.

As the intervening switch begins to select a non-ISDN facility, the switch returns an "interworking" ISDN message (PROGRESS) to the sending switch.

If the sending switch receives a public- or private-network interworking message from the intervening switch, the sending switch considers the call accepted and relinquishes control of the call.

"Route To" Steps at the Sending Switch

Like the destinations for standard Call Vectoring "route to" steps, the digit addresses for the destinations of Look-Ahead Interflow "route to" steps are contained in the Abbreviated Dialing group list that is designated for use by the Call Vectoring feature (in Procedure 030, Word 1). In fact, the Look-Ahead Interflow feature and the Call Vectoring feature share the same group list containing 95 items.

Beginning with DEFINITY Generic 2.2, the number of Abbreviated Dialing group-list items that can be used for Call Vectoring has been increased from 95 (1 group list) to 475 (5 group lists).

For Look-Ahead Interflow, the list item is usually programmed to contain a VDN* at the receiving switch. (In turn, the vector assigned to the receiving VDN performs the inflow processing for the receiving switch.)

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* At the receiving switch, this VDN should be dedicated for handling inflow calls from the sending switch. Also, the calling-party interface within the sending and receiving vectors should be coordinated so that the calling party does not hear an unusual or unexpected sequence of tones and announcements.

The following are the acceptable formats of the programmed digit strings for Look-Ahead Interflow.

System 85 and DEFINITY Generic 2.1:

- 1-Digit AAR Access Code + RN(X) (Location Code) + XXX(X) (VDN at receiving switch)
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + NXX (Office Code) + XXXX (VDN at receiving switch)
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + NPA (Area Code) + NXX (Office Code) + XXXX (VDN at receiving switch)
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + International Telephone Number (Including a VDN at receiving switch)
- If the sending and receiving switches belong to the same DCS and/or ENP subnetwork, XXXX or XXXXX (VDN at receiving switch)

DEFINITY Generic 2-2:

- 1- to 4-Digit WCR Network Access Code + ("1") (Prefix Digit) + WCR address string.
- If the sending and receiving switches belong to the same DCS and/or ENP subnetwork, XXXX or XXXXX (VDN at receiving switch)

Whenever a vector-group list item for the Look-Ahead Interflow feature contains either a ***null*** or ***invalid*** digit string vector processing at the sending switch either treats the "route to" step as a "stop" step (if the final effective step) or continues with the next sequential step in the vector.

Since the vector processing skips "route to" steps with null destinations, the controller of the vector-group list can quickly make a "route to" step ineffective by reprogramming the corresponding list item without destination digits.

The AAR, ARS, or WCR software refers to the contents of the Abbreviated Dialing list item specified by the sending vector's "route to" step to route Look-Ahead Interflow calls outside the switch (over ISDN—PRI facialities). So, these Look-Ahead Interflow calls are affected by some of the networking feature preference-selection rules.

For more details about how Abbreviated Dialing, AAR, ARS, WCR and ISDN—PRI criteria affect the successful routing of "route to" steps, refer to Table 78–A.

Backup Interflow Destinations

Backup interflow destinations can be provided by programming more than one "route to" step (to an outside destination) in the vector at the sending switch. When this is done, the sending switch executes the "route to" steps in the order that they are encountered. If the receiving switch rejects the first "route to" step executed, vector processing continues with the next step in the vector (in this case another"route to" step), and the Look-Ahead Interflow process begins again with the new destination. (Refer to Vector C of the "Vector

Programming Examples for Sending Switches" for an example of backup interflow destinations.)

Interflow Tandeming With Accepting the Call

As previously mentioned, vector processing at a receiving switch can first accept a Look-Ahead Interflow call from the sending switch and then elect to interflow the call on a Look-Ahead basis. For this operation to succeed, the receiving vector must be properly programmed, and Look-Ahead Interflow ***must be assigned*** at the receiving switch.

The concept of interflow tandeming is merely a functional concept Actually, only one ***untandemed*** interflow operation can be active at any one time. After the original sending switch positively acknowledges the acceptance from the receiving switch and then connects the interflow call over the B Channel, the first interflow operation is finished.

When vector programming at the receiving switch elects to interflow the call again, this switch, called the "tandeming switch," becomes the new "sending switch." This new Look-Ahead Interflow call follows the same kinds of procedures and constraints as the first interflow call that the tandeming switch originally accepted. (Refer to Vector A of the "Vector Programming Examples for Tandeming Calls at Receiving Switches" for an example of interflow tandeming.)

Table 78-A shows the failure criteria for "route to" commands in a sending switch's vector.

Interflow Tandeming Without Accepting the Call

During a Look-Ahead Interflow call, the receiving switch's vector can be programmed to tandem the call by executing a Look-Ahead Interflow "route to" step before accepting the call. Although this operation can be performed, it is not recommended. A simpler, more direct, and less processor-intensive approach would be to program the destination of the sending vector's "route to" step as the destination of the tandeming vector's "route to" step, thereby eliminating the need for the tandem vector.

In this way, the routing patterns at the original sending switch can be engineered to select a direct ISDN—PRI route to the subsequent tandeming destination, and the original sending switch can maintain direct control of the Look-Ahead Interflow transaction. Even when a direct ISDN—PRI route to the tandeming destination is not available, the intervening switch need only apply its own network routing patterns to tandem the call on an ISDN-preferred basis.

TABLE 78-A. Criteria for Success / Failure of "Route To" Steps

Succeed/Fail Criteria	Vector Processing Disposition
Look-Ahead Interflow Not Assigned to Feature Group Class of Service (Procedure 276).	If "route to" step succeeds, call will route on a ***non-***Look-Ahead basis.
Vector-Group List Item Has Invalid Format.	If final effective step in sending vector (or tandeming vector that has already accepted the call), treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step. If final effective step in tandeming vector that has not yet accepted the call, the tandeming switch returns a DISCONNECT message to the sending switch and the sending vector invokes its own failure disposition. Otherwise, continue vector processing with the next sequential step.
Vector-Group List Item Contains Special Characters (such as, Pause, Wait, and Mark).	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
Vector-Group List Item Is Empty.	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
No Network Pattern for Destination Digits. (The currently active network plan defines the current pattern.)	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
Digits Imply a Toll Call and the Prefix Digit "1" is Required but the Network Software Does Not Receive the "1."	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
Digit String Marked for Unauthorized Call Control.	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.

TABLE 78-A. Criteria for Success / Failure of "Route To" Steps (Contd)

Succeed/Fail Criteria	Vector Processing Disposition
Selected Preference in Network Pattern Not ISDN—PRI Trunk Group. (The currently active network plan defines the current pattern.)	If "route to" step succeeds, call will route on a ***non-*** Look-Ahead basis.
Incoming trunk group and all outgoing trunk groups in the selected pattern have incompatible BCCOSs (Bearer Capability Classes of Service).	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
Local calling party's COS and outgoing trunk group have incompatible BCCOSs.	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
VDN's FRL Not High Enough to Access Any Preference.	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
VDN's Alternate FRL Not High Enough to Access Any Preference.	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
An Attendant Activated ACTGA Toward Trunk Group.	If final effective step in sending (or tandeming) vector, treat this step as a "stop" step. Otherwise, continue vector processing with next sequential step.
Trunk Group of Selected Preference Has Optional Information Inhibited in Procedure 100, Word 3.	If "route to" step succeeds, call will route on a ***non-***Look-Ahead basis.
For System 85 and DEFINITY Generic 2.1, the DCS Preference Does Not Have Both the "Network Trunk" and "Main / Tandem" Fields Assigned in Procedure 103, Fields 4 and 5.	If the "Network Trunk" field is "0," the call will fail. If the "Main / Tandem" field is "0" and if the "route to" step succeeds, DCS transparency is lost.

TABLE 78-A. Criteria for Success / Failure of "Route To" Steps (Contd)

Succeed/Fail Criteria	Vector Processing Disposition
All Accessible Trunk Groups are Busy.	If final effective step in sending vector (or tandeming vector that has already accepted the call), retry at 2-second intervals. Otherwise, continue vector processing with next sequential step. If final effective step in tandeming vector that has not yet accepted the call, the tandeming switch returns a DISCONNECT message to the sending switch and the sending vector invokes its own failure disposition. Otherwise, continue vector processing with the next sequential step.
An ISDN Glare Condition occurs (i.e., both ends of the ISDN—PRI trunk group simultaneously send a Call Setup message for the same channel). *	If this trunk group ***is not*** also a DCS trunk group, ISDN glare resolution allows the call from the Network side (the switch that asserts control) to succeed on the disputed channel. Also, the Network side initiates channel negotiation to find an alternate channel (if available) for the User side (the switch that backed off). If an alternate channel is not available, the User side's call is blocked. (The Network side instructs the User side to return reorder tone to the calling party). Upon receiving this message for a final effective vector step (when the sending switch is the User side), the sending switch will retry the "route to" step at 2-second intervals. If not the final effective step, the sending switch will continue vector processing with the next sequential step. If this trunk group ***is*** also a DCS trunk group, ISDN glare resolution still allows the call from the Network side to succeed. However, to prevent the loss of DCS transparency, the ISDN / DCS Call Setup message includes information to inhibit channel negotiation by the Network side. Therefore, the User side's call is blocked. At this time, for a final effective vector step (when the sending switch is the User side), the sending switch will retry the "route to" step at 2-second intervals. If not the final effective step, the sending switch will continue vector processing with the next sequential step.

* The corresponding vector-processing disposition assumes that the outgoing trunk group is an ISDN—PRI trunk group.

TABLE 78-A. Criteria for Success / Failure of "Route To" Steps (Contd)

Succeed/Fail Criteria	Vector Processing Disposition
There is Another Resource Failure (for example, no Originating Register.)	If final effective step in sending vector (or tandeming vector that has already accepted the call), retry at 2-second intervals. Otherwise, continue vector processing with next sequential step. If final effective step in tandeming vector that has not yet accepted the call, the tandeming switch returns a DISCONNECT message to the sending switch and the sending vector invokes its own failure disposition. Otherwise, continue vector processing with the next sequential step.
For System 85 and DEFINITY Generic 2.1, Procedure 103, Field 3, "Network Trunk" field, not assigned at Intervening ETN switch.	Intervening ETN switch cannot tandem the Look-Ahead Interflow call. Look-Ahead Interflow call fails.
FRL of Incoming Intertandem Trunk Group {at Intervening [Between Sending (or Tandeming and Receiving] ETN Switch} Not High Enough to Access ISDN Facility at Intervening ETN Switch.	Intervening ETN switch waits for FRL TCM and checks pattern's preferences again.
FRL TCM Not High Enough to Access Any Preferences at Intervening ETN switch. *	Intervening ISDN switch denies the call. The intervening switch instructs the sending switch to return Intercept Treatment to the calling party. If not the final effective step, the sending switch will continue vector processing with the next sequential step. Otherwise, vector processing stops.
FRL TCM Not High Enough to Access ISDN Facility at Intervening ETN Switch.	Interflow call continues routing over non-ISDN facility on ***non-*** Look-Ahead basis. The sending switch considers the call accepted
ISDN—PRI Route Not Available at Intervening Switch.	Interflow call will ***continue routing*** on a ***non-*** Look-Ahead basis. The sending switch considers the call accepted.

* The corresponding vector-processing disposition assumes that the outgoing trunk group (between the sending and intervening switches) was an ISDN—PRI trunk group.

TABLE 78-A. Criteria for Success / Failure of "Route To" Steps (Contd)

Succeed/Fail Criteria	Vector Processing Disposition
No Facilities Available at Intervening ETN switch.	If ETN switch with Queuing assigned, call can queue on an off-hook basis. Otherwise, if final effective step in sending vector and the call ***is*** ISDN end-to-end, retry at 2-second intervals. If not the final effective step, continue vector processing with next sequential step.
Receiving Switch Rejects Call With "Forced Busy" Step.	If final effective step in sending vector (or tandeming vector that has already accepted the call), and the call ***is*** ISDN end-to-end, retry at 2-second intervals. If not the final effective step, continue vector processing with next sequential step. If final effective step in tandeming vector that has not yet accepted the call, and the call ***is*** ISDN end-to-end, the tandeming switch returns a DISCONNECT message to the sending switch and the sending vector invokes its own failure disposition. Otherwise, continue vector processing with the next sequential step. If the call ***is not*** ISDN end-to-end, the receiving switch returns busy tone to the calling party.
None of the Above.	Exit vector processing. Pass control to call processing. Look-Ahead Interflow call succeeds.

Sample Applications of Look-Ahead Interflow

Vector Programming Examples for Sending Switches

Vector A: Providing Conditional Interflow for an ACD Split

1. queue to main split 31 at medium priority
2. go to step 4, if oldest call in main split's queue has waited more than 45 seconds at low priority
3. stop
4. route to 8 + RNX + XXXX [private-network number (usually including VDN) of the interflow destination]. (Most ETN networks have a 7-digit Uniform Numbering Plan.)

Step 1 queues calls to split 31 at medium priority.

Step 2 tests split 31 and branches to Step 4 if the oldest call in queue has waited more than 45 seconds at low priority or higher.

Steps 2 and 4 must be separated by a "stop" command, otherwise, calls would always interflow whether the test performed in Step 2 passed or failed.

Vector processing directs calls to Step 4 only when the oldest call in split 31's queue has waited more than 45 seconds (Step 2). Step 4 routes calls to a backup private-network destination.

Vector B: Providing Unconditional Interflow for an ACD Split

1. go to step 5, if there are no staffed agents in split 32
2. queue to main split 32 at medium priority
3. delay (wait) 30 seconds with ringback
4. stop
5. route to 9 + NPA + NXX + XXXX [public-network number (including VDN) of the interflow destination]. (A "1" prefix digit is not required on this switch.) (End-to-end ISDN facilities are required for Look-Ahead Interflow operation.)

Step 1 tests split 32 and branches to Step 5 if no agents are staffed.

If split 32 is staffed (the test specified in Step 1 fails), Step 2 queues calls to split 32 at medium priority.

Step 3 provides 30 seconds of ringback tone. Steps 3 and 5 must be separated by a "stop" command, otherwise, calls that wait in queue for more than 30 seconds would be routed to the backup private-network destination specified in Step 5.

Vector processing directs calls to Step 5 only when split 32 is unstaffed. Step 5 interflows calls to a backup private-network destination.

Vector C: Providing Conditional Interflow to Backup Destinations for an ACD Split

1. queue to main split 33 at medium priority
2. go to step 5, if more than 16 calls in queue at medium priority
3. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with ringback
4. stop
5. route to 8 + 374 + 7474 [private-network number (including VDN) of first-choice interflow destination]
6. route to 8 + 374 + 8585 [private-network number (including VDN) of second-choice destination on same ETN switch]
7. route to 9 + 1 + 415 + 256 + 7000 [public-network number (including VDN) or third-choice destination]. (A "1" prefix digit is required on this switch.) (End-to-end ISDN facilities are required for Look-Ahead Interflow operation.)

Step 1 queues calls to split 33 at medium priority.

Step 2 tests split 33 and branches to Step 5 if more than 16 calls are in queue at medium priority or higher.

Step 3 provides caller feedback (ringback tone). Step 3 must be separated from Steps 5, 6, and 7 by a "stop" command, otherwise, all calls would interflow, even when 16 or fewer calls are in queue.

Steps 5, 6, and 7 interflow calls to backup destinations. Calls are directed to Step 6 if Step 5 fails or if the receiving switch (associated with Step 5) rejects the call. Calls are directed to Step 7 if Steps 5 and 6 fail or if the receiving switches associated with these steps reject the call.

Vector D: Providing Interflow for the Older Calls in Queue

1. queue to main split 45 at low priority
2. delay (wait) 30 seconds with ringback
3. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
4. delay (wait) 60 seconds with music
5. route to 8 + 479 + 3737 [private-network number (including VDN) of the first-choice interflow destination]
6. route to 8 + 336 + 4243 private-network number (including VDN) of the second-choice interflow destination].

Steps 1 through 3 queue calls to split 45 at low priority and provide a delay announcement if calls wait in queue for more than 30 seconds.

For calls that wait in queue another 60 seconds, Steps 5 and 6 interflow calls to backup destinations. Vector processing directs a call to Step 6 if Step 5 fails or if the receiving switch (associated with Step 5) rejects the call.

Vector E: Emulating the Intraflow and Interflow Provided by Standard ACD

1. queue to main split 34 at low priority
2. go to step 5, if more than 20 calls in queue at low priority
3. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with ringback
4. stop
5. check backup split 35, queue at low priority if less than 5 calls in backup split 35's queue
6. check backup split 36, queue at low priority if there is more than one available agent in split 36
7. delay (wait) 20 seconds with ringback
8. route to 6432 (local extension number of third-choice interflow destination)
9. route to 8 + RN + XXXX private-network number (including VDN) of interflow destination]. (This ETN network has a 6-digit UNP.)

Step 1 queues calls to split 34 at low priority.

Step 2 tests split 34 and branches to Step 5 if more than 20 calls are in queue at low priority or higher.

Step 3 provides caller feedback (ringback tone) while calls wait in split 34's queue.

Step 4, the "stop" command, separates Steps 1 through 3 from the rest of the vector. Without this step, all calls would reach Step 5, no matter how many calls were in split 34's queue.

Step 5 checks backup split 35 and queues calls at low priority if fewer than 5 calls are in queue at low priority or higher.

Step 6 checks backup split 36 and queues calls if more than one agent is available. Multiple "check backup split" commands should be sequenced in descending order of preference. Usually, when the conditions of more than one "check backup split" command are met, the call queues to the first split in the sequence that meets the conditions. A call will only be queued to split 36 if 5 or more calls are in split 35's queue.

Step 7 provides ringback tone for 20 seconds while vector processing continues to check backup splits 35 and 36 every 2 seconds.

If the conditions specified in Steps 5 and 6 are not met within 20 seconds, control passes to Step 8 and calls are intraflowed to a local extension number.

If the local extension number is not immediately available, control passes to Step 9, which interflows calls to a backup private-network destination.

Vector F: From a Node in a DCS Subnetwork, Using a Centralized AUDIX System or Message Center Split for Night Service

1. go to step 5, if T.O.D. (time of day) between 4:00 p.m. and 8:00 a.m.
2. queue to main split 37 at low priority
3. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with ringback
4. stop
5. route to 64000 (VDN to access the centralized AUDIX System or Message Center Service in the DCS subnetwork). (This DCS uses 5-Digit Dialing.)
6. forced disconnect with announcement 20 ("We are closed for the evening. Please call back between the hours of 8:00 a.m. and 4:00 p.m.")

Step 1 tests for time of day. During business hours, 8:00 a.m. to 4:00 p.m., Steps 2 through 4 are executed. After business hours, 4:00 p.m. to 8:00 a.m., Steps 5 through 7 are executed.

Steps 2 and 3 queue calls to split 37 at low priority and provide caller feedback (ringback tone) until calls are answered.

Step 4, the "stop" command, separates business-hours processing from after-hours processing. Without this step, all calls would reach Step 5 and be routed to AUDIX or Message Center Service.

After business hours, control passes from Step 1 to Step 5. Step 5 routes calls to a centralized AUDIX or Message Center Service.

If Step 5 fails, Step 6 plays an announcement and disconnects calls.

Vector G: Providing Conditional and Unconditional Interflow in One Vector

1. go to step 8, if there are no staffed agents in Split 38
2. queue to main split 38 at low priority
3. go to step 8, if oldest call in main split's queue has waited more than 60 seconds at low priority
4. delay (wait) 20 seconds with ringback
5. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")

6. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with music
7. stop
8. route to 8 + RNX + XXXX [private-network number (including VDN) of the interflow destination].

Step 1 tests split 38 and branches to Step 8 if no agents are staffed. This step provides unconditional interflow when split 38 is unstaffed. Remember, whenever a "go to step" command that tests for number of staffed agents precedes the first "queue to main split" command a split number must be specified or the test will fail.

If the test specified in Step 1 fails (split 38 is staffed), control passes to Step 2 which queues calls to split 38 at low priority.

Step 3 tests split 38 and branches to Step 8 if the oldest call in queue has waited more than 60 seconds at low priority or higher. This step provides conditional interflow based on oldest call wait time.

Steps 4, 5, and 6 provide caller feedback (ringback tone), a delay announcement, and music until calls are answered.

Step 7, the "stop" command separates Steps 1 through 6 from the rest of the vector. Without this step, all calls would reach Step 8 and interflow to the backup destination.

Vector processing directs calls to Step 8 if split 38 has no staffed agents (Step 1) or the oldest call in split 38's queue has waited more than 60 seconds (Step 3). Step 8 interflows calls to a backup private-network destination.

Vector H: Combining the Conditions of Conditional Interflow

1. queue to main split 39 at low priority
2. go to step 8, if oldest call in main split's (split 39's) queue has waited more than 60 seconds at low priority
3. go to step 8, if the number of staffed agents (in split 39) is less than 6
4. delay (wait) 20 seconds with ringback
5. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
6. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with music
7. stop
8. route to 8 + RNX + XXXX private-network number (including VDN) of the interflow destination].

Step 1 queues calls to split 39 at low priority.

Step 2 tests split 39 and branches to Step 8 if the oldest call in queue has waited more than 60 seconds at low priority or higher. Step 3 tests split 39 and branches to Step 8 if fewer than 6 agents are staffed. A call will interflow if either or both of the conditions specified in Steps 2 and 3 are met.

Steps 4, 5, and 6 provide ringback tone, a delay announcement, and music until calls are answered.

Step 7, the "stop" command separates Steps 1 through 6 from the rest of the vector. Without this step, all calls would reach Step 8 and interflow to the backup destination.

Vector processing directs calls to Step 8 if split 39 has fewer than 6 staffed agents (Step 3) or the oldest call in split 39's queue has waited more than 60 seconds (Step 2). Step 8 intaflows calls to a backup private network destination.

Vector 1: Combining the Parameters for Unconditional Interflow

1. go to step 7 if T.O.D. between 7:00 p.m. and 7:00 a.m.
2. go to step 6 if T.O.D. between 4:00 p.m. and 7:00 p.m.
3. queue to main split 40 at low priority
4. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with ringback
5. stop
6. route to 8 + RNX + XXXX
7. forced disconnect with announcement 19 ("We are closed for the evening. Please call back between the hours of 7:00 a.m. and 7:00 p.m.")

Steps 1 and 2 test for time of day. Step 1 branches to Step 10 for calls that arrive between 7:00 p.m. and 7:00 a.m. Step 2 branches to Step 6 for calls that arrive between 4:00 p.m. and 7:00 p.m.

Vector processing directs calls to Step 3 only when the time-of-day tests specified in Steps 1 and 2 fail (Every day from 7:00 a.m. to 4:00 p.m.) Step 3 queues calls to split 40 at low priority.

Step 4 provides caller feedback (ringback tone) until calls are answered.

Step 5, the "stop" command, separates Steps 1 through 4 from the rest of the vector. Without this step, all calls would reach Step 6 and interflow to the backup destination.

Calls are directed to Step 6 between 4:00 p.m. and 7:00 p.m. This step interflows calls to a backup private-network destination after the office closes. The backup destination could be another local office that stays open later or an office in a different time zone.

Calls are directed to Step 7 between 7:00 p.m. and 7:00 a.m. All offices are closed during these hours, so this step provides a "We are closed" announcement and then disconnects calls.

Vector Programming Examples for Receiving Switches

Vector A: Accepting or Rejecting Look-Ahead Interflow Calls Based on the Number of Queued Calls

1. go to step 5, if more than 10 calls in split 41's queue
2. queue to main split 41 at high priority (receiving switch accepts call)
3. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with ringback
4. stop
5. forced busy (receiving switch rejects call).

Step 1 tests split 41 and branches to Step 5 if more than 10 calls are in queue. This test determines whether the receiving switch accepts or rejects a call. If the test fails, the call is accepted (Step 2). If the test passes, the call is rejected (Step 5).

If 10 or fewer calls are in split 41's queue (the test specified in Step 1 fails), control passes to Step 2 and calls are queued to split 41 at high priority. Because Look-Ahead Interflow calls are usually more expensive than other calls to a VDN, it is a good idea to program the receiving vector to queue these calls at a higher priority.

Step 3 provides ringback tone until calls are answered.

Step 4, the "stop" command separates the accept-call portion of the vector (Steps 1 through 3) from the reject-call portion of the vector (Step 5). Without this step, all calls would be rejected.

Vector processing directs calls to Step 5 when more than 10 calls are in split 41's queue. Step 5 rejects Look-Ahead Interflow calls by returning busy tone.

Vector B: Accepting or Rejecting Look-Ahead Interflow Calls Based on the Oldest Call Wait Time

1. go to step 7, if oldest call in split 42's queue has waited more than 60 seconds at low priority
2. queue to main split 42 at top priority (receiving switch accepts call)
3. delay (wait) 12 seconds with ringback
4. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
5. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with music
6. stop
7. forced busy (receiving switch rejects call).

Step 1 tests split 42 and branches to Step 7 if the oldest call in queue has waited more than 60 seconds at low priority or higher. This test determines whether the receiving switch accepts or rejects a call. If the test fails, the call is accepted (Step 2). If the test passes, the call is rejected (step 7).

If the test specified in Step 1 fails, control passes to Step 2 and calls are queued to split 42 at top priority.

Steps 3 through 5 provide ringback tone, a delay announcement, and music until calls are answered.

Step 6, the "stop" command, separates the accept-call portion of the vector (Steps 1 through 5) from the reject-call portion of the vector (Step 7). Without this step, all calls would be rejected.

Vector processing directs calls to Step 7 when the oldest call in split 41's queue has waited more than 60 seconds at low priority or higher. Step 7 rejects Look-Ahead Interflow calls by returning busy tone.

Vector C: Accepting or Rejecting Look-Ahead Interflow Calls Based on Time of Day

1. go to step 5, if T.O.D. between 9:00 p.m. and 7:00 a.m.
2. queue to main split 43 at medium priority (receiving switch accepts call)
3. delay (wait) 2 seconds (0 seconds beginning with DEFINITY Generic 2.2) with music
4. stop
5. forced busy (receiving switch rejects call).

Step 1 tests for time of day. This test determines whether the receiving switch accepts or rejects a call. If the test fails (between 7:00 a.m. and 9:00 p.m.) the call is accepted (Step 2). If the test passes (between 9:00 p.m. and 7:00 a.m.) the call is rejected (Step 5).

If the test specified in Step 1 fails, control passes to Step 2 and calls are queued to split 43 at medium priority. Because Look-Ahead Interflow calls are usually more expensive than other calls to a VDN, it is a good idea to program the receiving vector to queue these calls at a higher priority.

Step 3 provides caller feedback (music) until calls are answered.

Step 4, the "stop" command separates the accept-call portion of the vector (Steps 1 through 3) from the rejected portion of the vector (Step 5). Without this step, all calls would be rejected.

Vector processing directs calls to Step 5 between 9:00 p.m. and 7:00 a.m. Step 5 rejects Look-Ahead Interflow calls by returning busy tone.

Vector D: Using the Attendant Queue as a Backup Destination

1. go to step 8, if more than 12 calls in split 44's queue at low priority
2. queue to main split 44 at medium priority (receiving switch accepts call)
3. delay (wait) 16 seconds with ringback
4. check backup split 45, queue at medium priority if oldest call in backup split's queue has waited less than 40 seconds
5. delay (wait) 20 seconds with music
6. route to "0" (dial access code for local attendant queue)
7. stop
8. forced busy (receiving switch rejects call).

Step 1 tests split 44 and branches to Step 8 if more than 12 calls are in queue at low priority or higher. This test determines whether the receivingswitch accepts or rejects a call. If the test fails, the call is accepted (Step 2). If the test passes, the call is rejected (Step 8).

If the test specified in Step 1 fails, control passes to Step 2 and calls are queued to split 44 at medium priority.

Step 3 provides 16 seconds of ringback tone. This "delay" command gives agents in the main split time to answer the call before the backup split is checked.

Step 4 checks backup split 45 and queues calls at low priority if the oldest call in split 45's queue has waited less than 40 seconds at low priority or higher.

Step 5 provides 20 seconds of caller feedback (music).

If calls are not answered after 36 seconds (the combined wait time for Steps 3 and 5), Step 6 routes calls to the local attendant queue.

Step 7, the "stop" command, separates the accept-call portion of the vector (Steps 1 through 6) from the reject-call portion of the vector (Step 8). Without this step, all calls would be rejected.

Vector processing directs calls to Step 8 when more than 12 calls are in split 44's queue. Step 8 rejects Look-Ahead Interflow calls by returning busy tone.

Vector E: At a CAS Branch, Using the CAS Queue as a Backup Destination

1. go to step 8, if more than 12 calls in split 46's queue at low priority
2. queue to main split 46 at medium priority (receiving switch accepts call)
3. delay (wait) 16 seconds with ringback
4. check backup split 47, queue at medium priority if oldest call in backup split's queue has waited less than 40 seconds

5. delay (wait) 20 seconds with music
6. route to "*49" (dial access code for CAS queue at main)
7. stop
8. forced busy (receiving switch rejects call).

Step 1 tests split 46 and branches to Step 8 if more than 12 calls are in queue at low priority or higher. This test determines whether the receiving switch accepts or rejects a call. If the test fails, the call is accepted (Step 2). If the test passes, the call is rejected (Step 8).

If the test specified in Step 1 fails, control passes to Step 2 and calls are queued to split 46 at medium priority.

Step 3 provides 16 seconds of ringback tone. This "delay" command gives agents in the main split time to answer the call before the backup split is checked.

Step 4 checks backup split 47 and queues calls at low priority if the oldest call in split 47's queue has waited less than 40 seconds.

Step 5 provides 20 seconds of caller feedback (music).

If calls are not answered after 36 seconds (the combined wait time for Steps 3 and 5), Step 6 routes calls to the CAS queue at the main location.

Step 7, the "stop" command separates the accept-call portion of the vector (Steps 1 through 6) from the reject-call portion of the vector (Step 8). Without this step, all calls would be rejected.

Vector processing directs calls to Step 8 when more than 12 calls are in split 46's queue. Step 8 rejects Look-Ahead Interflow calls by returning busy tone.

Vector Programming Example for Tandeming Calls at Receiving Switches

Vector A: Combining the Conditions for Rejection, and Tandeming Accepted Calls Based on Accumulated Time in Queue

1. go to step 12, if T.O.D. between 5:00 p.m. and 8:00 a.m.
2. go to step 12, if oldest call has waited more than 70 seconds at low priority
3. queue to main split 50 at medium priority (receiving switch accepts call)
4. delay (wait) 20 seconds with ringback
5. announcement 16 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
6. delay (wait) 20 seconds with music
7. check backup split 51, queue at low priority if more than 1 available agent

8. delay (wait) 30 seconds with music
9. route to 8 + RNX + XXXX [private-network number (including VDN) of the interflow destination for this tandeming switch]
10. route to 9 + NPA + NXX + XXXX public-network number (including VDN) of another interflow destination for this tandeming switch]
11. stop
12. forced busy (receiving switch rejects call).

Step 1 tests for time of day and branches to Step 14 between 5:00 p.m. and 8:00 a.m. If the test specified in Step 1 fails, Step 2 tests split 50 and branches to Step 14 if the oldest call in queue has waited more than 70 seconds at low priority or higher. These tests determine whether the receiving switch accepts or rejects a call. If both tests fail, the call is accepted (Step 3). If either test passes, the call is rejected (Step 14).

If the tests specified in Steps 1 and 2 fail, control passes to Step 3 and calls are queued to split 50 at medium priority.

Steps 4 through 6 provide ringback tone, a delay announcement, and music. These steps give agents in the main split time to answer the call before the backup split is checked.

Step 7 checks backup split 51 and queues calls at low priority if more than one agent is available.

Step 8 provides 30 seconds of caller feedback (music). This "delay" command gives agents in the backup split time to answer the call before it is interflowed to a backup destination.

Steps 9 and 10 interflow calls to backup private-network destinations. Calls are directed to Step 10 if Step 9 fails or if the receiving switch (associated with Step 9) rejects the call.

Step 11, the "stop" command separates the accept-call portion of the vector (Steps 1 through 10) from the reject-call portion of the vector (Step 12). Without this step, all calls would be rejected.

Vector processing directs calls to Step 12 if the tests specified in Steps 1 and 2 pass (between 5:00 p.m. and 8:00 a.m. or if the oldest call in split 50's queue has waited more than 70 seconds). Step 12 rejects Look-Ahead Interflow calls by returning busy tone.

User Operations

The following are the user operating procedures for this feature.

To Program a List Item in the Vector-Group List

The controller of the vector-group list should:

1. Press the **[ABRVDIAL PROGRAM]** button,

   or

   Dial the Program access code (Encode 93). [Confirmation tone]

2. Press the **[GROUP]** list-selection button,

   or

   Dial the Group-List access code. [Dial tone]

3. Dial an item number (from 1 to 95) in the group list. [Second dial tone]

4. Dial the number to be used as a destination for a "route to" step. (Do not program ***special function*** characters.)

5. Press the **[GROUP]** list-selection button again to enter the new number,

   or

   Dial **[#]** if no list-selection button is provided. [Confirmation tone]

Considerations

Inhibited IEs in Procedure 100, Word 3

For each network routing preference, entering a "1" in Field 8 of Precedure 100, Word 3 inhibits optional ISDN information from being sent with calls placed over the trunk group. If this assignment is made for a network routing preference used with Look-Ahead Interflow "route to" steps, these calls will route on a non-Look-Ahead basis whenever they succeed.

Permanent Seizure Counter

To help a sending (or tandeming) switch recover from permanent trunk seizures, the Look-Ahead Interflow software uses the "Route To" Retry Counter of the Call Vectoring feature.

While a final effective "route to" step is being retried, this counter's task is invoked at 5-minute intervals. After the counter has incremented 6 times (approximately 30 minutes), the switch assumes a permanent seizure condition and tears down the connection.

Special Function Characters and "Route To" Commands

When an Abbreviated Dialing list item for Call Vectoring is programmed, the special function characters (for example, pause, wait, and mark) ***must not*** be used. The subnetwork trunking function of the network routing features (ARS, AAR, or WCR) automatically handle the timing to complete these calls. So, only the digits for a "route to" destination should be programmed. If special characters ***are*** programmed into a Call Vectoring group-list item, then the "route to" command that uses the list item will be considered to have an invalid destination.

Limited Amount of Vector Steps

Currently, without a "route to VDN*" step to chain vectors for additional vector steps, each vector can have up to 15 vector steps. In order to add the sequence of vector steps needed for one Look-Ahead Interflow destination to a preexisting vector, the switch administrator would normally add ***two*** steps to the vector: a conditional "go to" step and a "route to" step. Four vector steps are usually needed for a primary destination and two backup destinations a conditional "go to" step and three "route to" steps.

Therefore, when a sequence of vector steps for Look-Ahead Interflow is combined with an already complicated vector, the desired amount of vector steps may exceed the limit of 15.

Beginning with DEFINITY Generic 2.2, the "go to vector" command can be used to chain two or more vectors together. Like the "route to VDN command, check-backup split scanning stops when a "go to vector" command is invoked. Therefore, whenever scanning should proceed in the continuation vector, the "check backup split" step ***must be repeated*** in the continuation vector. However, unlike the "route to VDN" command, calls that were queued by the current (or a previous vector in the chain) remain queued when a "go to vector" command is invoked.

NFAS (Nonfacility Associated Signaling) Compatibility

The Look-Ahead Interflow feature is compatible with the NFAS function of the ISDN—PRI feature. Using a shared D channel, the NFAS arrangement provides call-control signaling for ISDN—PRI B channels residing in different DS1 spans. For example, the first of two DS1 interfaces could be configured to have 23 B channels and 1 D channel. Meanwhile, the second DS1 interface contains 24 B channels. This particular configuration is referred to as 47B + D. In the general case, NFAS can provide an nB + D arrangement between two ISDN—PRI switches, where $n \leq 479$ (up to 20 DS1 interfaces).

Providing a D-Channel Backup

Using NFAS arrangements with a single D-channel, the reliability of these configurations decreases as the number of B-channels controlled by the single D channel increases. Providing a D-Channel backup will minimize this risk. Whenever ISDN—PRI uses parallel DS1 spans to interconnect the same two switches, a primary D-channel can be assigned to one DS1 interface while a backup D channel is assigned to a different DS1 interface. If a failure occurs using

* Although "route to VDN" steps can provide additional vector steps, executing a "route to VDN" step removes the call from any ACD queue where it may reside and discontinues the execution of "check backup split" commands.

duplicated D channels, call-control signaling automatically switches from the failed D channel to the backup D channel.

If a backup D channel is provided, NFAS can provide ISDN—PRI arrangements of Up to 478B + D.

Providing Direct Routes for Look-Ahead Interflow Calls

The successful operation of Look-Ahead Interflow depends on ISDN—PRI connectivity from the sending switch through to the receiving switch. However, when the network routing software at the sending switch is assigned to route Look-Ahead Interflow calls to an intervening switch, this ISDN—PRI connectivity is no longer guaranteed. Instead, the intervening switch will continue routing these calls on an ISDN-Preferred basis. So, when the intervening switch has no permitted and available ISDN facilities for a call, the call continues routing on a non-Look-Ahead basis.

Therefore, whenever possible, private-network routing should be used to route Look-Ahead Interflow calls. Also, whenever possible, the routing preferences corresponding to the destination should be ***direct*** ISDN—PRI trunk groups to the receiving switch.

"Answer Supervision" and Incoming ISDN—PRI Trunks

When Call Vectoring controls call processing for an incoming ISDN call, the R2 V4 System 85 or DEFINITY Generic 2 returns answer supervision (a CONNECT message) just before the call is answered, a "delay (wait) with music" step, an "announcement" step, or a "forced disconnect with announcement"* step.

As the R2 V4 System 85 or the DEFINITY Generic 2 receives an incoming ISDN—PRI call destined for vector processing, the R2 V4 System 85 or DEFINITY Generic 2 returns a CALL PROCEEDING message to the serving switch. When the billing switch receives this message, it infers that the ISDN call has successfully negotiated for a B channel in the ISDN—PRI trunk group, but that answer supervision is not yet being returned. Then, just before an agent answers or a vector step executes, the R2 V4 System 85 or DEFINITY Generic, 2 returns an "Accept" (CONNECT, ALERTING, or PROGRESS) message to the serving switch. When the billing switch receives a CONNECT message, billing begins for the ISDN—PRI call.

Answer Supervision and Look-Ahead Interflow

At a sending tandeming or receiving switch, answer supervision is not returned to the billing switch until a CONNECT message is received. During a Look-Ahead Interflow call, the CONNECT message is sent just before the call is answered, or vector processing (at one of the switches) executes a "delay (wait) with music" step, an "announcement" step, or a "forced disconnect with announcement" step.

* The System 85 or DEFINITY Generic 2 does not return answer supervision for "forced disconnect without announcement" steps during ISDN—PRI calls. Using ISDN—PRI facilities, a call need not be answered to be disconnect.

Remember, a "forced disconnect without announcement" step will not return answer supervision for ISDN—PRI calls.

Coordination of Sending and Receiving Vectors

When Look-Ahead Interflow is used to divert calls to a VDN on another switch, the sending and receiving vectors should be coordinated to provide an appropriate interface to the calling party. As an example, if the calling party will have already heard ringback, a delay announcement, and then music before the interflow "route to" step is executed in the sending vector, then the receiving vector should not execute a "delay (wait) with ringback" step for the accepted calls. Instead, the receiving vector might execute another "announcement" step and then another "delay (wait) with music" step.

Applying a Higher Priority to Interflowed Calls

Since, on the average, interflow calls require more trunk facilities than regular incoming calls to a VDN, interflow calls are usually more expensive than regular rolls to a VDN. However, the switch administrator at the receiving switch can minimize this additional cost by programming the receiving vector to queue these calls to a split with a higher priority. In this way, the interflowed calls are handled as quickly as possible, and the use of the additional trunk facilities is minimized.

Interflow Patterns to Avoid

If vector processing at a receiving switch elects to tandem an interflow call to another distant destination, this destination probably shouldn't reside at the original sending switch. Even if the tandeming destination is ***different*** from the originally called VDN, trunk facilities in the network could have been saved by ***intra***flowing to this destination in the first place. However, if the tandeming destination is the originally called VDN, then this VDN's vector is likely to interflow the call right back to the tandeming switch. Looping the interflow call back to the original VDN can cause all the ISDN—PRI facilities between the two switches to become unavailable due to this one call. This type of loop-around routing must be avoided.

Testing the "Route To" Steps

Look-Ahead Interflow calls are invoked by a "route to" step with a destination outside the switch. Since the Abbreviated Dialing feature, one of the network routing features, and the ISDN—PRI feature are all involved in this process, the process can become quite complicated. If the routing process fails, the "route to" step is usually skipped in vector processing. However, some failures in the process can cause the interflow call to route on a non-Look-Ahead basis or to completely fail. Therefore, it is strongly recommended that each "route to" step (especially "route to" steps with previously untried NPAs, NXXs, NPA-NXXs, RN(X)s, or IDDD numbers) be fully tested before it is actually used to divert calls outside the switch.

Forced Entry of Account Codes

On System 85 and DEFINITY Generic 2.1, for Look-Ahead Interflow calls, FEAC (Forced Entry of Account Codes) should not be assigned to trunk groups in an AAR/ARS pattern or to the system class of service (for ARS calls).

Beginning with DEFINITY Generic 2.2, for Look-Ahead Interflow calls, FEAC (Forced Entry of Account Codes) should not be assigned to trunk groups in a WCR pattern or to a WCR network.

Hard and Soft Processor Swaps

Stable Look-Ahead Interflow calls over ISDN—PRI trunk groups endure a hard processor swap. However, calls cannot be placed over ISDN—-PRI trunk groups during a hard processor swap.

Interactions With Other Features

The following R2 V4 System 85 or DEFINITY Generic 2 features affect or are affected by the Look-Ahead Interflow feature.

Abbreviated Dialing

Look-Ahead Interflow "route to" steps share the Abbreviated Dialing group list that is designated as the vector-group list for Call Vectoring (in Procedure 030, Word 1).

To use one of these list items for Look-Ahead Interflow, the list item is programmed to contain a VDN at the receiving switch. In turn, the vector assigned to the receiving VDN performs the inflow processing for the receiving switch. The following are the acceptable formats of the programmed digit strings for Look-Ahead Interflow.

System 85 and DEFINITY Generic 2.1:

- 1-Digit AAR Access Code + RN(X) (Location Code) + XXX(X) (VDN at Receiving switch)
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + NXX (Office Code) + XXXX (VDN at Receiving Switch)
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + NPA (Area Code) + NXX (Office Code) + XXXX (VDN at Receiving Switch)
- 1- to 4-Digit ARS Access Code + ("1") (Prefix Digit) + International Telephone Number (Including a VDN at Receiving Switch)
- If the sending and receiving switches belong to the same DCS and/or ENP subnetwork, XXXXX (VDN at receiving switch)

DEFINITY Generic 2.2:

- 1- to 4-Digit WCR Network Access Code + ("1") (Prefix Digit) + WCR address string.
- If the sending and receiving switches belong to the same DCS and/or ENP subnetwork, XXXXX (VDN at receiving switch)

Whenever a vector-group list item for the Look-Ahead Inteflow feature contains either a ***null*** or ***invalid*** digit string vector processing at the sending switch either treats the "route to" step as a "stop" step (if the final effective step) or continues with the next sequential step in the vector.

Beginning with DEFINITY Generic 2.2, the number of Abbreviated Dialing group-list items that can be used for Call Vectoring has been increased from 95 (1 group list) to 475 (5 group lists).

ACTGA (Attendant Control of Trunk Group Access)

If the vector processing for Look-Ahead Interflow encounters a "route to" step where the call would route over a trunk group that is currently controlled by the attendant, the "route to" step is not executed. Instead, if the "route to" step is the final effective step in the sending switch's vector, the switch treats the "route to" step as a "stop" step. If vector steps follow the "route to" step, vector processing continues with the next sequential step.

At an intervening switch, if the only trunks available to route Look-Ahead Interflow calls are controlled by the attendant, calls will be routed to the attendant.

APLT (Advanced Private Line Termination)

Although APLT trunk groups can reside in network routing patterns, the APLT trunk types (12-15) cannot be assigned to ISDN—PRI trunk groups. Therefore, APLT trunk groups cannot be used to route Look-Ahead Interflow calls.

Attendant Display

At a receiving switch, using a "route to" step with the Attendant Dial Access code (Encode 8) as the destination, an incoming Look-Ahead Interflow call can reach the attendant queue. When this is done, the receiving switch will deliver the normal ICI (Incoming Call Identification) display associated with the incoming trunk group to the answering attendant.

AUDIX (Audio Information Exchange)

Using the Look-Ahead Interflow feature, a "route to" step within a sending vector can divert calls ***from a local AUDIX*** System to the VDN of an alternate destination outside the switch (usually within the private network).

Using the Look-Ahead Interflow feature, a "route to" step within a sending vector can divert (direct or redirected) calls from a local destination ***to the VDN of a centralized AUDIX System*** within the DCS subnetwork. [When a diverted call undergoes DCS routing the distant switch receives the reason for redirection, the identity of the originally called principal, and IMN (Integrated Message Notification) information in the DCS message.]

Authorization Codes

The Authorization Codes feature does not apply to Look-Ahead Interflow calls. The R2 V4 System 85 or DEFINITY Generic 2 does not return recall dial tone to request an Authorization Code for Look-Ahead Interflow calls.

If the VDN associated with the sending switch's vector does not have a high enough FRL to access an outgoing trunk, the Look-Ahead Interflow software considers the destination of the "route to" step invalid. (If the "route to" step is the final effective step in the vector, the step is treated as a "stop" step. Otherwise, the sending switch continues vector processing with the next sequential step.)

AAR (Automatic Alternate Routing)

For System 85 and DEFINITY Generic 2.1 switches, the AAR feature is required to route Look-Ahead Interflow calls through a private network. To provide the AAR feature, the "Standard Networking" field in Procedure 276 must be assigned. Beginning with DEFINITY Generic 2.2, the WCR feature is used to route Look-Ahead Interflow calls.

When the AAR feature is used to route Look-Ahead Interflow calls, the contents of a vector-group list item for a "route to" step must conform to the UNP (Uniform Numbering Plan) within the ETN network. When System 85s, DEFINITY Generic 2.2s, and DIMENSION FP8, Issue 3 switches are part of the ETN, the UNP can have one of the following forms:

- RNX (3-Digit Location Code) + XXXX (4-Digit Extension Number)
- RN (2-Digit Location Code) + XXXX (4Digit Extension Number)
- RNX (3-Digit Location Code) + XXX (3-Digit Extension Number)
- RN (2-Digit Location Code) + XXX (3-Digit Extension Number).

Besides conforming to the UNP for the ETN, the vector-group list items for "route to" steps must be prefixed by the AAR dial access code.

Besides conforming to the dialing plan for the private network, an AAR pattern must be translated for the destination digits of a "route to" step. When this is not done, the "route to" step is treated as having an invalid destination. If the "route to" step is the final effective step in the sending (or tandeming) vector, vector processing treats the step as a "stop" step. Otherwise, vector processing continues with the next sequential step in the vector.

At a sending (or tandeming) switch, outgoing-trunk queuing does not apply to Look-Ahead Interflow calls. Instead, if every trunk in the selected trunk group is busy, the Look-Ahead Interflow software will either retry routing the call at 2-second intervals (if final effective step) or continue vector processing with the next sequential vector step.

For voice calls, the BCCOS (Bearer Capability Class of Service) is not a significant consideration. This is because voice calls are usually compatible with any carrier facility. However, the AAR feature does check the BCCOS of calls that are diverted outside the switch by a "route to" step. Therefore, when applicable, the BCC of the selected trunk group must be compatible with the BCC in the local calling party's COS or the BCC assigned to the incoming trunk group.

The Look-Ahead Interflow feature is compatible with AAR Conditional Routing when assigned. For Look-Ahead Interflow calls, the AAR software will increment the conditional routing count whenever a "route to" step diverts a call over an AAR preference with a conditional route (for example, a route that contains a satellite link). Also, the AAR software will send the current value of the conditional routing count (if active for the selected trunk group) as the second TCM for Look-Ahead Interflow calls.

The Look-Ahead Interflow feature is compatible with AAR subnetwork trunking. For Look-Ahead Interflow calls that are not routed over intertandem tie trunks, the AAR subnetwork trunking function can internally modify the acceptable digit formats for vector-group list items so that the next ETN switch receives the expected digits.

As part of the Look-Ahead Interflow SETUP message, an intervening private-network switch is always requested to route the interflow call on ISDN-Preferred basis. Then, according to its AAR software, the intervening switch gives ISDN routes first preference during its route-selection process. If the private-network intervening switch cannot find an available ISDN route, the intervening switch will return a "Private-Network Interworking" message to the sending switch. For private-network calls, the sending switch will accept this message and allow the call to route on a non-Look-Ahead basis.

ACD (Automatic Call Distribution)

The Look-Ahead Interflow feature can divert ACD calls from and interflow ACD calls to ACD splits that use any distribution algorithm: direct, circular, or most idle agent. While a call resides in a particular ACD split's queue, that queue's distribution algorithm distributes calls to an available agent. When the call is diverted to another queue, that queue's algorithm assumes control of the call's distribution.

At a receiving switch, the Look-Ahead Interflow feature and the Abandon Call Search function of the ACD feature may not be compatible. Look-Ahead Interflow calls arrive at the receiving switch over ISDN—PRI facilities (that provide Positive Immediate Disconnect), and the receiving switch does not verify whether the calling party has abandoned before distributing a queued interflow call to an available agent.

However, for Look-Ahead Interflow calls, the incoming trunk facility to the sending switch ***may or may not*** be an ISDN facility with Positive Immediate Disconnect. If the incoming trunk group at the sending switch is not ISDN, the receiving switch could distribute a "ghost call" to the answering agent.

If an ACD agent becomes available at a sending (or tandeming) switch during ISDN messaging with the receiving switch, the agent at the sending switch is given precedence to answer the call. However, ISDN messaging is gracefully exited before the ACD agent at the sending switch receives the call. The sending (or tandeming) switch will wait to negatively acknowledge the receiving switch's message before distributing the call to the idle agent.

Since a Look-Ahead Interflow call always contains the Look-Ahead IE as part of the Call Setup message, the receiving switch can always identify the incoming call as an "interflowed" call. Therefore, for Look-Ahead Interflow calls that are queued to an ACD split at the receiving switch, the receiving switch provides 3-burst zip tone to the answering agent, if the Look-Ahead Interflow feature is assigned at the receiving switch.

The receiving switch for a Look-Ahead Interflow call can deliver a VDN-, city-, or queue-of-origin announcement to the answering ACD agent. At the receiving switch, multiple VDNs can access the same receiving vector. Therefore, separate VDNs with different corresponding announcements (translated in Procedure 033, Word 1) can be assigned for each sending switch or sending vector that interflows calls to the receiving vector.

At a receiving switch, the Look-Ahead Interflow feature and the Queue-Status Display function of the ACD feature are compatible. An ACD agent (with a display set and Queue-Status Display assigned to the class of service) at the receiving switch will normally receive the queue-status information for the local ACD split with incoming Look-Ahead Interflow calls. Table 78-B is a sample display with queue-status information.

TABLE 78-B. Look-Ahead Interflow Queue-Status Display Information

Type of Call	Display
ISDN call	a=212-281-7733 to DETROIT CLAIMS 17 045

However, within this Queue-Status Display, the length of time that the oldest call has waited (in this example, 45 seconds) does ***not*** include the amount of time that an interflowed call may have already waited at the sending switch.

At a receiving switch, the Look-Ahead Interflow feature and the Queue-Warning Lamp Control option of the ACD feature are compatible. When a Look-Ahead Interflow call enters an ACD queue at a receiving switch, the Queue-Warning Lamp software recognizes the interflowed call and includes this call in the threshold total for lighting the corresponding lamp on the 30A8 System Status Indicator.

At a receiving switch, the Look-Ahead Interflow feature and the Service Observing function of the ACD feature are compatible. A local voice terminal user (with a SERVICE OBSERVE button) at the receiving switch can normally monitor Look-Ahead Interflow calls that are answered by ACD agents (with multiappearance voice terminals) at the receiving switch.

At a receiving switch, the Look-Ahead Interflow feature and the Agent Override function of the ACD feature are compatible. A voice terminal user (with Agent Override in the class of service) at the receiving switch can normally monitor Look-Ahead Interflow calls that are answered by ACD agents (except split supervises) at the receiving switch.

At a receiving switch, the Look-Ahead Interflow feature and the Multiple Call Handling function of the ACD feature are compatible. An ACD agent (with a HOLD button and multiple call handling assigned to the agent's split) can put a Look-Ahead Interflow call on hold and become available to receive another ACD call from the split's queue. Conversely, the ACD agent can put a regular ACD call on hold and become available to receive a Look-Ahead Interflow call.

At a sending (or tandeming) switch, an ACD split that is assigned as Automatic Available (Procecdure 026, Word 2) can outflow Look-Ahead Interflow calls to/from the other switch. Likewise, at a receiving switch, an Automatic Available split can inflow Look-Ahead Interflow calls from a sending switch.

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, the ARS feature is required to route Look-Ahead Interflow calls through the public network. Beginning with DEFINITY Generic 2.2, the WCR feature is used to route Look-Ahead Interflow calls through the public network. In order to provide the 10- to 7-Digit Conversion function to route Look-Ahead Interflow calls (with public-network destinations) through the private network, the "Standard Networking" field (in Procedure 276) must also be assigned.

When the ARS feature is used to route Look-Ahead Interflow calls, the contents of a vector-group list item for a "route to" step must conform to the public-network rules for DDD (Direct Distance Dialing). The DDD formats for the public network can have one of the following forms

- NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- NPA (3-Digit Area Code) + NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- IXC (Interchange Carrier) + NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- International Telephone Number.

Other than conforming to the public-network rules for DDD, the vector-group list items for Look-Ahead Interflow "route to" steps must be prefixed by the ARS dial access code and (if required in Procedure 275, Word 3) a "1" prefix digit for toll calls.

Besides conforming to the dialing plan for the public network, an ARS pattern must not be translated to the Intercept Pattern for the first three or six digits specified within the destination digits of a "route to" step. When this is done, the "route to" step is treated as having an invalid destination. If the "route to" step is the final effective step in the sending (or tandeming) vector, vector processing treats the step as a "stop" step. Otherwise, vector processing continues with the next sequential step in the vector.

The ARS pattern selection for "route to" steps conforms to the currently active ARS routing plan. (This is the case whether the currently active ARS plan was invoked by an automatic plan change, clocked manual override, or manual override.) Therefore, whenever an ARS plan is active where the first three or six digits of a "route to" destination are translated to the Intercept Pattern, the "route to" step is considered to have an invalid destination. Also, whenever an ARS plan is active and the selected trunk group is not an ISDN—PRI trunk group, "route to" steps (if successful in diverting calls) will route the calls on a non-Look-Ahead basis.

The ARS routing of Look-Ahead Interflow "route to" steps can be blocked by Unauthorized Call Control. Whenever a vector-group list item for the Look-Ahead Interflow feature contains an ARS digit string that is marked for call control, vector processing at the sending switch either treats the "route to" step as a "stop" step (if the final effective step) or continues with the next sequential step in the vector.

For voice calls, the BCCOS (Bearer Capability Class of Service) is not usually a significant consideration. This is because voice calls are compatible with any carrier facility. However, the ARS feature does check the BCCOS of calls that are diverted outside the switch by a "route to" step. Therefore, when applicable, the BCC of the selected trunk group must be compatible with the BCC in the local calling party's COS or the BCC assigned to the incoming trunk group.

At a sending (or tandeming) switch, ARS Toll Restriction does not prevent "route to" steps from routing calls to destinations outside the switch. If ARS Toll Restriction is assigned to a VDN's class of service, the assignment is ignored.

At a sending (or tandeming) switch, outgoing-trunk queuing does not apply to Look-Ahead Interflow calls. Instead, if every trunk in the selected trunk group is busy, the Look-Ahead Interflow software will either retry routing the call at 2-second intervals (if final effective step) or continue vector processing with the next sequential vector step.

The Look-Ahead Interflow feature is compatible with the 10- to 7-Digit Conversion function of the ARS feature. When 10- to 7-Digit Conversion applies to the programmed destination of a Look-Ahead Interflow "route to" step, the 10-digit NPA-NXX-XXXX is converted to the corresponding RNX-XXXX and the new digits are passed to the AAR software for routing.

The Look-Ahead Interflow feature is compatible with subnetwork trunking. For Look-Ahead Interflow calls, the subnetwork trunking function can modify the contents of vector-group list items so that the next switch receives the expected digits.

As part of the Look-Ahead Interflow SETUP message, an intervening public-network switch is always requested to route the interflow call on ISDN-Preferred basis. Then, according to its routing algorithm, the intervening switch gives ISDN routes first preference during it route-selection process. If the public-network intervening switch cannot find an available ISDN route, the intervening switch will return a "Public-Network Interworking" message to the sending switch. For public-network calls, the sending switch will accept this message and allow the call to route on a non-Look-Ahead basis.

Call Coverage

The Look-Ahead lnterflow feature is compatible with the Call Coverage feature. At a sending switch, a VDN with a Look-Ahead Interflow "route to" step can be assigned as the final point in a coverage path. Moreover, when the Call Vectoring subroutine screens these redirected calls (on a call-by-call basis) to limit undesirable vector treatment of the calls, the subroutine will find the Look-Ahead Interflow "route to" step (whenever the interflow applies to the call) and allow the redirected calls to cover to the VDN. At this point, the "route to" step is executed for the redirected calls as it would be for direct calls to the VDN.

CDR (Call Detail Recording)

The sending switch only generates one CDR record for a Look-Ahead Interflow call, unless the "route to" step is retried. If the VDN call originated outside the switch, the switch records the incoming trunk-group dial access code as the calling number and the outpulsed digits from the Call Vectoring group list as the dialed number. If this incoming call is answered at the sending switch before the receiving switch accepts the call, the sending switch records the extension of the local answering destination as the dialed number.

Unless recording of ineffective attempts is assigned, the receiving switch only generates a CDR record for an accepted Look-Ahead Inteflow call. When a call is accepted, the receiving switch records the incoming trunk-group dial access code as the calling number and the extension of the local answering destination as the dialed number.

On System 85 and DEFINITY Generic 2.1, for Look-Ahead Interflow calls, FEAC (Forced Entry of Account Codes) should not be assigned to trunk groups in an AAR/ARS pattern or to the system class of service (for ARS calls).

Beginning with DEFINITY Generic 2.2, for Look-Ahead Interflow calls, FEAC (Forced Entry of Account Codes) should not be assigned to trunk groups in a WCR pattern or to a WCR network.

Call Forwarding—Busy and Don't Answer

At either a sending or receiving switch, the Call Forwarding—Busy and Don't Answer feature cannot be used to forward calls to a VDN.

Call Forwarding—Don't Answer

At either a sending or receiving switch, the Call Forwarding—Don't Answer feature cannot be used to forward calls to a VDN.

Call Forwarding—Follow Me

At a sending switch, the Call Forwarding—Follow Me feature can be used to forward calls to a VDN. If the VDN's associated vector contains a Look-Ahead Interflow "route to" step, calls that forwaded to the VDN will receive the same treatment from the "route to" step as calls that were directly dialed to the VDN.

Call Pickup

At a receiving switch, the Look-Ahead Interflow feature and the Call Pickup feature are compatible. A member of a Call Pickup group at the receiving switch can answer Look-Ahead Interflow calls that are ringing at another extension in the group.

Call Vectoring

The Call Vectoring feature is required at both the sending and receiving switches in a Look-Ahead Interflow configuration. For Look-Ahead Interflow, vector processing at the sending switch first decides whether the interflow operation is necessary. Subsequently, vector processing at the receiving switch makes the decision whether to accept or reject the Look-Ahead Interflow call.

The full flexibility of the Call Vectoring feature can be utilized at both the sending and receiving switches. Branches of any type can appear in the vector programs at both switches. Vector commands of any type can also appear within these vectors. (Refer to the "Sample Applications of Look-Ahead Interflow" portion of this feature description for examples of these capabilities.)

DNIS Names and VDN Override

For direct calls to a VDN, the VDN Override option of the Call Vectoring feature can always apply to a VDN call within the switch that originally received the VDN call. Before the sending switch executes a Look-Ahead Interflow "route to" step destined for the receiving switch, VDN Override can change the name associated with the originally called VDN.

As the Look-Ahead Interflow "route to" step is executed, the sending switch sends the most recent VDN name in the DNIS (Dialed Number Identification Service) Name field of the Look-Ahead Interflow IE (Information Element). When this name arrives at the receiving switch, this switch will use the received name regardless of its own VDN Override assignments. If the sending switch sends a blank DNIS Name field (because no names were assigned) in the Look-Ahead Interflow IE or the Look-Ahead Interflow feature is not assigned to the receiving switch, then the receiving switch will apply the name associated with its receiving VDN to the call, and this name can change according to the rules of VDN Override.

CAS (Centralized Attendant Service)

If a receiving switch is also the main location in a CAS arrangement, the receiving Look-Ahead Inteflow vector can contain a "route to" step with the Attendant Dial Access code (Encode 8) as the destination. This "route to" step (usually used as an ***alternate*** destination) will unconditionally deliver calls to the local CAS queue.

If a receiving switch is also a branch location in a CAS arrangement, the receiving Look-Ahead Interflow vector can contain a "route to" step with the Call to CAS Attendant dial access code (Encode 49) as the destination. This "route to" step (usually used as an ***alternate*** destination) will unconditionally deliver calls to the CAS queue at the main location.

Conference—Three-Party

At a receiving switch, the Look-Ahead Interflow feature and the 3-Party Conference feature are compatible. The answering voice terminal user at the receiving switch can normally conference a Look-Ahead Interflow call with a third party inside or outside the receiving switch.

Dial Access to Attendant

If the Dial Access to Attendant feature is enabled at a receiving switch, the receiving Look-Ahead Interflow vector can contain a "route to" step with the Attendant Dial Access code (Encode 8) as the destination. This "route to" step (usually used as an ***alternate*** destination) will unconditionally deliver calls to the local attendant queue.

DID (Direct Inward Dialing)

At a sending switch, incoming DID calls can terminate to a VDN with a vector assigned that contains commands for Look-Ahead Interflow. When this is done, these incoming DID calls will interflow normally (that is, according to the commands in the sending and receiving vectors).

At a receiving switch, interflowed calls from over ISDN—PRI trunk groups can enter the receiving switch on a DID basis (that is, using digit-oriented routing). To provide this capability, the ISDN SETUP message that is sent by the sending switch contains the digits of the destination VDN at the receiving switch.

DOD (Direct Outward Dialing)

From a sending (or tandeming) switch, calls can interflow over the public network using ISDN—PRI trunk groups assigned as DOD Trunk Types 17, 19, 22, 24, and 27. When this is done, these interflow calls will succeed if every public-network switch between the sending switch and the destination telephone number is ISDN—PRI-capable.

Display—Voice Terminal

With the exception of the "called party's name," an ACD agent at a receiving switch who answers a Look-Ahead Interflow call (with a display set) receives the normal display for incoming ACD calls. Some sample displays are shown in Table 78-C.

The first field identifies the calling party. If the call routes over DCS facilities, this field contains the calling party's name or number as provided in the DCS message. Otherwise, this field contains the calling party's name or telephone number if provided in the Look-Ahead SETUP message. If the Look-Ahead IE is blank, this field contains the name assigned to the incoming trunk group at the receiving switch.

The second field identifies the called party. This field contains the original or changed* called party's name as provided in the Look-Ahead SETUP message.

Also, if the Look-Ahead Interflow call routes over DCS facilities, the last character of the display shows the "reason for redirection" as provided in the DCS message.

* Within a local switch, the original called VDN's name can be changed by assiging VDN Override to a VDN in Procedure 031, Word 1.

TABLE 78-C. Look-Ahead Interflow Display Information

Type of Call	Display
ISDN call	a=R JONES to DETROIT CLAIMS
ISDN call	a=212-281-7733 to DETROIT CLAIMS
Default	a=OUTSIDE CALL to DETROIT CLAIMS
DCS call	a=RUTH A JONES to DETROIT CLAIMS b

DCS (Distributed Communications System)

Look-Ahead Interflow can divert calls over DCS facilities that are set up to use ISDN—PRI facilities within AAR/WCR patterns. To implement Look-Ahead Interflow over DCS facilities, an ETN configuration is required where 4- or 5-Digit Network Dialing or ENP converts the dialed 4- or 5-digit DCS extension to an RN(X) + XXX(X).

This type of DCS arrangement is required for routing Look-Ahead Interflow calls to a centralized AUDIX or Message Center split.

To prevent the loss of transparency for DCS calls over ISDN—PRI trunk groups, ISDN glare resolution does not allow the Network side to negotiate a channel for the User side after a glare condition. Instead, the Network side blocks the call and sends the User side a message to return reorder tone. If the User side is the sending switch in a Look-Ahead Interflow call, this switch, upon receiving the message, will retry the vector step (if the final effective step) at 2-second intervals. Otherwise, the sending switch will continue vector processing with the next sequential vector step.

ETN (Electronic Tandem Network)

Private-network ISDN—PRI tie trunks must be assigned as the ISDN Dynamic trunk type (Trunk Type 120) or as ETN Trunk Types 41, 42, 43, 46, or 47. In either case, an ETN configuration [with the Standard Networking field assigned (in Procedure 276) for each switch] is required so that the AAR/WCR feature can route Look-Ahead Interflow calls through the private network.

FRL (Facilities Restriction Level)

The Look-Ahead Interflow feature uses the FRL assigned to the VDN that the calling party originally dialed to divert incoming calls outside the R2 V4 System 85 or DEFINITY Generic 2.

If the VDN associated with the sending switch's vector does not have a high enough FRL to access an available preference in the AAR/ARS/WCR pattern determined by the destination digits of the "route to" step, the Look-Ahead Interflow software considers the "route to" step as having an invalid destination. If the "route to" step is the final effective step in the vector, the step is treated as a "stop" step. Otherwise, the sending switch continues vector processing with the next sequential step.

The AAR/ARS/WCR routing of a Look-Ahead Interflow "route to" step can be denied when Alternate FRLs are activated. After an attendant at the sending switch activates Alternate FRLs, the FRL of the VDN associated with the sending switch's vector changes to the assigned (in Procedure 286, Word 1) Alternate FRL. If the Alternate FRL is not high enough to access an available preference in the AAR/ARS/WCR pattern determined by the destination digits of the "route to" step, the "route to" step is considered to have an invalid destination.

FX (Foreign Exchange) Access

From a sending (or tandeming) switch, calls can interflow over the public network using ISDN—PRI trunk groups assigned as FX Trunk Type 22 or 24. Full Look-Ahead Interflow capabilities are only available if every public-network switch between the sending switch and the destination telephone number is ISDN—PRI-capable.

At a sending switch, incoming FX Access calls can terminate to a VDN with a vector assigned that contains commands for Look-Ahead Interflow. When this is done, these incoming FX Access calls will interflow normally (that is, according to the commands in the sending and receiving vectors).

ISDN—PRI (Primary Rate Interface)

For Look-Ahead Interflow calls (routed through ***either*** the private or public network) to succeed, ISDN—PRI connectivity is required from the sending switch to (and including) the receiving switch. When Look-Ahead Interflow is assigned to the Feature Group Class of Service (Procedure 276) at the sending (or tandeming) switch, interflow is always performed on a look-ahead basis when these ISDN facilities are available.

To increase the likelihood of "end-to-end" ISDN connectivity, Look-Ahead Interflow calls always route as "ISDN Preferred."

Intercept Treatment

For Look-Ahead Interflow Calls where the sending (or tandeming) switch routes the call to the receiving switch over ISDN—PRI facilities, no type of Intercept Treatment (for example, intercept tone, reorder tone, or recorded announcement) is given to the calling party. Instead of returning Intercept Treatment, the receiving switch rejects the call with a D-channel message, and the sending switch relies on its own vector to provide a suitable alternate treatment.

For Look-Ahead Interflow Calls where the sending (or tandeming) switch routes the call to the receiving switch over ***non-*** ISDN—PRI facilities, some form of Intercept Treatment may be returned to the original calling party. Without an ISDN—PRI facility, the receiving switch has no way of rejecting the call. Moreover, the final switch does not know the trunk type of the original incoming trunk.

IXC (Interexchange Carrier Access)

The Look-Ahead Interflow feature is compatible with the IXC access feature. The AAR and ARS subnetwork trunking functions can modify the contents of vector-group list items (used by "route-to" commands) so that Look-Ahead Interflow calls can route over IXC Access trunk groups.

Therefore, from a sending switch, calls can interflow over the public network using ISDN—PRI trunk groups assigned as IXC Access trunk groups. When this is done, these interflow calls will succeed if every public-network switch between the sending switch and the destination telephone number is ISDN—PRI-capable.

Beginning with DEFINITY Generic 2.2, an IXC code can be included as part of a vector-group list item. WCR digit-sending parameters are used to determine if an IXC code should be sent and what IXC code to send.

Main/Satellite/Tributary

The Look-Ahead Interflow feature requires ISDN—PRI facilities from the sending switch to the receiving switch. However, since Main / Satellite trunk groups cannot be used for ISDN trunking arrangements, the Look-Ahead Interflow feature cannot be used in a Main / Satellite / Tributary arrangement.

If a private network's switches are configured as a Main / Satellite arrangement prior to the implementation of Look-Ahead Interflow, these switches must be reconfigured as an ETN (Electronic Tandem Network) AAR / WCR routing. To retain the extension-number dialing capabilities (offered by Main / Satellite) within this new ETN configuration, Main / Satellite (70-series) trunk groups can reside in ***parallel*** to the ETN (40-series) trunk groups. Or, as an alternative, the Extension Number Portability feature can be used to emulate the desired Main / Satellite capabilities.

Malicious Call Trace

At a receiving switch, the Look-Ahead Interflow feature and the Malicious Call Trace feature are compatible. An attendant or an authorized voice terminal user at the receiving switch can activate Malicious Call Trace toward Look-Ahead Interflow calls in the normal manner. The controlling attendant can trace these calls to (and through) the sending switch in the normal manner. Also, as the controlling attendant executes the trace, the attendant will receive the 10-digit ISDN number as the source of the malicious call if this number was originally provided to the sending switch.

MEGACOM 800 Service Access

At a sending switch, incoming ISDN MEGACOM 800 Service calls can terminate to a VDN with an associated vector that contains commands for Look-Ahead Interflow. When this is done, these incoming 800 Service calls will interflow normally (that is, according to the commands in the sending and receiving vectors) and, if available, ANI (Automatic Number Identification) will be sent to the receiving switch.

MCS (Message Center Service)

Using the Look-Ahead Interflow feature, a "route to" step within a sending vector can divert calls ***from a local Message Center split*** to the VDN of an alternate destination outside the switch (usually within the private network).

Using the Look-Ahead Interflow feature, a "route to" step within a sending vector can divert (direct or directed) calls from a local destination ***to the VDN of a centralized Message Center split*** within the DCS subnetwork. [When a diverted call undergoes DCS routing the distant switch receives the reason for redirection, the identity of the originally called principal, and IMN (Integrated Message Notification) information in the DCS message.]

Messaging Server

Using the Look-Ahead Interflow feature, a "route to" step within a sending vector can divert calls ***from a local Messaging Server split*** to the VDN of an alternate destination outside the switch (usually within the private network).

Using the Look-Ahead Interflow feature, a "route to" step within a sending vector can divert (direct or redirected) calls from a local destination ***to the VDN of a centralized Messaging Server split*** within the DCS subnetwork. [When a diverted call undergoes DCS routing the distant switch receives the reason for redirection, the identity of the originally called principal, and IMN (Integrated Message Notification) information in the DCS message.]

Music-on-Hold Access

At a Look-Ahead Interflow receiving switch, the music interface that can be provided with a "wait" step is functionally independent of the system-wide Music-on-Hold feature.

Precedence Calling

The Look-Ahead Interflow feature is not compatible with the Precedence Calling feature. AUTOVON Access trunk groups are APLT trunk groups (with Trunk Types 12 to 15), but ISDN—PRI trunk groups cannot be assigned as an APLT trunk type. Also, the AAR/WCR feature does not route AUTOVON calls, but AAR/WCR is required to route Look-Ahead Interflow calls through the private network.

Remote Access

At a sending switch, a Remote Access user is allowed to dial a VDN after the sending switch returns second dial tone. When this is done, the Remote Access call will complete to the VDN with a vector assigned that may contain command(s) for Look-Ahead Interflow. In turn, these incoming Remote Access calls will interflow normally (that is, according to the commands in the sending and receiving vectors).

Restriction—Attendant Control of Voice Terminals

An attendant is not allowed to activate an Attendant Control of Voice Terminals restriction against a Look-Ahead Interflow VDN. When this is attempted, the switch returns intercept tone.

Restriction—Code Restriction

The Code Restriction feature does not prevent "route to" steps from routing calls outside the switch. "Route to" steps use AAR/ARS/WCR software, which does not make Code Restriction checks.

Restriction—Miscellaneous Trunk Restrictions

The Miscellaneous Trunk Restrictions feature denies preselected lines dial access to preselected trunk groups. Miscellaneous Trunk Restrictions do not limit preference selection within AAR/ARS/WCR patterns. Therefore, when Miscellaneous Trunk Restrictions and Look-Ahead Interflow are both assigned, the Miscellaneous Trunk Restrictions feature does not apply to Look-Ahead Interflow calls being routed with a "route to" step.

Restriction—Toll Restriction

The Toll Restriction feature does not prevent "route to" steps from routing calls outside the switch. "Route to" steps use AAR/ARS/WCR software, which does not make Toll Restriction checks.

Restriction—Voice Terminal Restrictions

Voice terminal restrictions ***do not apply*** to VDNs. For example, if Termination Restriction is assigned to Class of Service 1, and Class of Service 1 is assigned to VDN 7300, the restriction is ignored by the Call Vectoring feature. Calls are allowed to terminate to the vector.

Voice terminal restrictions do not prevent "route to" steps from routing calls to destinations outside the switch. For example, if Origination or Outward Restriction is assigned to a VDN's class of service, the assignment is ignored.

Route Advance

The Route Advance feature has no effect on the way the AAR/ARS/WCR features select preferences within patterns and, therefore, it has no effect on the routing of Look-Ahead Interflow calls. If a trunk group in a pattern is also the first trunk group in a Route Advance sequence, the AAR/ARS/WCR software ***ignores*** the alternate Route Advance trunk groups while selecting an available (and accessible) trunk group in the pattern.

Tenant Services

The Look-Ahead Interflow feature can be provided on a partitioned R2 V4 System 85 or DEFINITY Generic 2. When implemented, the Look-Ahead Interflow feature (and the supporting Call Vectoring feature) are provided for every extension partition in the switch.

At a sending (or tandeming) partitioned R2 V4 System 85 or DEFINITY Generic 2, the ARS/WCR partitioning provided by the Tenant Services feature (the ability to associate a routing designator with pattern identified by a ***different*** number) can be applied to Look-Ahead Interflow calls as they are diverted outside the R2 V4 System 85 DEFINITY Generic 2 to the public network.

At a receiving partitioned R2 V4 System 85 or DEFINITY Generic 2, the switch also observes partition boundaries when the ISDN—PRI routing uses a form of "digit-oriented" routing (Trunk Type 46, 47, or 120). As the receiving switch receives the digits from within the Look-Ahead D-channel message, the receiving switch makes the necessary partitioning checks (based on the VDN's association with an extension partition in Procedure 000, Word 4).

Touch-Tone Calling Senderized Operation

Since the Look-Ahead Interflow feature uses ISDN—PRI trunk facilities where the dialed digits are contained in the interflow SETUP message, Look-Ahead interflow calls ***do not*** rely on SN251 receiver circuits and SN252 sender circuits to outpulse and receive touch-tone digits.

Transfer

At a receiving switch, the Look-Ahead Interflow feature and the Transfer feature are compatible. The answering voice terminal user at the receiving switch can normally transfer a Look-Ahead Interflow call to a third party inside or outside (if Trunk-to-Trunk Transfer is assigned) the receiving switch.

WATS (Wide Area Telecommunications Service) Access

From a sending (or tandeming) switch, calls can interflow over the public network using ISDN—PRI trunk groups assigned as WATS Trunk Type 27. Full Look-Ahead Interflow capabilities are only available if every public-network switch between the sending switch and the destination telephone number is ISDN—PRI-capable.

At a sending switch, incoming 800 Service calls can terminate to a VDN with a vector assigned that contains commands for Look-Ahead Interflow. When this is done, these incoming 800 Service calls will interflow normally (that is, according to the commands in the sending and receiving vectors).

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, the WCR feature is required to route Look-Ahead Interflow calls. To provide private network routing for interflow calls within WCR, the "Standard Networking" field (Procedure 276, Word 1) must be assigned.

Dialing Plan

When the WCR feature is used to route Look-Ahead Interflow calls, the contents of a vector-group list item for a "route to" step must conform to the dial plan of the network. When System 85s, DEFINITY Generic 2 switches, and DIMENSION FP 8, Issue 3 switches are part of a private network, the dial plan can have one of the following forms:

- RNX (3-Digit Location Code) + XXXX (4-Digit Extension Number)
- RN (2-Digit Location Code) + XXXX (4-Digit Extension Number)
- RNX (3-Digit Location Code) + XXX (3-Digit Extension Number)
- RN (2-Digit Location Code) + XXX (3-Digit Extension Number).

Public Network Routing

When Look-Ahead Interflow calls are to be routed over public network facilities, the contents of a vector-group list item for a "route to" step must conform to the public network rules for DDD (Direct Distance Dialing). The DDD formats for the public network can have one of the following forms:

- NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- NPA (3-Digit Area Code) + NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- IXC (Interchange Carrier) + NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- International Telephone Number.

Prefixing for Look-Ahead Interflow Calls

Besides conforming to the dial plan for the network, the vector-group list items for Look-Ahead Interflow "route to" steps must be prefixed by the appropriate network DAC. For public network routing, a prefix digit (typically the digit 1) for toll calls or an international access code may also be required.

Besides conforming to the dialing plan for the network, a pattern must be translated for the destination digits of a "route to" step. When this is not done, the "route to" step is treated as having an invalid destination. If the "route to" step is the final effective step in the sending (or tandeming) vector, vector processing treats the step as a "stop" step. Otherwise, vector processing continues with the next sequential step in the vector.

Queuing

At a sending (or tandeming) switch, outgoing-trunk queuing does not apply to Look-Ahead Interflow calls. Instead, if every preference in the best-choice preference is busy, the Look-Ahead Interflow software will either retry routing the call at 2-second intervals (if final effective step) or continue vector processing with the next sequential vector step.

Bearer Capability Classification

For voice calls, the Bearer Capability Class of Service (BCCOS) is not a significant consideration. This is because voice calls are usually compatible with any carrier facility.

However, the WCR feature does check the BCCOS of calls that are diverted outside the switch by a "route to" step. Therefore, when applicable, the BCC of the outgoing (best-choice) preference must be compatible with the BCC in the local calling party's COS or the BCC assigned to the incoming trunk group.

Trunk Reservation Limits

The Trunk Reservation Limit (assigned in Procedure 103, Word 1) does not prevent Look-Ahead Interflow calls from accessing the first preference. Rather, assigning a Trunk Reservation Limit to the trunk group has the effect of reserving trunks in the preference to ensure the routing of Look-Ahead Interflow calls.

Conditional Routing

The Look-Ahead Interflow feature is compatible with Conditional Routing. For Look-Ahead Interflow calls, software increments the conditional routing count whenever a "route to" step diverts a call over a conditional route (for example, a route that contains a satellite link). Also, software will send the current value of the conditional routing count (if active for the selected trunk group) as the second TCM for Look-Ahead Interflow calls.

Time-of-Day Plan

The routing pattern selection for "route to" steps conforms to the currently active time-of-day plan. (This is the case whether the currently active time-of-day plan was invoked by an automatic plan change, clocked manual override, or manual override.)

Whenever a time-or-day plan is active where a pattern's best-choice preference results in the selection of a non-PRI trunk group, "route to" steps (if successful in diverting calls) will route the calls on a non-Look-Ahead basis.

Unauthorized Call Control

The routing of Look-Ahead Interflow "route to" steps can be blocked by unauthorized call control. Whenever a vector-group list item for the Look-Ahead Interflow feature contains a digit string that is marked for call control, vector processing at the sending switch either treats the "route to" step as a "stop" step (if the final effective step) or continues with the next sequential step in the vector.

When the digits of a "route to" destination are undefined in WCR or translate to the intercept pattern (VNI 0), the "route to" step is considered to have an invalid destination.

WCR Toll Restriction

Toll Restriction does not limit the routing of Look-Ahead Interflow "route to" steps to an answering destination. If toll restriction is assigned to a VDN's class of service, this assignment is ignored.

Tandem Processing

As part of the Look-Ahead Interflow SETUP message, an intervening (tandem) switch is always requested to route the interflow call on ISDN-Preferred basis. The tandeming switch attempts to select ISDN routes first during its route-selection process. If the tandeming switch cannot find an available ISDN route, a non-ISDN route is selected and an "Interworking" CALL PROGRESS message is returned to the sending switch. When routing is modified to a non-ISDN route, the call continues to route on a non-Look-Ahead basis.

Restricting Feature Use

At a receiving switch for Look-Ahead Interflow calls, diverted calls to VDNs are not restricted The following is a summary of these interactions:

- DID restriction is ignored (if the ISDN—PRI facilities have a DID trunk type).
- Incoming Attendant Control of Voice Terminals cannot be activated.
- Incoming Voice Terminal Restrictions are ignored.

At either a sending switch or a tandeming switch for Look-Ahead Interflow calls, the routing of "route to" steps to an external destination is controlled by the FRL of the VDN's class of service and by the BCC of the incoming trunk group or local calling party's COS. The following is a summary of these interactions:

- "Route to" steps cannot route a call to the private or public network unless the VDN's FRL (or Alternate FRL) is high enough to allow access to an available preference in the corresponding routing pattern.
- "Route to" steps cannot route a call over an ISDN—PRI facility unless the BCC of either the incoming trunk group or the local calling party's COS is compatible with the BCC of the outgoing ISDN—PRI facility.
- ARS/WCR Toll Restriction is ignored.
- Outgoing Attendant Control of Voice Terminal restrictions cannot be activated.
- Code Restrictions do not apply.
- Toll Restriction does not apply.
- Outgoing Voice Terminal Restrictions are ignored.

Hardware Requirements

Beyond the basic hardware chosen to implement other related features [for example, the ETN featuew, the ISDN—PRI feature, the regular ACD or the Call Vectoring feature, the Music-on-Hold Access feature, the CMS (Call Management System), or recorded announcements there is no additional hardware required to implement the Look-Ahead Interflow feature.

Feature Administration

Assignment of the Look-Ahead Interflow feature is on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

The following are the applicable administration procedures.

ADMNISTRATION PROCEDURES — LOOK-AHEAD INTERFLOW		
PROCEDURE	**WORD**	**PURPOSE**
100	3	Assigns Optional ISDN Info Inhibited restriction to an ISDN—PRI trunk group. This restriction must be removed (Field 8 set to 0) to enable a PRI trunk group for use with Look-Ahead Interflow.
276	—	Assigns Look-Ahead Interflow to the feature-group class of service

Notes:

Loudspeaker Paging Access

Description

The Loudspeaker Paging Access feature allows attendants and voice terminal users to make announcements over a voice paging system. This simplifies contacting individuals who are not generally at a specific location by reducing the need for a messenger or repeated dialing. Paging is also useful for announcements during safety or emergency situations.

Paging Zones

Announcements are directed to paging zones (specific areas). Them can be up to 18 paging zones. Announcements can be made over several paging zones at one time. This is called ***all-zones paging.*** All-zones paging makes the announcement over paging zones 1 through 5 by bridging them together.

Paging Zone Codes

Paging zones are accessed by dialing a paging zone code. Paging zone codes are numbered 1 through 18. The all-zones (zones 1 through 5) paging code is 0. If more than ten paging zones are used, a 2-digit code is used when dialing any paging zone. For example, the all-zones paging code "0" becomes "00" and the paging zone code of "4" becomes "04."

Loudspeaker Paging Access provides the following:

- Attendant Direct Access

  Allows an attendant to access paging by pressing and holding paging buttons. An all-zones paging button, as well as buttons for each paging zone, may be provided. Also priority paging buttons may be provided for all-zones and for each individual zone. When direct access is used for paging answer-back is not provided.

- Attendant and Voice Terminal Dial Access

  Allows an attendant or a voice terminal user to dial-access each zone individually or to dial-access all-zones when multizone paging is provided. Answer-back is provided when the paged party dial accesses the paging zones.

- Priority Paging

  Allows an attendant and/or voice terminal user to dial-access a paging zone and answer-back channel and preempt any voice terminal connected to that zone or channel. When priority paging is desired, the attendant or voice terminal user should dial "1" for the answer-back code number.

- Paging With the Transfer Feature

  Allows voice terminal users, that have the Transfer feature assigned, to access paging while on a 2-party call. An answer-back channel must be activated.

- Answer-Back

  Provides nine answer-back channels over which paged parties can be connected to calls.

- Music Option

  Provides music instead of ringback tone while waiting for an answer-back.

Feature History and Development

This feature was first available with System 85 in Release 1. The enhancements to this feature include:

- In Release 2, Version 2, automatic attendant recall was provided.
- In Release 2, Version 3, an option to have music instead of ringback tone while waiting for an answer-back was added. This option is a administered in Procedure 275, Word 1, Field 7.
- In Release 2, Version 4, an administrable recall button was provided. This enhancement can be retrofitted to the R2 V2 and R2 V3 software packages.
- In DEFINITY Generic 2, the Universal Module is added which provides hardware options with economies in both expense and space requirements.

User Operations

To Page a Party Using Attendant Direct Access:

1. Press and hold the all-zones button, the paging button the the appropriate zone, or the priority paging button. [PA lamp is lit PAGE lamp lights.]
2. Make the announcement over the loudspeaker system.
3. Release PAGE. [PAGE lamp goes out.]

To Page a Party using Attendant Dial Access

With no source party and no answer-back wanted:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights.]
2. Press **[START]** . [Dial tone]
3. Dial the paging trunk-group access code. [Second dial tone]

4. Dial the individual paging zone number or **[0]** for all zones.
5. Dial the answer-back channel number **[0]** . [Confirmation tone is heard, and the RING lamp lights.]
6. Make the announcement over the loudspeaker system.
7. Press **[RELEASE]** . [RING and ATND lamps go out, PA lamp lights, and paging equipment is released.]

With no source party and an answer-back wanted:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights.]
2. Press **[START]** . [Dial tone]
3. Dial the paging trunk-group access code. [Second dial tone]
4. Dial the individual paging zone number or **[0]** for all zones.
5. Dial an answer-back channel number "1" through "9." [Confirmation tone is heard, and the RING lamp lights.]
6. Make the announcement over the loudspeaker system. (Be sure to request the reply over the appropriate answer-back channel.)
7. Press **[HOLD]** . [ATND lamp goes out. PA and HOLD lamps light.]
8. Wait for answer-back
9. When the HOLD lamp flashes, press the appropriate loop button to answer the answer-back call. [RING and HOLD lamps go out, and ATND lamp lights.]
10. Talk with the paged party.
11. When finished with the paged party, press **[RELEASE]** . [Attendant and paged party are disconnected, ATND lamp goes out, PA lamp lights, and paging equipment is released.]

With a source party and no answer-back wanted:

1. Be sure a connection is made with a source party. [ATND lamp is lit, and ICI (Incoming Call Identification) displays the extension number.]
2. Press **[START]** . [Source party is split away from the attendant, the SPLIT lamp lights, and the attendant hears dial tone.]
3. Dial the paging trunk-group access code. [Second dial tone]
4. Dial the individual paging zone number or **[0]** for all zones.
5. Dial the answer-back code number **[0]** . [Confirmation tone is heard, and the RING lamp lights.]
6. Make the announcement over the loudspeaker system.
7. Press **[CANC]** . [Attendant is reconnected to the source party, RING lamp goes out, and paging equipment is released.]

8. Press **[RELEASE]** . [Attendant is disconnected from the source party, ATND lamp goes out, and PA lamp lights.]

With a source party and an answer-back wanted:

1. Be sure a connection is made with a source party. [ATND lamp is lit, and ICI displays the extension number.]
2. Press **[START]** . [Source party is split away from the attendant, the SPLIT lamp lights, and the attendant hears dial tone.]
3. Dial the paging trunk-group access code. [Second dial tone]
4. Dial the individual paging zone number or **[0]** for all zones.
5. Dial an answer-back channel number "1" through "9." [Confirmation tone is heard, and the RING lamp light.]
6. Make the announcement over the loudspeaker system. (Be sure to request the reply over the appropriate answer-back channel.)
7. Press **[RELEASE].** [Attendant is disconnected from the source party. The source party hears ringback tone or music until the paged party answers back. ATND and SPLIT lamps go out, and the PA lamp lights.]
8. Paged party answers back [Both parties receive confirmation tone, and RING lamp goes out.]

To Page a Party Using Voice Terminal Dial Access

Without an answer-back wanted:

1. Go off-hook. [Dial tone]
2. Dial the paging trunk-group access code. [Second dial tone]
3. Dial the individual paging zone number or **[0]** for all zones.
4. Dial the answer-back code number **[0]** . [Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
5. Make the announcement over the loudspeaker system.
6. Go on-hook. [Paging lamps on the attendant consoles go out.]

With an answer-back wanted (priority):

1. Go off-hook. [Dial tone]
2. Dial the paging trunk-group access code. [Second dial tone]
3. Dial the individual paging zone number or **[0]** for all zones.
4. Dial the answer-back code number **[1]** . [Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
5. Make the announcement over the loudspeaker system. (Be sure to request the reply over answer-back channel 1.)

6. Press **[RECALL]** or momentarily press the switchhook. [Ringback tone or music is heard, and the paging lamps on the attendant consoles go out.]
7. The paged party answers the page. [When paging and paged parties are connected, ringback tone or music is removed, and confirmation tone is heard by both parties. The answer-back channel is released.]

Without an answer-back wanted (priority):

1. Go off-hook. [Dial tone]
2. Dial the paging trunk-group access code. [Second dial tone]
3. Dial the individual paging zone number or **[0]** for all zones.
4. Dial the answer-back code number **[1]** . [Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
5. Make the announcement over the loudspeaker system.
6. Go on-hook. [The paging equipment is released. Paging lamps on the attendant consoles go out.]

With a 2-party connection at a single-line voice terminal and no answer-back wanted (priority):

1. Press **[RECALL]** or momentarily press the switchhook. [The second party is put on soft hold. Recall dial tone is heard.]
2. Dial the paging trunk-group access code. [Second dial tone]
3. Dial the individual paging zone number or **[0]** for all zones.
4. Dial the answer-back channel number **[1]** . [Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
5. Make the announcement over the loudspeaker system.
6. Press **[RECALL]** or momentarily press the switchhook. [The paging circuit is disconnected, ringback tone is heard, and the paging lamps on the attendant consoles go out.]
7. Press **[RECALL]** or momentarily press the switchhook. [The paging party is reconnected with the party on soft hold. The answer-back channel is released.]

With a 2-party connection at a multiappearance voice terminal and no answer-back wanted (priority):

1. Press **[HOLD]** . [The second party is put on hold, and the green status lamp flashes.]
2. Press an idle line appearance. [Dial tone]
3. Dial the paging trunk-group access code. [Second dial tone]
4. Dial the individual paging zone number or **[0]** for all zones.

5. Dial the answer-back channel number **[1]** . [Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
6. Make the announcement over the loudspeaker system.
7. Press the line appearance of the party on hold. [The two parties are reconnected. The green status lamp lights steady, the paging equipment is released, and the paging lamps on the attendant console go out.]

With a 2-party connection on a single-line voice terminal and no answer-back wanted:

1. Press **[RECALL]** or momentarily press the switchhook. [The second party is put on soft hold. Recall dial tone is heard.]
2. Dial the paging trunk-group access code. [Second dial tone]
3. Dial the individual paging zone number or **[0]** for all zones.
4. Dial the answer-back code **[0]** . [Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
5. Make the announcement over the loudspeaker system.
6. To reconnect to the party on soft hold, press **[RECALL]** or momentarily press the switchhook. The paging equipment is released, and the paging lamps on the attendant consoles go out.]

With a 2-party connection on a multiappearance voice terminal and no answer-back wanted:

1. Press **[HOLD]** . [The second party is put on hold, and the green status lamp flashes.]
2. Press an idle line appearance. [Dial tone]
3. Dial the paging trunk-group access code. [Second dial tone]
4. Dial the individual paging zone number or **[0]** for all zones.
5. Dial the answer-back code **[0]** . [Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
6. Make the announcement over the loudspeaker system.
7. To reconnect to the party on soft hold, press the line appearance of the party on hold. [The two parties are reconnected. The green status lamp lights steady, the paging equipment is released, and the paging lamps on the attendant console go out.]

With a 2-party connection at a single-line voice terminal and an answer-back wanted:

1. Press **[RECALL]** or momentarily press the switchhook. [The other party is put on soft hold. Recall dial tone is heard.]

2. Dial the paging trunk-group access code. [Second dial tone]
3. Dial the individual paging zone number or **[0]** for all zones.
4. Dial the answer-back code (1 for priority, from 2 to 9 for another channel). [Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
5. Make the announcement over the loudspeaker system. (Be sure to request the reply over the appropriate answer-back channel.)
6. Press **[RECALL]** or momentarily press the switchhook. [The paging equipment is released, and paging party is connected to the answer-back channel. Ringback tone or music is heard, and the paging lamps on the attendant consoles go out.]
7. The paged party answers the page. [When the paging and paged parties are connected, ringback tone or music is removed, the answer-back channel is released, and confirmation tone is heard by both parties. Held party gets ringback tone.]
8. To reconnect to the held party, press **[RECALL]** . [Paged party is disconnected, and ringback tone is removed.]

With a 2-party connection at a multiappearance voice terminal and an answer-back wanted:

1. Press **[HOLD]** . [The second party is put on hold. The green status lamps flashes.]
2. Press an idle line appearance. [Dial tone]
3. Dial the paging trunk-group access code. [Second dial tone]
4. Dial the individual paging zone number or **[0]** for all zones.
5. Dial the answer-back code (1 for priority, from 2 to 9 for another channel). [Confirmation tone is heard, and the appropriate paging lamps on the attendant consoles light.]
6. Make the announcement over the loudspeaker system. (Be sure to request the reply over the appropriate answer-back channel.)
7. Press **[RECALL]** . [Ringback tone is heard, and the paging lamps on the attendant console go out. The paging equipment is released, and the paging party is connected to the answer-back channel.]
8. The paged party answers the page. [When the paging and paged parties are connected, ringback tone is removed, the answer-back channel is released, and confirmation tone is head by both parties.]

With a 2-party connection and the answer-back transferred to the second party at a single line voice terminal:

1. Press **[RECALL]** or momentarily press the switchhook. [The second party is put on soft hold. Recall dial tone is heard.]
2. Dial the paging trunk-group access code. [Second dial tone]

3. Dial the individual paging zone number or **[0]** for all zones.
4. Dial the answer-back code (1 for priority, from 2 to 9 for another channel). Confirmation tone is heard. The appropriate paging lamps on the attendant consoles light.]
5. Make the announcement over the loudspeaker system. (Be sure to request the reply over the appropriate answer-back channel.)
6. Go on-hook. [The second party is now connected to the answer-back channel, and the paging lamps on the attendant consoles go out.]
7. The paged party answers the page. [When the second party and paged parties are connected, ringback tone is removed, the answer-back channel is released, and confirmation tone is heard by both parties.]

With a 2-party connection and the answer-back transferred to the second party at a multiappearance voice terminal:

1. Press **[TRANSFER].** [The second party is put on soft hold. Recall dial tone is heard.]
2. Dial the paging trunk-group access code. [Second dial tone]
3. Dial the individual paging zone number or **[0]** for all zones.
4. Dial the answer-back code (1 for priority, from 2 to 9 for another channel). [Confirmation tone is head. The appropriate paging lamps on the attendant consoles light.]
5. Make the announcement over the loudspeaker system. (Be sure to request the reply over the appropriate answer-back channel.)
6. Press **[RECALL].**
7. Press **[TRANSFER]** or **[CONFERENCE].** [The second party is now connected to the answer-back channel, and the paging lamps on the attendant consoles go out. The green status light goes out.]
8. The paged party answers the page. [When the second party and paged parties are connected, ringback tone is removed, the answer-back channel is released, and confirmation tone is heard by both parties.]

To Answer a Page:

1. GO off-hook [Dial tone]
2. Dial the answer-back access code.
3. Dial the answer-back channel number. [Confirmation tone is heard by both the paging and paged parties. The answer-back channel is released. Paging lamps on the attendant consoles go out.]

Considerations

Administrable Recall Buttons

Some multiappearance voice terminals do not have a fixed RECALL button. If RECALL buttons are needed for these voice terminals, they can be assigned to the terminals using Procedure 054, Word 1.

Answer-Back

If a paging party wants the paged party to answer-back, the paging party should always request an answer-back and announce the answer-back channel number during the announcement.

Answer-Back Channels

Nine answer-back channels are available. These are the same answer-back channels used with the Call Park feature. These answer-back channels are shared by both features.

Busy Tone

Busy tone is heard when a specific zone or all-zone number is dialed and it is busy, when a terminal user attempts to preempt an attendant doing a priority page, or when an answer-back channel is not available.

Intercept Tone

Intercept tone is heard when a user dials an invalid answer-back access code or does not dial the channel number within 10 seconds after dialing the page zone number. Intercept tone is heard by the attendant when the attendant attempts to answer a page.

Outgoing Trunk Calls Waiting for Answer-Back

Beginning with R2 V2, outgoing trunk calls (without Disconnect Supervision assigned in Procedure 101, Word 1) that are waiting for answer-back are automatically disconnected after two minutes.

Paging Permission

Paging over the loudspeaker system is limited to paging from a voice terminal, an attendant console, or via tie trunk access. Callers from the public network (Direct Inward Dialing or Remote Access) cannot use the Loudspeaker Paging feature.

Paging Zone Lamps

When a specific zone is accessed, the lamps adjacent to that zone button and the all-zone button light. When all zones are accessed, the lamps adjacent to the all-zone button and every specific zone button light.

Preempting Page

An attendant using a paging zone and/or a priority paging channel cannot be preempted.

A voice terminal user using a priority paging channel can be preempted by another user using the same priority paging channel. If a preempted party did not request answer-back, it is dropped (nothing heard). If the preempted party requested answer-back, ringback tone is heard.

Reorder Tone

Reorder tone is heard when no intercom trunk is available.

Sharing Equipment

Code Calling Access and Loudspeaker Paging Access can share common equipment depending on the versions and arrangements used.

Code Calling Access — Universal

When Code Calling Access — Universal is used all equipment used by both features is shared.

Code Calling Access — Traditional

When Code Calling Access — Traditional is used, the amplifier and speaker equipment serving common paging zones can be shared.

When Code Calling Access — Traditional is used, a separate 89A Control Unit is used for Code Calling Access. This separate 89A Control Unit should be connected, via the seizure indication (CBS1 & 2) and busy-out input (COS1 & 2) leads, to the 89A Control Unit(s) used for Loudspeaker Paging Access in the common paging zones. This will prevent either feature from accessing shared equipment while it is in use by the other feature. This arrangement is shown in Figure 79-1.

This precaution is needed only when Code Calling Access — Traditional is used. In all other cases switch software can detect and prevent conflicting demands for shared equipment.

Trunk Calls Waiting for Answer-Back

Beginning with R2 V2, incoming trunk calls and outgoing trunk calls (with Disconnect Supervision assigned in Procedure 101, Word 1) that are waiting for answer-back will automatically recall an attendant after two minutes. At this time, the attendant can appropriately handle the call.

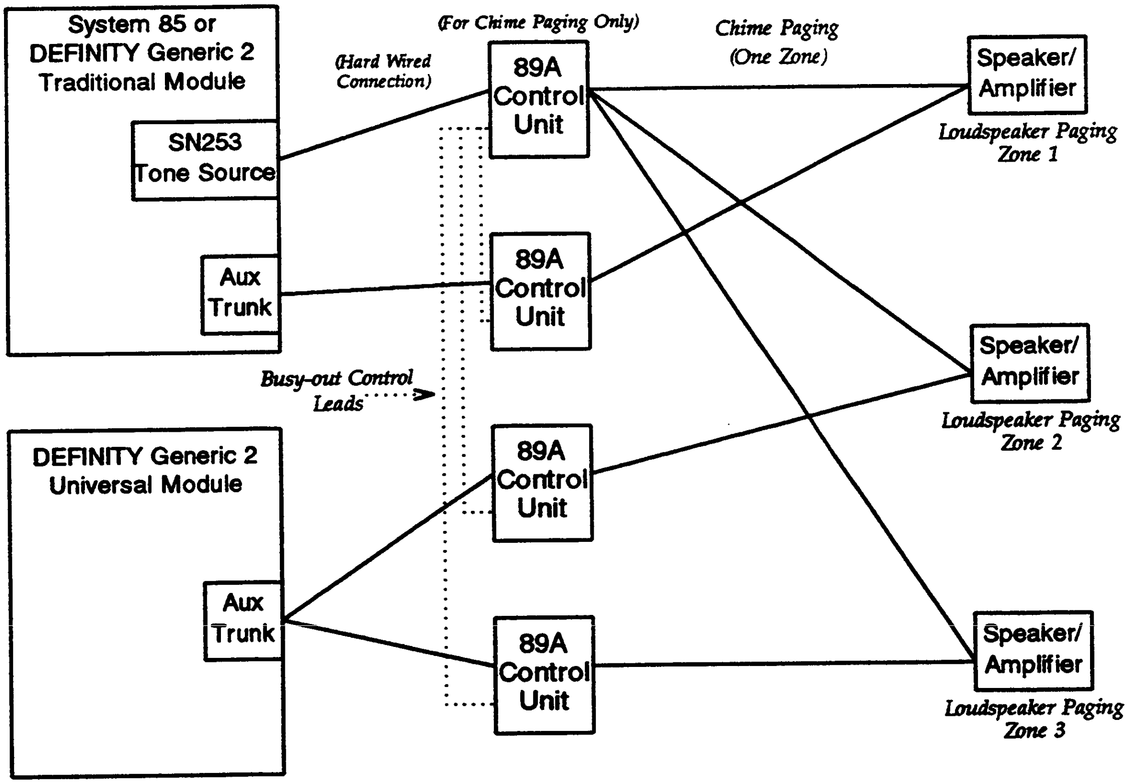

Figure 79-1. Lockout Arrangement For Shared Equipment With 89A Control Units

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

A call is not allowed to wait (via Attendant Call Waiting) on a line that has accessed Loudspeaker Paging (making a page or waiting for answer-back).

Busy Verification of Lines

Busy verification is denied on a voice terminal line that has accessed Loudspeaker Paging. Attempts to busy verify a voice terminal line in this state results in the attendant hearing reorder tone.

CDR (Call Detail Recording)

An attendant with an incoming trunk call can dial the CDR access code and account code and then use Loudspeaker Paging. The call record shows the call not the page.

Call Park

The Loudspeaker Paging Access feature and the Call Park feature are closely related. Administering Loudspeaker Paging, in effect, also enables Call Park. A paging zone that is not assigned for Loudspeaker Paging is assigned for use by Call Park. The paging zone assigned to Call Park requires an auxiliary trunk circuit to prevent alarms; however, it is not necessary to connect a paging amplifier to Call Park auxiliary trunk circuits. Both Call Park and Loudspeaker Paging Access share the same nine answer-back channels.

Call Waiting

A call is not allowed to wait (via Call Waiting) on a line that has accessed Loudspeaker Paging (making a page or waiting for answer-back).

Centralized Attendant Service

The Centralized Attendant Service attendant has only 10 seconds to make a page and then the paging circuit releases. This prevents the release link trunks associated with the attendant from being tied up.

Code Calling Access

Code Calling Access — Traditional

Code Calling Access and Loudspeaker Paging features may use either common amplifiers and speakers or separate amplifiers and speakers. In either case, care should be taken to insure that both features are not being used at the same time over the same equipment or in the same paging zones. Simultaneous use of both features produces interference between the two audible signals.

To prevent this interference, the lockout option on the 89A control units can be used. This arrangement is shown in Figure 79-1. The seizure indication (CBS1 & 2) and busy-out input (COS1 & 2) leads on the 89A control units serving common paging zones (whether common amplifiers and speakers are used or not) should be connected. This will prevent either feature from accessing amplifier and speaker equipment for common paging zones at the same time.

Code Calling Access — Universal

With the Code Calling Access — Universal and the Loudspeaker Paging Access features, all equipment is shared. This arrangement is shown in Figure 79-2.

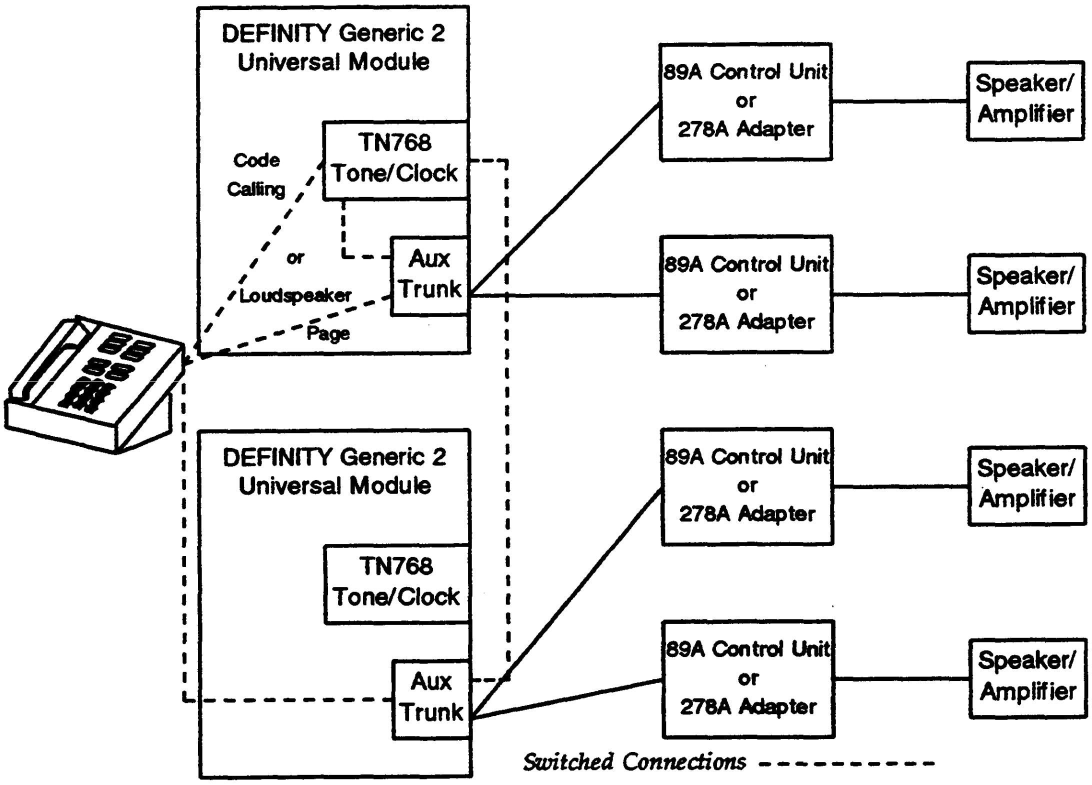

Figure 79-2. Shared Equipment With Universal Modules

With Code Calling Access — Universal, the code calling (chime) signal is sent to the amplifier/speaker equipment through a time slot (software connection). With this arrangement it is not possible for the Code Calling Access feature and the Loudspeaker Paging Access feature to access the same station (amplifier/speaker equipment) at the same time. If either feature attempts to use a paging zone which is already in use, the caller will receive busy tone.

Conference—Attendant Five Party

When establishing a Conference—Attendant Five Party call, an attendant cannot connect the conference to the paging loudspeakers

Conference—Attendant Six Party

When establishing a Conference—Attendant Six Party call, an attendant cannot connect the conference connection to the paging loudspeaker.

Music-on-Hold Access

When Music-on-Hold is implemented, music can be provided for a call on hold in an answer-back channel.

Override

Override is denied to a voice terminal line connected to Loudspeaker Paging.

Priority Calling

A call is not allowed to wait (via Priority Calling) on a line that has accessed Loudspeaker Paging (making a page or waiting for answer-back).

Remote Access

The Remote Access feature is not compatible with the Loudspeaker Paging Access feature. Incoming calls over Remote Access trunks are blocked from access to Loudspeaker Paging equipment.

Tenant Services

The paging zones for the Loudspeaker Paging Access feature are not partitioned. By default, the provided zones are equally accessible to attendants in any attendant partition and to voice terminal users in any extension partition.

Voice terminal access to the Loudspeaker Paging Access feature can be limited or enhanced in the extension class of service. To limit voice terminal access, assign a Miscellaneous Trunk Restrictions group containing the Loudspeaker Paging trunk group to an extension class of service in Procedure 010, Word 3. To enhance voice terminal access, priority paging can be assigned to an extension class of service in Procedure 010, Word 1.

Answer-back channels for the Loudspeaker Paging Access feature are not partitioned. A page can be answered by dialing the answer-back access code from any voice terminal in the switch.

Transfer

A single-line voice terminal user in a 2-party talking connection cannot access Loudspeaker Paging using the Transfer feature unless an answer-back channel is provided.

Trunk Verification—Voice Terminal

The testing capability of the Trunk Verification—Voice Terminal feature does not include the Loudspeaker Paging trunk.

Restricting Feature Use

Access to the Loudspeaker Paging Access feature can be restricted from a voice terminal line when miscellaneous trunk restrictions are assigned to the voice terminal line via the extension class of service.

Hardware Requirements

The Loudspeaker Paging feature requires the following additional or special hardware.

For Traditional Modules:

- An SN231 auxiliary trunk circuit for each paging zone (four circuits per circuit pack)

For Universal Modules:

- A TN763C auxiliary trunk circuit for each paging zone (four circuits per circuit pack)

Regardless of the Module Type:

- Coupling Arrangement and Power supply

  Two arrangements are possible, depending on the overall needs of the system *(see* Interaction with Code Calling Access feature):

  - An 89A control unit and 2012D power transformer

    This arrangement can be used for any configuration of the Loudspeaker Paging and Code Calling Access features. This arrangement is required when Loudspeaker Paging and Traditional Code Calling Access share the same equipment.

    or

  - A 278A Adapter and 24 V Power Supply

    This arrangement can be used when the Loudspeaker Paging Access does not share equipment with the Traditional Code Calling Access feature. The 278A Adapter requires a 24 Volt power source which can be provided using a D18132 Parts Kit (converts -48 V to -24 V).

- An amplifier
- Speaker(s).

Figure 79-3 shows typical setups of hardware for a paging system using either a Traditional module or a Universal module.

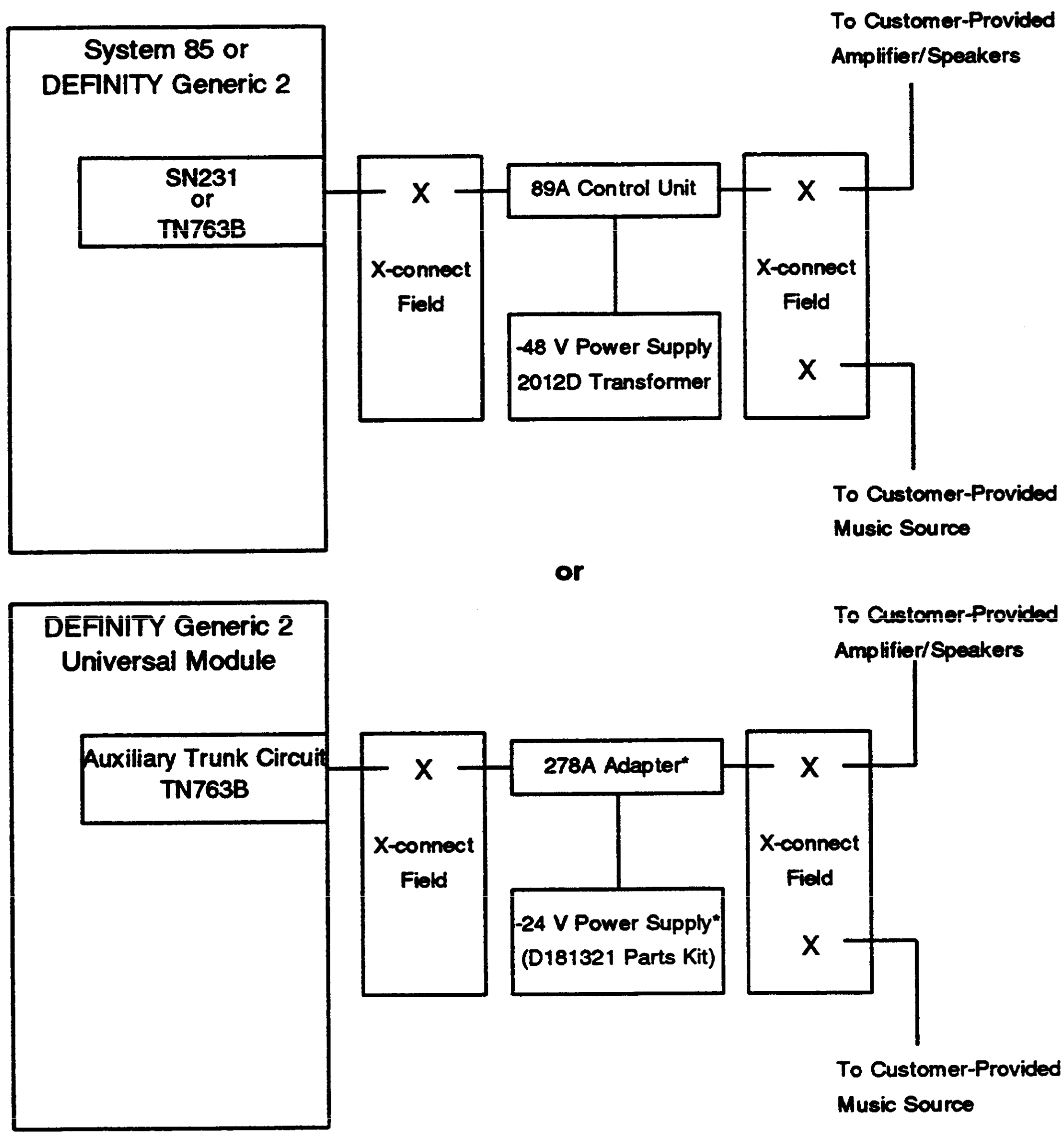

Figure 79-3. Hardware Setup for Loudspeaker Paging

Feature Administration

Assignment of the Loudspeaker Paging Access feature is on a per-system class of service and a per-extension class of service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal), or the TCM (Terminal Change Management) feature. On

* The Universal Module can also use the 89A Control Unit with -48 Volt Power Supply, however, the Traditional Module cannot use the 278A Adapter.

Generic 2 switches, this feature is assigned using the DEFINITY Manager II. This feature can also be administered using the Manager IV.

Feedback can occur when all-zones paging is active in three or more zones. To eliminate this problem, assign the trunk circuits serving zones 1 through 5 to their own trunk group, then access Procedure 101 Word 1 field 13 and set the pad group to "6" for that trunk group. Trunk circuits supporting zones 6 through 18 should be assigned to a separate trunk group — do ***not*** use Procedure 101 Word 1 field 13 on this separate trunk group.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — LOUDSPEAKER PAGING ACCESS

PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the class of service to an extension number.	Yes
010	1	Assigns priority paging to an extension class of service.	Yes
010	3	Assigns a Miscellaneous Trunk Restrictions group to an extension class of service.	Yes
054	1	Assigns the RECALL button to a multiappearance voice terminal without a fixed RECALL button. The applicable encode is: 27 Recall.	Yes
100	1	Assigns the trunk-group dial access code and the trunk type to the trunk group number. The applicable trunk-type encode is as follows: 54 Loudspeaker paging interface.	No
101	1	Specifies a new pad group. Assign only when all-zones paging will be used in three or more zones.	
102	1	Assigns the paging trunk group to a trunk restriction group.	Yes
150	1	Assigns an equipment location to a trunk group and to give a paging zone assignment to the trunk circuit.	No
203	1	Administers paging buttons on the attendant console.	No
275	1	Assigns Loudspeaker Paging to the system class of service and allows the option to have music-on-hold instead of ringback tone (Field 7).	Yes

ADMINISTRATION PROCEDURES — LOUDSPEAKER PAGING ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
350	1	Assigns the first digit of the dial access code (if required).	No
350	2	Assigns the answer-back dial access code. The applicable encode is: 17 Paging answer-back.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — LOUDSPEAKER PAGING ACCESS	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns priority paging to an extension class of service and assigns a Miscellaneous Trunk Restrictions group to the extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Main/Satellite/Tributary

Description

A Main / Satellite / Tributary configuration (see Figure 80-1) is an independent private network configuration that can also function as an Electronic Tandem Network access arrangement. For a Main / Satellite configuration, attendant positions and public network trunk facilities are concentrated at the main, and calls to or from satellite locations pass through the main. To a caller outside the Main / Satellite complex, the system appears to be a single switch with one listed directory number. A tributary location is similar to a satellite location with the following exceptions (1) a tributary has one or more attendant positions, and (2) a tributary has its own listed directory number.

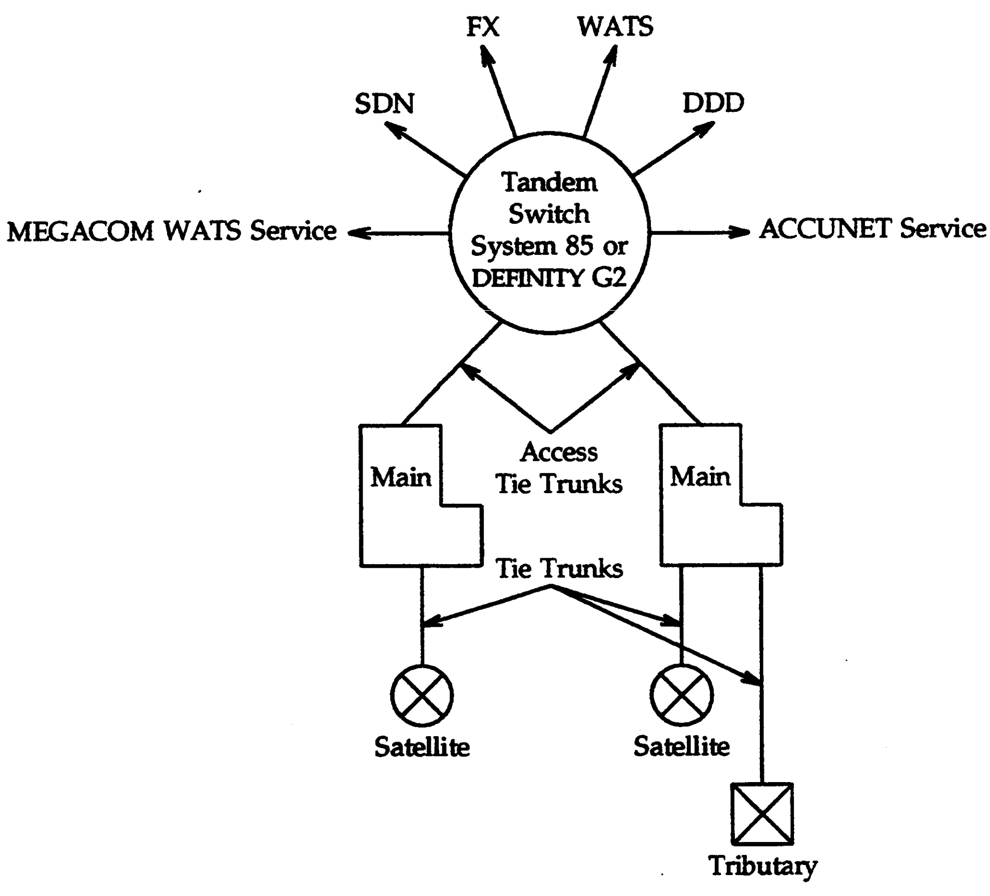

Figure 80-1. Main / Satellite and Main / Tributary Configuration

A small business can start with a single Main / Satellite or Main / Tributary complex and add trunk and switching facilities as the business grows. Figure 80-2 illustrates a Main / Satellite configuration in which tie trunks connect the main locations within an urban area and the public network carries intercity call traffic. This arrangement favors a medium-sized organization or one that has small isolated locations where intercity traffic volume is not heavy enough to justify the cost of tie trunks. In a partially connected network such as this, the Main/Satellite features are only available between switches connected by tie trunks.

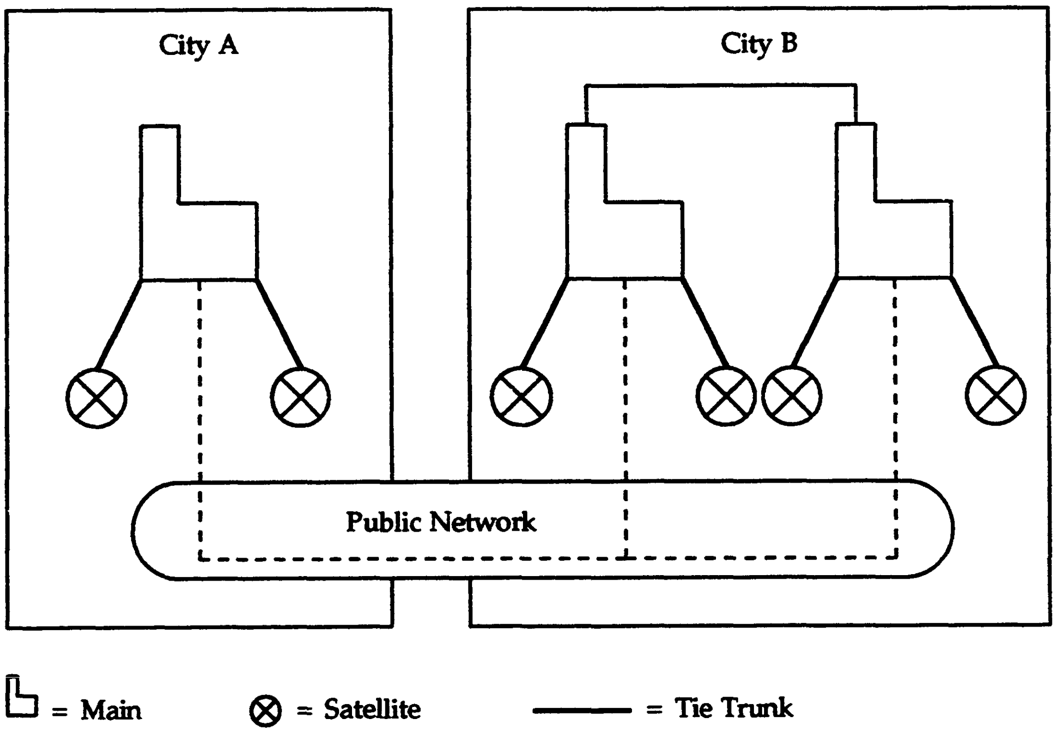

Figure 80-2. Partially Connected Main/Satellite Configuration

The following are attributes of the Main/Satellite feature.

Indialing Through Main

Incoming calls destined for a satellite location route through the main. This routing scheme reduces the number and types of trunks terminating on the satellite location.

Call Transfer

Voice terminal users within the Main/Satellite complex can transfer a call to a terminal on the same switch or any other switch in the complex. The dialing procedure, including the number of digits required to transfer a call, is consistent throughout the complex.

Uniform Numbering

A uniform numbering plan is the foundation for the Main/Satellite feature. Each switch in the complex has a unique group of extension numbers. This allows switches to distinguish between local calls and calls to other switches in the complex. The same digits are used to dial any extension on any switch in the network. Extension number steering is used to select trunks for extensions on remote switches.

Extension Number Steering

The switches of a Main / Satellite complex can use extension number steering to route calls within the complex. The first digit (or any number of digits) in the extension number steers the call to the appropriate switch, which then routes the call to the voice terminal (station). One factor that determines the amount of digits in an extension number that a complex uses for extension number steering is the number of switches in the complex. Any complex with more than ten switches must use multidigit steering.

In a Main / Satellite complex with extension number steering, extension numbers are defined as either normal or special. Normal extension numbers are those assigned to the local switch. Special extension numbers are those assigned to other switches in the complex. Any extension number that uses extension number steering is a special extension number. When callers dial a special extension number, the switch converts it to a trunk or feature access code (depending on how the extension number is administered) and processes the call accordingly.

DAC (Dial Access Code) Steering and ETA (Extended Trunk Access)

In a Main / Satellite complex (using an example with single-digit steering), a satellite location could be assigned all extension numbers beginning with the digit "3" (from 3000 to 3999). At the main, DAC steering can be used whereby the digit "3" is assigned as the dial access code for the Main / Satellite trunk group connecting to that satellite.

In turn, the satellite location can use either a similar steering arrangement or ETA (Extended Trunk Access) to route calls to the main. If ETA trunk groups are assigned at a satellite location, calls to extension numbers that are not assigned to that satellite are sent to the main for interpretation and subsequent routing.

Digit Inference and Deletion

System 85 and DEFINITY Generic 2 use digit inference and digit deletion. This two part process shortens call-processing time on interswitch calls. For a Main / Satellite complex, all extension numbers at a satellite location usually have the same leading digits (one or more digits depending on the number of digits used for steering). For example, all of the extension numbers at satellite A begin with the digits "38," all of the extension numbers at satellite B begin with the digits "42," and so on. On a call from the main to satellite A, the main deletes the digits "3" and "8" and sends the remaining digits. The satellite infers, from the trunk group on which the call arrives, that the digits "3" and "8" were deleted. Therefore, before processing the call, the satellite replaces the deleted digits.

Feature History and Development

The Main / Satellite / Tributary feature was first available for System 85 in Release 1. In Release 2, Version 3, this feature was enhanced to allow the attendant in a Main / Satellite configuration to use DXS (Direct Extension Selection) to reach stations on the satellite switch when a 4-digit dialing plan is used.

User Operations

Call routing within the network is controlled by software. No special user operations are required for this feature.

Considerations

Extension Numbering at the Main Location

At a main location, the numbering plan (Procedure 350, Word 1) and the extension-number blocks (Procedure 354, Word 1) ***must contain*** assignments for every extension in the Main / Satellite arrangement. Based on its complete extension-numbering information, the main can route Main / Satellite calls over Main / Satellite trunk groups according to the local assignments in Procedure 354, Word 2.

Extension Numbering at a Satellite Location

At a satellite location, the numbering plan (Procedure 350, Word 1) and the extension-number blocks (procedure 354, Word 1) ***can contain*** assignments for every extension in the Main / Satellite arrangement. When these assignments are made, the satellite routes outgoing Main / Satellite calls to the main location or to another satellite location according to the local assignments in Procedure 354, Word 2.

At a satellite location, ETA (Procedure 104, Word 1) can serve as an alternative to complete extension numbering for the Main / Satellite arrangement. When ETA is assigned to a satellite location, only the satellite's own extensions need be explicitly assigned in Procedures 350, Word 1 and 354, Word 1. (Dialed numbers that the satellite does not otherwise know how to handle are outpulsed to the main over the designated trunk group.)

Dial Access Restriction and Main/Satellite Trunk Groups

The Dial Access Restriction field (in Procedure 100, Word 1) should not be assigned for Main / Satellite trunk groups. This restriction interferes with the extension number steering assignments (in Procedure 354, Word 2) that route calls through the Main / Satellite configuration. Whenever an extension number steers to the dial access code of a dial-access restricted Main / Satellite trunk group, the switch denies further routing of the call. (Instead of the dial access restriction, use the Miscellaneous Trunk Restrictions feature to restrict dial access to Main / Satellite trunk groups.)

Trunk Queuing

Calls should not be permitted to queue on Main / Satellite trunk groups. There should be enough of these trunks so that queuing is not necessary.

Switch Applications

The following switches can function as a Main:

- DEFINITY Generics 1 and 2
- System 85
- System 75, Release 1, Version 2 and later
- DIMENSION Systems.

Any release and version of DIMENSION System, System 75, System 85 and DEFINITY switch can function as a satellite or tributary.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Features Limited to Local Switch

The following features, when provided, can only be used on the switch (main or satellite) where a call originates (not between switches). For example, a station assigned to satellite "A" can forward calls to another station on satellite "A,"' but it cannot forward calls to a station assigned to satellite "B."

NOTE: The use of a Main/Satellite arrangement in a DCS (Distributed Communications System) alters some of these specific interactions. See the DCS feature for these exceptions.

Station Features

- Automatic Callback
- Bridged Call
- Call Coverage
- Call Forwarding
- Call Pickup
- Call Waiting
- Code Restriction
- Distinctive Ringing
- Hunting
- Intercom
- Loudspeaker Paging
- Manual Signaling

- Message Center
- Override
- Personal Central Office Line
- Priority Calling
- Ringing Transfer
- Station Message Detail Recording
- Terminal Busy Indication
- Toll Restriction
- Voice Terminal Restrictions
  - Inward
  - Origination
  - Outward
  - Termination.

Attendant Features

- Attendant Display
  - Calling Number Display to Attendant
  - Class of Service Display to Attendant
  - Incoming Call Identification
  - Trunk Identification
- Attendant Call Waiting
- Attendant Control of Trunk Group Access
- Attendant Control of Voice Terminals
  - Outward Restriction
  - Terminal-to-Terminal Restriction
  - Termination Restriction
  - Total Restriction
- Attendant Direct Extension Selection With Busy Lamp Field*
- Attendant Direct Trunk Group Selection

* The limitation of Attendant DXS to local stations is removed in Release 2, Version 3, and later switches if a 4-digit dialing plan is used.

- Busy Verification of Lines
- Intercept Treatment-Attendant
- Multiple Listed Directory Numbers
- Preselected Call Routing
- Serial Calls
- Timed Recall on Outgoing Calls
- Trunk Group Busy / Warning Indicators
- Trunk Verification

Attendant Direct Extension Selection With Busy Lamp Field

An attendant for a Main / Satellite complex with extension number steering can use this feature to place a call to a distant switch. However, the busy lamp associated with the extension number does not light.

AAR (Automatic Alternate Routing)

Although, not required, the AAR feature can be assigned to a main switch (System 85 or DEFINITY Generic 2.1) producing what is known as an "intelligent main." This is usually done when the Main / Satellite / Tributary configuration accesses an ETN (Electronic Tandem Network) tandem switch. Assigning AAR to a main location gives the main location the private-network routing capabilities that apply to the main itself. Some of these capabilities include:

- DDD (Direct Distance Dialing) overflow for outgoing calls when the tie trunks between the main and the tandem are heavily used
- Authorization-code screening by the local main for outgoing calls
- Traveling class marks (FRLs and / or Call Categories) generated locally by the main and passed on to the ETN tandem for outgoing calls
- Filtering a partial thousand's group of extension numbers to decide which extension numbers for incoming calls terminate to the Main / Satellite / Tributary configuration and which numbers should be routed to the local CO as tail-end hop off calls.

Call Coverage

Call Coverage cannot redirect a call over a tie trunk. Therefore, a terminal user at a satellite location cannot provide coverage for a terminal user at the main.

CAS (Centralized Attendant Service)

A Main / Satellite complex cannot use the Centralized Attendant Service feature. If a satellite location in a Main / Satellite arrangement is also translated as a branch location in a CAS arrangement, neither feature will work properly. (Tie trunks in the network will lock up.)

For Main / Satellite configurations, a centralized attendant capability can be provided by using the Extended Trunk Access function of the Main / Satellite feature.

DDC (Direct Department Calling) and UCD (Uniform Call Distribution)

Extension numbers that use extension number steering cannot be the controlling extension for a UCD (Uniform Call Distribution) or DDC (Direct Department Calling) group.

ISDN—PRI (Primary Rate Interface)

The Main / Satellite / Tributary networking feature is not compatible with the ISDN—PRl feature. That is, ISDN trunking arrangements cannot be used within a Main / Satellite or Tributary network The Main (assuming it is a System 85, Release 2, Version 4 or DEFINITY Generic 2 switch) can use the ISDN—PRI feature to terminate trunks from outside the Main / Satellite network (private- or public-network trunks); however, ISDN service cannot be used within the Main / Satellite or Main / Tributary network. This is because ISDN—PRI trunks require ETN type routing with a message type signaling with is not provided in a Main / Satellite arrangement. See also the interaction discussion for the DCS feature.

Intercept Treatment

At a satellite location, the ETA function of the Main / Satellite feature overrides one cause of Intercept Treatment. When ETA is assigned, calls to vacant dial access codes do not receive Intercept Treatment from the satellite. Instead, the satellite outpulses the collected digits to the main over the designated trunk group. For these calls, if the main determines that Intercept Treatment is required, the main will return intercept across the trunk to the calling party at the satellite. Otherwise, the main will handle the call based on its complete numbering plan.

The two other forms of Intercept Treatment (calls to recently disconnected stations and calls to restricted features or trunks) operate normally at a satellite location.

Last Extension Dialed

The Last Extension Dialed feature cannot be used for calls to DCA (Data Communications Access) trunk groups that use extension number steering.

Look-Ahead Interflow

The Look-Ahead Interflow feature requires ISDN—PRI facilities from the sending switch to the receiving switch. However, since Main / Satellite trunk groups cannot be used for ISDN trunking arrangements, the Look-Ahead Interflow feature cannot be used in a Main / Satellite / Tributary arrangement.

If a private network's switches are configured as a Main / Satellite arrangement prior to the implementation of Look-Ahead Interflow, these switches must be reconfigured as an ETN (Electronic Tandem Network) with office code routing. To retain the extension-number dialing capabilities (offered by Main / Satellite) within this new ETN configuration, Main / Satellite (70-series) trunk groups can reside in ***parallel*** to the ETN (40-series) trunk groups. Or, as an alternative, the Extension Number Portability feature can be used to emulate the desired Main / Satellite capabilities.

Modem Pooling

Data rolls placed through main / satellite arrangements over analog tie trunks require Modem Pooling connections at each end of the call where digital facilities are used. A data call to or from a DCP-interfaced endpoint at a main or satellite switch must be supported by Modem Pooling. Modem Pooling at the main cannot provide conversion for a data endpoint at a satellite location.

Multiple Listed Directory Numbers

If a Main / Satellite complex uses DAC steering, the LDN (Listed Directory Number) at the main should not begin with the same first digits that are assigned to a Main / Satellite trunk group. Furthermore, LDNs should not begin with a digit assigned to a feature dial access code. The exception is the attendant access code. The leading digit(s) of an LDN can be the same as the attendant access code. Likewise, an LDN can have the same leading digit(s) as local extension numbers.

Queuing

When a Main / Satellite complex serves as an access arrangement for an ETN, the tandem switch and the main switch can both have ringback queuing. However, a single queue (located at the tandem) serves both switches. If the tandem switch attempts to ringback a local terminal user (no tie trunk involved), the switch ignores certain features. These features are:

- Call Forwarding—Follow Me
- Call Forwarding—Busy and Don't Answer
- Call Pickup
- Call Waiting
- Call Coverage
- Hunting.

When a tandem or main routes a ringback call to a subtending location by way of a tie trunk, the preceding features work normally. The reason is that the subtending switch sees the call as an ordinary incoming tie trunk call.

Route Advance

To route calls through a Main / Satellite / Tributary configuration, extension number steering can point to the dial access code corresponding to the first trunk group in a Route Advance sequence. When this is done, an alternate route can be selected so that the call can either route over a parallel direct trunk group to the same location or route to another location for subsequent steering to the desired location.

Tenant Services

A partitioned System 85 or DEFINITY Generic 2 can serve as a satellite or a tributary in a Main / Satellite / Tributary arrangement. As long as the partitioned switch is serving as an endpoint in the arrangement, an extension partition and optional attendant partition within the partitioned switch can access the main over dedicated trunk groups. However, a partitioned System 85 or DEFINITY Generic 2 should not serve as the main. When this is done, the main does not provide trunk-to-trunk partitioning to the public or private networks.

WCR (World Class Routing)

The Main / Satellite feature functions with the WCR feature (on DEFINITY Generic 2.2 switches) in the same way that it did with the earlier networking features, AAR and ARS. Although, not required, the WCR feature can be assigned to a main producing what is known as an "intelligent main." This is usually done when the Main accesses an ETN (Electronic Tandem Network). Assigning WCR to a main location gives the main network routing capabilities that apply to the main itself. Some of these capabilities include:

- DDD (Direct Distance Dialing) overflow for outgoing calls when the tie trunks between the main and the network are heavily used
- Authorization-code screening by the local main for outgoing calls
- Traveling class marks (FRLs and / or Call Categories) generated locally by the main and passed on to the ETN for outgoing calls
- Filtering a partial thousand’s group of extension numbers to decide which extension numbers for incoming calls terminate to the Main / Satellite / Tributary configuration and which numbers should be routed to the local CO as tail-end hop off calls.

Hardware Requirements

The Main / Satellite feature requires the following special hardware.

For Traditional Modules

Each Main / Satellite trunk requires:

- One circuit of an SN233 circuit pack (four circuits per SN233).

or

- One channel of an ANN11 (DS1) circuit pack (24 channels per ANN11).

For Universal Modules

Each Main / Satellite trunk requires:

- One circuit of a TN760C circuit pack (four circuits per TN760C).

or

- One channel of a TN767 (DS1) circuit pack (24 channels per TN767).

Feature Administration

Assignment of the Main/Satellite/Tributary feature is on a per-switch basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administer using the Manager IV.

To administer location code routing patterns for a Main/Satellite/Tributary complex, refer to the Feature Administration section of the AAR, ARS, or WCR feature descriptions.

The following are the administration procedures.

ADMINISTRATION PROCEDURES — MAIN/SATELLITE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
100	1	Assigns the trunk-group access code and the trunk type to a Main/Satellite trunk group. The applicable trunk-type encodes are: 70 1-way in immediate start 71 1-way out immediate start 72 2-way immediate start both ways 73 1-way in wink start 74 1-way out wink start 75 2-way wink start both ways 76 1-way in delay dial 77 1-way out delay dial 78 2-way delay dial both ways.	No
104	1 & 2	Assigns Main/Satellite network and trunk group parameters including ETA (Extended Trunk Access) and Remote Dial Transfer.	No

(Continued)

ADMINISTRATION PROCEDURES — MAIN/SATELLITE *(Continued)*			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
276	1	Assigns the Standard Network and Multipremises capabilities to the feature-group class of service.	No
350	1	Assigns the first digit of a feature dial access code, trunk-group dial access code, or extension number. The call type for extension numbers that use extension number steering must be "Extension Number."	No
350	2	Assigns the feature dial access codes.	No
354	1	Administers blocks of extension numbers for extension number steering.	No
354	2	Assigns an extension number or an extension number steering code to a dial access code.	No

Malicious Call Trace

Description

The MCT (Malicious Call Trace) feature allows the customer to obtain information that may identify the calling party of a malicious call. This feature is useful for tracing the source of bomb threats, menacing calls, and obscene calls, and is especially desirable when these types of calls are more than a rare occurrence. Organizations that may wish to use the Malicious Call Trace feature include airlines, corporations, government agencies, and universities.

Activation and Control of a Trace

The Malicious Call Trace feature can be activated from any voice terminal or attendant console in the switch. However, this feature is assigned to voice terminal classes of service as a way of preventing accidental or mischievous activation.

Attendant Alerting

When the Malicious Call Trace feature is activated, every attendant on the local DCS (Distributed Communications System) node is alerted that there is a malicious call that needs to be traced. In a ***partitioned System 85 or DEFINITY Generic 2 (Tenant Services),*** every attendant in Partition 0 is alerted to trace the malicious call. While the attendants are being alerted, trace information is stored and a voice recorder is connected to the malicious call.

Controlling Attendant

The first attendant to respond to this alert acts as the controlling attendant who performs the trace, and subsequently deactivates the feature. The controlling attendant could be the person who activated the MCT feature or another attendant.

> **NOTE:** The recorder is connected, and the trace information is stored regardless of whether an attendant takes control.

During the trace, the controlling attendant receives the trace information on the console's 8-character alphanumeric display. An ordered list of the information displayed is shown in Table 81-A.

Switch Response

When the MCT feature is activated, the switch gathers the trace information and all equipment used to place the call is held up (that is, cannot be disconnected) until this feature is deactivated. For example, the system would hold up the called parties' extension and the trunk used in a malicious call that originated at another switch. If a third party is connected to this call when the MCT feature is activated, the third party's equipment (trunk or extension) will be held up until the MCT feature is deactivated.

TABLE 81-A. Format of the Controlling Attendant's 8-Character Display

Item	Definition	Possible or Sample Displays
1	Identification of Called Party	67257 (voice terminal extension), or CONSOL1 (attendant)
2	Source of Malicious Call	LINE, or TRUNK, or ISDNCALL
3	Identification of Malicious Caller	65423 (extension), or 0112182 (7-digit equipment location), or 212 2745843 (10-digit ISDN number)
4	Called Party's status	ACTIVE*, or IDLE
5	Type of Third Party (if applicable)	3RDLINE, or 3RDTRUNK
6	Third Party Identification (if applicable)	67743 (extension), or 0213163 (trunk equipment location)
7 or 5	Identification of Activating Party	67257 (called extension), or 67266 (neighbor), or CONSOL1 (called attendant)
8 or 6	Identification of Voice Recorder	0022051 (7-digit equipment location), or NO REC (no voice recorder)
9 or 7	End of Trace Information	END

* Whenever the controlling attendant is also the called party, this display will show "ACTIVE."

Feature History and Development

The Malicious Call Trace feature was first available for System 85 in Release 2, Version 4.

User Operations

To Activate Malicious Call Trace From a Multiappearance Voice Terminal Without an EMERGENCY Button:

1. Be sure you are active on a malicious call.
2. Press the **[CONFERENCE]** , **[TRANSFER]** , or **[HOLD]** button* to place the malicious call on hold. [The calling party is placed on hard hold, and hears silence or music (if Music-on-Hold is administered).]
3. Select a new line appearance.

* CONFERENCE or TRANSFER is the recommended method. If all multiple line appearances are active (or fluttering), the operation for activating MCT (Malicious Call Trace) is like the one used by a neighboring voice terminal.

4. Dial the MCT Emergency access code. [Second dial tone]
5. Press the **[#]** button. [The switch returns confirmation tone. The attendants are alerted to trace the call. Trace information is stored. A voice recorder is activated.]
6. Press the appropriate appearance button to return to the held call,

   or

   Go on-hook.

To Activate Malicious Call Trace From a Multiappearance Voice Terminal With an EMERGENCY Button:

1. Be sure you are active on the call that you determined to be a malicious call.
2. Press the **[EMERGENCY]** button. [The green status lamp lights. The attendants are alerted to trace the call. Trace information is stored. A voice recorder is activated.]

To Activate Malicious Call Trace From a Rotary Voice Terminal:

1. Be sure you are active on a malicious call.
2. Momentarily press the switchhook. [The switch returns recall dial tone. The calling party is placed on soft hold and hears silence.]
3. Dial the MCT Emergency access code. [Second dial tone]
4. Momentarily press the switchhook again. [The switch returns confirmation tone. The attendants are alerted to trace the call. Trace information is stored. A voice recorder is activated.]
5. Momentarily press the switchhook to return to the held call,

   or

   Go on-hook.

To Activate Malicious Call Trace From a Touch-Tone Voice Terminal:

1. Be sure that you are active on the call which you determined to be a malicious call.
2. Press the **[RECALL]** button (if provided),

   or

   Momentarily press the switchhook. [The switch returns recall dial tone. The calling party is placed on soft hold and hears silence.]
3. Dial the MCT Emergency access code. [Second dial tone]

4. Press the **[#]** button. [The switch returns confirmation tone. The attendants are alerted to trace the call. Trace information is stored. A voice recorder is activated.]

5. Press the **[RECALL]** button or momentarily press the switchhook to return to the held call,

   or

   Go on-hook

To Activate Malicious Call Trace for a Neighboring Voice Terminal:

1. Be sure a neighboring voice terminal user is receiving a malicious call.
2. Go off-hook. [Dial tone]
3. Dial the MCT Emergency access code. [Second dial tone]
4. Dial the neighbor's s extension number. [The switch returns confirmation tone. The attendants are alerted to trace the call. Trace information is stored. A voice recorder is activated.]
5. Go on-hook.

To Activate Malicious Call Trace From an Attendant Console:

1. Be sure you are active on a malicious call.
2. Press the **[MCT EMERGENCY ACTIVATE]** button. [The POS BUSY lamp lights. The attendants are alerted to trace the call. Trace information is stored. A voice recorder is activated]
3. Do one of the following options.

Option One

4. Allow another attendant to press the **[MCT CONT]** button and to act as the controlling attendant. [The activating attendant has a talking connection with the malicious caller. The controlling attendant does not.]
5. Continue to monitor the malicious call.
6. After the calling party disconnects, press the **[RELEASE]** button.

   NOTE: This is the recommended attendant operation.

Option Two

4. Press the **[MCT CONT]** button to act as both the activating and the controlling attendant. [The attendant maintains a talking connection with the malicious caller.]
5. While tracing the malicious call, continue to monitor the call.*

---

* Refer to the following instructions for tracing a malicious call.

6. After the calling party disconnects, press the **[RELEASE]** button.
7. After tracing the call, deactivate the Malicious Call Trace feature.*

Option Three:

4. Ignore the call and return to normal activity. To do this, press the **[RELEASE]** button and then the **[POS BUSY]** button. [Talking connection with the malicious caller ceases, and the POS BUSY lamp goes out.]

To Trace a Malicious Call:

When the MCT feature is activated, the MCT CONT lamp flashes at every attendant console. The first attendant to press this button is the controlling attendant. The controlling attendant is responsible for tracing the malicious call. The controlling attendant should:

1. Press the **{MCT CONT]** button. [Audible signal stops, MCT CONT lamp continues to flash, the POS BUSY lamp lights, and the first ICI (Incoming Call Identification) message is displayed. Unless this attendant also activated the feature, there is no talking connection with the malicious caller.]
2. Record the ICI 8-character display. Examples are shown in Table 81-A.
3. Press the **[MCT CONT]** button to receive each subsequent display, and record the information. (This process is finished when the ICI display shows "END.")
4. If the call arrived over a trunk, call the Central Office or the distant private network node to continue the trace at that location. [The distant attendant should have a cross-reference and be able to find the corresponding equipment location at the distant switch. This is described further in "Considerations".]
5. While the call is being traced, the controlling attendant can place the console into normal operation by doing the following. (If the controlling attendant is also the activating attendant, the attendant should continue to monitor the call until the trace is finished.)
   a. Place the MCT switch loop on hold.
   b. Press an idle loop. [The console is no longer locked up.]
6. When the trace is finished, the information has been recorded, and the malicious call has completed; deactivate the MCT feature.*

To Deactivate the Malicious Call Trace Feature

CAUTION: *Deactivating the* MCT *feature will disconnect the voice recorder and all talking parties.*

---

* Refer to the following instructions for deactivating a malicious call.

The controlling attendant should:

1. Press an idle loop button. [ATND lamp lights.]
2. Press the **[START]** button. [Dial tone]
3. Dial the MCT Deactivate access code. [MCT CONT quits flashing, the voice recorder is deactivated, and POS BUSY lamp goes out.]

If the controlling attendant is also the activating attendant, do the following:

1. Place the MCT loop on hold.
2. Press an idle loop button. [ATND lamp lights.]
3. Press the **[START]** button. [Dial tone]
4. Dial the MCT Deactivate access code. [MCT CONT quits flashing, the voice recorder is deactivated, and POS BUSY lamp goes out.]

Considerations

Cross-Reference of Trunk Equipment Locations

During the tracing of a malicious call, System 85 and DEFINITY Generic 2 provide an ICI display of the equipment location of the incoming trunk (when ISDN information is not received, or when the malicious call did not originate from within a DCS network). In order for this information to be useful, each attendant should be supplied with an up-to-date cross-reference of the equipment location of each incoming (and 2-way) trunk circuit in the local switch and the corresponding equipment location at the distant switch. It would also facilitate the trace to provide a third column in the cross-reference containing an appropriate telephone number to the distant switch. In this way, a controlling attendant is provided with the necessary information to easily extend the trace.

It might also be useful for the switch administrator to keep a cross-reference between the equipment locations of SN231 recorder trunks and the corresponding voice recorders. During the trace, the switch provides the equipment location of the seized auxiliary trunk (the recorder trunk Item 8 or 6 of Table 81-A). So, the cross-reference would facilitate the process of obtaining the appropriate tape.

Deactivating From an Accidentally Activated Trace

If a voice terminal user erroneously activates a Malicious Call Trace, the method of deactivation depends on whether the activation is with a single-appearance or a multiappearance terminal.

Single-Appearance Terminal — A second switchhook flash or Recall press before or after time-out to confirmation tone activates Malicious Call Trace. To deactivate the trace, hang up before confirmation tone.

Multiappearance Terminal — To deactivate the trace, return to held appearance before time-out to confirmation tone.

Once confirmation tone is heard, the only way to deactivate the feature is to have an attendant on the switch deactivate this feature.

Dial Access Activation

When a voice terminal user activates Malicious Call Trace toward the user's own extension number with the dial access code, there are four legitimate responses when the switch returns the second dial tone:

- The reommended (and quickest) response is to momentarily press the switchhook.
- If that is not possible, press the "#" button. In this way, the amount of dialing is minimal and activation is immediate.
- A third acceptable response is to dial the user's own extension number. This method requires extra dialing and is somewhat slower.
- The fourth acceptable response is to do nothing. When time-out occurs (in approximately 10 seconds), the switch infers that activation is toward the user's own extension number.

Disconnecting All Parties

Once a Malicious Call Trace is activated, controlled, and deactivated, all previously connected parties are disconnected. This releases all held up equipment.

Freeing the Activating Attendant's Console

When Malicious Call Trace is activated from one attendant's console but controlled from another console, the activating console is locked up. Place the MCT switch loop on hold and press an idle loop on the activating console to allow further call processing.

Legal Considerations

Laws governing the use of the Malicious Call Trace feature vary in different locations and are subject to change. It is the responsibility of the customer's switch administrator to understand and comply with the applicable regulations.

Multiple Appearances of an Extension

Users with more than one active appearance of their own extension numbers can activate Malicious Call Trace toward their own extensions using the dial access code. In this situation, when a user presses the CONFERENCE, TRANSFER, or HOLD* button before dialing the MCT access code, the switch presumes that the appearance that was ***just placed on hold*** (as a result of the CONFERENCE, TRANSFER, or HOLD activation) is the appearance to which Malicious Call Trace ***is being activated.***

* CONFERENCE or TRANSFER is the recommended method. If all multiple line appearances are active (or fluttering), the operation for activating MCT is like the one used by a neighboring voice terminal.

The user of a neighboring voice terminal is not allowed to activate Malicious Call trace toward another extension with more than one appearance active (or fluttering). In this situation, the switch has no way of knowing which appearance actually has the malicious call on it. When this is attempted, the switch returns intercept tone.

On stations that receive more than one malicious call, the MCT feature can be activated toward more than one appearance of an extension. As long as there is no ambiguity during ***each*** activation, Malicious Call Trace could be activated toward every call on a multiappearance voice terminal.

Periodic Check of Voice Recorders

Since this feature does not provide electronic monitoring of tape availability, the voice recorder(s) should be checked regularly to ensure there is plenty of tape available for recording malicious calls.

Published Numbers and Malicious Call Trace

For many applications of this feature, malicious calls tend to be placed by individuals outside the organization. These people are likely to place malicious calls to numbers found in the public directory. When this is the case, attendants, ACD agents, and other voice terminal users answering calls directed to published numbers should be equipped with a feature (EMERGENCY ACTIVATE) button for convenient activation of Malicious Call Trace.

Switch Capacity

System 85 and DEFINITY Generic 2 software allows as many as 15 malicious calls to be traced at any one time. As soon as MCT is activated, all ICI data is stored and a Voice Recorder is connected to the call. This information is maintained until an attendant takes control and eventually deactivates MCT for that call. However, if an attendant is not available to control the MCT and read the ICI data, there will be a delay in identifying the caller — especially if a distant operator must be called to trace the call through another switch.

Hard and Soft Processor Swaps

The contents of the MCT trace information are stored in a status portion of switch memory. Therefore, if a hard swap occurs before or while the controlling attendant is reviewing the ICI data, the unread part of the trace information is lost.

Interactions With Other Features

Attendant Auto-Manual Splitting

If an attendant activates Malicious Call Trace while a calling party is split from the console, the split condition is disabled. On the attendant console, the split lamp goes out For the previously split party, the switch returns dial tone to an internal caller or disconnects an outside caller from the switch.

Attendant Interposition Calling and Transfer

If an attendant selectively calls the activating or the controlling attendant during an active Malicious Call Trace, the switch returns ringback to the calling attendant. However, the called attendant will only receive ringing if the attendant has unbusied the console.

For the controlling attendant, priority lamp activity is provided for interposition calls during a trace. However, the controlling attendant does not receive an ICI display for these calls. Instead, the alphanumeric display is reserved for displaying trace information.

For the activating attendant to receive an interposition call, the malicious caller must be placed on hold, and the console must be unbusied. Then, an interposition call will cause the loop lamp to flash and the PR lamp to go out.

ACD (Automatic Call Distribution)

If an ACD agent receives a malicious call, the Malicious Call Trace feature can be activated by the ACD agent. It is preferable, when malicious calls are more than a rare occurrence, to assign the EMERGENCY button to every agent’s multiappearance voice terminal.

If an ACD agent receives a malicious call and activates the Malicious Call Trace feature while the agent is being observed using service observing (as opposed to Agent Override), the observer may continue to monitor the malicious call and subsequent calls normally. Furthermore, service observing allows the obsever to ***begin observation*** after Malicious Call Trace has been activated.

City/Queue-of-Origin Announcement

An ACD agent cannot activate Malicious Call Trace during a city-of-origin or a queue-of-origin announcement. When this 2-second announcement finishes, the trace can be activated.

CMS (Call Management System)

With CMS, "agent is on MCT" messages are sent to the processor (3B or AP 16) and interpreted as EXCEPTION messages by the CMS software. An Exception Message will appear on any real-time report displaying data for that split.

Agent Override

If an Agent Override call is attempted toward a line involved in a Malicious Call Trace, the malicious caller receives intercept tone.

Call Waiting

Calls are not allowed to wait on a line involved in a Malicious Call Trace. Instead, the calling party receives busy tone.

CAS (Centralized Attendant Service)

A malicious call placed to a branch location in a CAS arrangement cannot be controlled by a CAS attendant at the main switch. However, when the MCT feature is assigned at a branch location, malicious calls can be controlled by a special services attendant at the branch location if one is assigned.

A malicious call placed to the main location in a CAS arrangement can be controlled by a CAS attendant at the main switch.

Conference—Three Party

When Malicious Call Trace is activated during a 3-party conference, the malicious caller's identity in the trace information becomes uncertain. The contents of Items 2 and 5 and of Items 3 and 6 in the controlling attendant's display can be reversed.

If there is one (and only one) party in the 3-party conference who is connected to the switch over an incoming trunk. The MCT software presumes that this party is the malicious caller. This party is identified as the malicious caller in Items 2 and 3 of the controlling attendant's display.

Sometimes, this ambiguity can be avoided with human intervention. Since the malicious caller usually initiates a malicious call, the originally called party can press the DROP button to disconnect the third party in the conference, verify that the malicious caller is still on the connection, and then quickly activate the trace. (However, when this is done, the third party cannot be readded to the conference.)

Dial Access to Attendant

If a voice terminal user selectively calls the activating or the controlling attendant during an active Malicious Call Trace, the call is denied. The switch returns busy tone to the calling party. A controlling attendant unbusying his or her console will receive the call without disturbing the ICI display that contains the trace information.

DCS (Distributed Communications System)

If a malicious call originates from a voice terminal from within the DCS network, the ICI display on the controlling attendant console displays the extension number of the calling voice terminal. Beyond this functionality, DCS transparency is not provided for the Malicious Call Trace feature.

A voice terminal user cannot activate Malicious Call Trace unless an attendant is in the same DCS node as the user. Also, a third party cannot activate the feature unless the third party is in the same node as the user.

When Malicious Call Trace is activated in a DCS environment, every System 85 or DEFINITY Generic 2 attendant in the local DCS node is alerted to trace the call.

ISDN (Integrated Services Digital Network)

When a trace is performed toward a malicious call that came in over ISDN facilities and the ISDN facilities deliver the calling number to the local System 85 or DEFINITY Generic 2, the ICI display on the controlling attendant console will display the 10-digit calling number. The first three digits are displayed first followed the last seven digits.

Look-Ahead Interflow

At a receiving switch, the Look-Ahead Interflow feature and the Malicious Call Trace feature are compatible. An attendant or an authorized voice terminal user at the receiving switch can activate Malicious Call Trace toward Look-Ahead Interflow calls in the normal manner. The controlling attendant can trace these calls to (and through) the sending switch in the normal manner. Also, as the controlling attendant executes the trace, the attendant will receive the 10-digit ISDN number as the source of the malicious call if this number was originally provided to the sending switch.

Music-on-Hold Access

When Malicious Call Trace is activated on a switch that provides Music-on-Hold the originator of the malicious call receives music whenever this caller is placed on hard hold. This effect usually occurs when Malicious Call Trace is activated from a multiappearance voice terminal using the dial access code.

If this functionality is considered unacceptable, there are several ways to attenuate or circumvent the problem. To minimize the caller's alarm, the person activating Malicious Call Trace can advise the caller, "I need to put you on hold for a short time. I'll be right back." Other alternatives are to provide MCT ACTIVATE buttons for voice terminals that are likely to receive malicious calls, or to activate Malicious Call Trace from a neighboring voice terminal. If the previous alternatives are considered unacceptable, Music-on-Hold can be disabled using Procedure 275, Word 1.

Override

If an Override call is attempted toward a line involved in a Malicious Call Trace, the calling party receives busy tone.

Priority Calling

If a priority call is attempted toward a line involved in a Malicious Call Trace, the calling party receives busy tone.

Tenant Services

An attendant in any attendant partition or a voice terminal user in any extension partition can activate the Malicious Call Trace feature in response to a malicious call.

When Malicious Call Trace is activated in a partitioned switch, only the attendants in Partition 0 are alerted to the malicious call.

Voice recorder trunks are not partitioned. These trunks, when available, are equally accessible to any partition needing them.

Timed Recall on Outgoing Calls

During a malicious call, the Timed Recall feature is deactivated for a trunk that is involved in the malicious call.

The activating console receives modified Timed Recall operation during a Malicious Call Trace. If the activating console is position busy, the Priority lamp lights during a timed recall, but the tone does not sound. If the console is not position busy, the Priority lamp lights and the tone sounds during a timed recall.

The controlling console also receives modified Timed Recall operation during a Malicious Call Trace. While the controlling attendant is performing the trace, the Priority lamp lights during a timed recall, but the tone does not sound. After the trace is deactivated, the Priority lamp lights and the tone sounds for a timed recall.

Timed Reminder

The activating console receives modified Timed Reminder operation during a Malicious Call Trace. While the activating attendant is connected to the malicious call, the Ring lamp flashes during a timed reminder but the tone does not sound. After the activating attendant disconnects from the malicious call, the Ring lamp flashes and the tone sounds during a timed reminder.

The controlling console also receives modified Timed Reminder operation during a Malicious Call Trace. While the controlling attendant is performing the trace, the Ring lamp flashes during a timed reminder, but the tone does not sound. After the trace is deactivated, the Ring lamp flashes and the tone sounds during a timed reminder.

Trunk Verification—Attendant

The Trunk Verification—Attendant can assist in tracing malicious calls that originate from or tandem through distant switches in the private network. After an attendant at the distant switch is called by the controlling attendant at the local switch, the distant attendant can enter the call using the Trunk Verification—Attendant feature and then activate Malicious Call Trace at the distant end.

> **NOTE:** Since the warning tone provided by trunk verification could arouse suspicion by the malicious caller, the verification should be deactivated as quickly as possible.

Unattended Console Service—Alternate Console Position

While the Alternate Console Position feature is activated, the alternate attendant position (instead of the regular attendant position) is alerted to trace malicious calls. If the alternate attendant responds to an alert first (by pressing the MCT CONT button), this attendant can also trace the call.

Unattended Console Service—Call Answer From Any Voice Terminal

While the CAAVT (Call Answer From Any Voice Terminal) feature is active, the switch does not sound the CAAVT signaling device in response to a malicious call. Instead, the switch will activate a voice recorder and alert the attendant consoles in the normal manner. After an attendant returns to an attendant position, the trace can be performed and the MCT feature can be deactivated.

Unattended Console Service—Preselected Call Routing

While the Preselected Call Routing feature is active, preselected voice terminals are not alerted to trace malicious calls. Instead, the switch will activate a voice recorder and alert the attendant consoles in the normal manner. After an attendant returns to an attendant position, the trace can be performed and the MCT feature can be deactivated.

Visually Impaired Attendant Service

A visually impaired attendant should not attempt to trace a malicious call. The controlling attendant must be able to see and record the ICI messages. Also, the controlling attendant must be able to read and communicate the information from the cross-reference of trunks.

Restricting Feature Use

Voice terminal users can be prevented from inadvertently or mischievously activating Malicious Call Trace by denying access in the class of service. This is done with the Restrictions — Voice Terminal Restrictions feature, using Procedure 010, Word 2.

Hardware Requirements

The following special hardware is required for the Malicious Call Trace feature.

For Traditional Modules:

- SN231 Auxiliary Trunk Circuit

  An auxiliary trunk circuit on an SN231 circuit pack is required for each voice recorder (four circuits per SN231). An auxiliary trunk circuit provides the switchable talking path between the malicious call and the recorder. When MCT is activated, the SN231 passes audio signals and on/off control to the recorder via the 278A interface.

  Each trunk port on this circuit pack used for this feature should be optioned (assigned) as a 1-way outgoing and 4-wire signaling trunk.

For Universal Modules:

- TN763C Auxiliary Trunk Circuit

  An auxiliary trunk circuit on an TN763C circuit pack is required for each voice recorder (four circuits per TN763C). An auxiliary trunk circuit provides the switchable talking path between the malicious call and the recorder. When MCT is activated, the TN763C passes audio signals and on/off control to the reorder via the 278A interface.

  Each trunk port on this circuit pack used for this feature should be optioned (assigned) as a 1-way outgoing and 4-wire signaling trunk.

Regardless of the Module Type:

- Attendant consoles with 8-character alphanumeric displays.

  An attendant console with 8-character display is needed to perform the control functions needed for Malicious Call Trace.

- Voice Recorder

  This is CPE (Customer Provided Equipment) used to record malicious calls. The recorder selected must be suitable for remotely controlled operations. It should have an automatic gain control or have a recording level adjustment that can be set to an appropriate level.

- 278A Adapter

  One 278A Adapter is needed for each voice recorder. This unit is recommended because it provides isolation between SN231 or TN763C trunk equipment and the recorder. Without the 278A, there is no protection on the Audio Path from power surges.

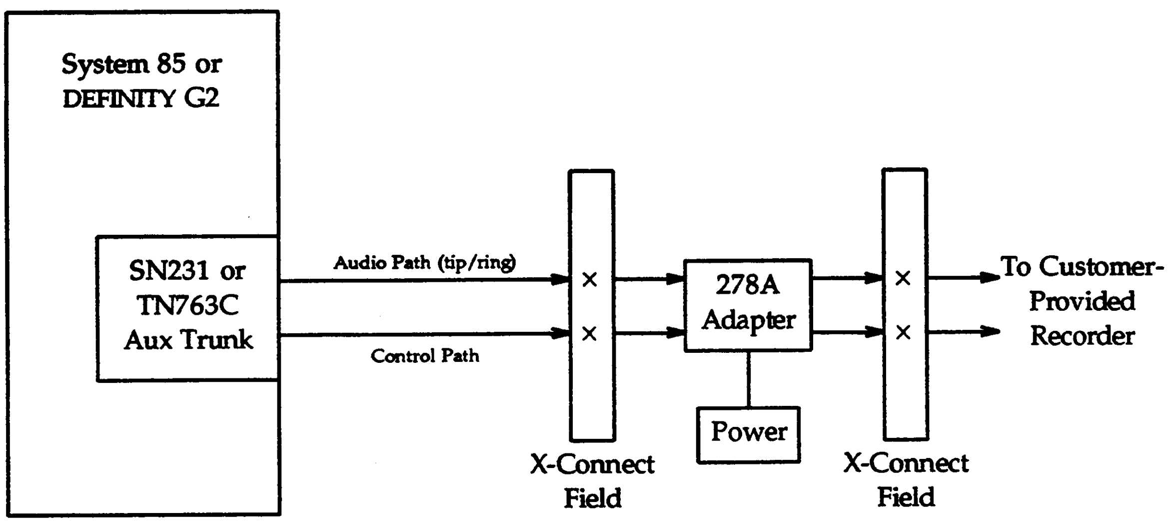

Figure 81-1. Malicious Call Trace Hardware Configuration

The 278A has two options and both can protect the audio path on the trunk circuit. The C1/C2 option is the recommended option because it has 17 ohms of resistance (impedance) in series with its relay contacts that provide current-limiting protection to the trunk circuit. If the recorder is unable to work with this impedance to the microphone input of the recoder, use the BY1/BY2 option. This option does not have current-limiting protection but has approximately zero ohms of impedance to the microphone input.

The 278A can be mounted in the Auxiliary Cabinet or wall mounted next to the cross-Connect fields.

- Power supply

  A D-181321 power kit is required for each 278A control unit. This unit converts -48 V dc to -24 V dc power. If the 278A is wall mounted, then it should be powered by a plug-in wall power supply (KS-22911L1). If the 278A is installed in the Auxiliary Cabinet, use cabinet power. In either case, the ground lead which goes to the 278A should eventually be attached to the single-point ground in the equipment room.

Feature Administration

The Malicious Call Trace feature is assigned on a per-voice terminal and voice terminal class of service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — MALICIOUS CALL TRACE			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the voice terminal class of service to an extension number.	Yes
010	2	Assigns the ability to activate Malicious Call Trace to a voice terminal class of service.	Yes
054	2	Assigns the EMERGENCY button to a multiappearance voice terminal. The applicable encode is as follows: 9 Malicious Call Trace.	Yes
100	1	Assigns the trunk-group dial access code and trunk type of a voice recorder trunk to its trunk-group number. The applicable trunk-type encode includes: 93 Malicious call trace recorder.	No
150	1	Assigns an SN231 or TN763C equipment location to its trunk-group number administered in Procedure 100, Word 1.	No
203	1	Assigns the MCT—Activate and the MCT—Control buttons to the attendant console(s). The applicable encodes are as follows: 48 MCT—Activate 49 MCT—Control.	No
350	1	Assigns the first digit of the Malicious Call Trace feature dial access codes (if required).	No
350	2	Assigns the Malicious Call Trace dial access codes. The applicable encodes are: 99 MCT — Deactivation 100 MCT — Activation.	No

Manual Signaling

Description

This feature allows a multiappearance terminal user to signal a preselected multiappearance terminal. Each button press at the calling terminal causes one short beep at the called terminal. This feature may be used in conjunction with the Manual Intercom feature to provide signaling.

Manual Signaling can also be used to signal a preselected group of voice terminals. In this way, a message can be quickly and effectively conveyed (such as, "Departmental meeting is convening") to an organized group.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since its introduction.

User Operations

The following is the user operating procedure for this feature.

To Signal a Terminal(s):

Press **[SIGNAL].** [Tone is generated at the signaled terminal(s).]

Considerations

Manual Signaling Tone

The tone is a 750-hertz buzzing tone.

Feature Parameters

It is recommended that one button should signal no more than 17 multiappearance terminals. Each Manual Signaling pair [i.e., association of a signaling terminal and the signaled terminal(s)] requires a separate signaling button.

Straight Line Sets

Straight line sets cannot be assigned as either the signaling or a signaled terminal in a Manual Signaling pair. When this is attempted, an administration error will occur in Procedure 053, Word 1.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Intercom—Automatic

When Automatic Intercom ringing is in progress, Manual Signaling is prevented. Broken fluttering on the Manual Signaling status lamp is used to convey the denial.

Intercom—Dial

When Dial Intercom ringing is in progress, Manual Signaling is prevented. Broken fluttering on the Manual Signaling status lamp is used to convey the denial.

Ringing—Ringing Cutoff

Ringing Cutoff denies Manual Signaling for the given terminal.

Ringing—Distinctive Ringing

When Distinctive Ringing is in progress, Manual Signaling is denied. Broken fluttering on the Manual Signaling status lamp is used to convey the denial.

Tenant Services

There are no tests in Procedure 053, Word 1 to ensure that the members of a Manual Signaling pair belong to the same extension partition. It is the responsibility of the system manager to ensure that Manual Signaling pairs do not cross partition boundaries.

Hardware Requirements

Multiappearance voice terminals are required for this feature.

Feature Administration

Assignment of the Manual Signaling feature is on a per-terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can fully administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE — MANUAL SIGNALING			
PROCEDURE	WORD	PURPOSE	SMT
053	1	Assigns the Manual Signaling button to a multiappearance voice terminal. Also, the procedure associates a signaling terminal with a signaled terminal(s).	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREEN — MANUAL SIGNALING	
PATH NAME	PURPOSE
terminal-change terminal buttons	Assigns the Manual Signaling button to a multiappearance voice terminal.
terminal-change terminal manual-signaling	Associates the signaled terminal(s) with a signaling terminal.

Notes:

Message Waiting — Automatic

Description

Message Waiting—Automatic provides two ways of automatically notifying a user that a message is waiting to be retrieved from one or more of the System 85 or DEFINITY Generic 2 messaging services. Two different message waiting indicators are used because not all voice terminals have the same alerting facilities. The Message Waiting — Automatic indicators should meet the needs of most users The two Message Waiting — Automatic indicators are:

- Audible Message Waiting

  The audible message waiting indicator is a distinctive ***stutter tone,*** heard through the voice terminal ear piece or speaker for 2-seconds when the terminal first goes off-hook. This message waiting indicator can be used as follows:

  — For voice terminals that do not have a message waiting lamp

  — To supplement the function of an available message waiting lamp

  — To indicate messages waiting on an alternate extension.

  When this tone is heard, the user knows that there is a message waiting to be retrieved.

- Message Waiting — Lamp

  The message waiting — lamp is a green LED, usually located below and to the left of the voice terminal dialing pad. The message waiting lamp is available on every AT&T digital (7400D Series), hybrid (7300H Series), and the 7100A series analog voice terminals. The 234A Message Waiting Indicator is available as a voice terminal adjunct that can be used with voice terminals that do not have an integral message waiting lamp. When the message waiting lamp is lit, the user knows that a message is waiting to be retrieved.

Both forms, Audible Message Waiting and the Message Waiting Lamp, are fully compatible and can be used on the same voice terminal or even for the same extension if desired.

Basic Operation

The messaging service features of the System 85 and DEFINITY Generic 2 notify the switch when a message is left for an extension. This is discussed in more detail in the Unified Messaging feature chapter. The switch then checks the translations for the called extension.

Message Waiting Lamp

If a message waiting lamp is assigned to that extension, call processing software sends the necessary control signals to light the message waiting lamp at the called voice terminal.

Audible Message Waiting

If audible message waiting is assigned, call processing software records the call waiting status for the called extension. When any appearance of the extension goes off-hook, the switch sends ***Stutter Tone*** to the extension for 2-seconds.

Note that these operations are not mutually exclusive. That is, the same extension can be alerted by both the message waiting lamp and by audible message waiting tone.

Canceling the Automatic Message Waiting Indicators

Under certain circumstances, the Leave Word Calling feature provides a means of canceling a message from the originating voice terminal when Leave Word Calling was used to create the message. However, as a general rule the automatic message indicators will remain in effect until all messages have been retrieved. This means that with the message waiting lamp, the lamp remains lit. With audible message waiting, the stutter tone is heard each time a voice terminal goes off-hook on the assigned extension.

Feature History and Development

The function of Message Waiting Automatic was first available with System 85, Release 2, Version 1, in the form of the message waiting lamp. In this form, it was considered part of the Leave Word Calling feature, and could be used only for local (on the same switch as the terminal users) messaging services. In Release 2, Version 3, the message waiting lamp became available to remote messaging services in a DCS (Distributed Communications System).

Audible Message Waiting is first available with Release 2, Version 4, Issue 1.1. With this enhancement, Message Waiting — Automatic was given separate feature status.

User Operations

Placing an Automatic Message Waiting Call

The applicable user operations for the Message Waiting — Automatic feature depend on the messaging feature used.

For more detailed information on a specific messaging feature, see the ***User Operations*** portion of the appropriate feature chapter (AUDIX or Leave Word Calling feature).

In addition to the System 85 and DEFINITY Generic 2 messaging features, the AP adjunct feature Message Center Service can also be used to activate Message Waiting — Automatic and store messages for a voice terminal user.

In all cases, the function of the Message Waiting — Automatic feature is controlled by switch administration. The caller has no control over the type of message waiting indicator used.

Responding to Automatic Message Waiting

If your extension is assigned a ***Message Waiting Lamp,*** you should check the lamp frequently to see if it is lit. This is particularly important if you use the ***Send All Calls*** function of the Call Coverage feature or if your extension is part of a ***Send All Calls Group.***

If your extension is assigned ***Audible Message Waiting,*** you can check for audible message waiting by going off-hook and listening for message waiting (stutter) tone. This should be done any time you have been away from your voice terminal. You should also check for message waiting tone periodically when you use the ***Send All Calls*** function of the Call Coverage feature or if your extension is a member of a ***Send All Calls*** Group.

For information on retrieving messages, see the USER OPERATIONS section of the applicable messaging or message service feature. These features include:

- AUDIX
- Call Coverage
- Display — Voice Terminal
- Leave Word Calling
- Unified Messaging.

Considerations

Message Waiting Lamps per Extension

Message waiting lamps can be assigned to as many as three voice terminals using the same extension. With this limit, the switch can simultaneously send message notification to a primary extension comprised of a desk voice terminal, a guest voice terminal, and the secretary's voice terminal.

Sharing an Executive’s Message Waiting Lamp With the Secretary

Normally, the message waiting lamp corresponding to a voice terminal’s primary extension is assigned to Button Location 2 on the voice terminal. However, the button location of ***any feature button*** on the voice terminal could have an automatic message waiting lamp corresponding to another extension.

This capability is useful for secretaries whose bosses are frequently away from the office. When the boss calls the secretary asking whether there are messages, the secretary need only check the boss’s shared message waiting lamp (residing on the secretary’s voice terminal) to provide an answer.

Message Center Service

Message Center Service is an AP feature that can communicate with the switch processor. When a message is left with a message center agent, the Message Waiting — Automatic feature can be used to alert the called party to retrieve the message.

Off-Hook Time-Out Sequence

The 2-second stutter tone provided by audible message waiting does not add to the 10-second off-hook time-out sequence. That is, if the user does not dial the first digit within 10-seconds of going off-hook, the connection will time out and be given intercept treatment, whether or not audible message waiting tone is heard.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature

AUDIX

AUDIX is one of the messaging features that uses Message Waiting — Automatic. When a message is left in a user's ***AUDIX Mail Box,*** the Message Waiting — Automatic feature is activated to alert the called party to retrieve the message.

Call Coverage

Call Coverage, while not a messaging service as such, can become involved in the messaging process. Through ***implied principal addressing,*** messages left on calls that have redirected to coverage will activate Message Waiting — Automatic for the principal (the originally called party). Also, the ***Coverage Callback function*** will activate Message Waiting — Automatic for the principal.

DCS (Distributed Communications Service)

For System 85, Release 2, Version 3 and earlier switches, and for switches other than System 85 or DEFINITY Generic 2, Message Waiting — Automatic will not work for internodal DCS calls. Beginning with System 85, Release 2, Version 4 switches, where the ES (Enhanced Services) message set is used, the Message Waiting — Automatic feature will work for internodal calls.

Leave Word Calling

Leave Word Calling is one of the messaging features that uses Message Waiting — Automatic. When a Leave Word Calling message is left for a user, the Message Waiting — Automatic feature is activated to alert the called party to retrieve the message.

Tenant Services

The operations of Message Waiting — Automatic may be influenced by the Tenant Services feature, depending on how the messaging feature is administered.

- If every elment involved (calling party, called party, and messaging service) are assigned to the same partition, Message Waiting — Automatic will work normally.
- If the calling party and the messaging service are assigned to Partition 0, Message Waiting — Automatic will work normally.
- If the messaging service and the called party are in the same partition or if the messaging service is in Partition 0 and the calling party is in some other partition (other than 0 and the same partition as the called party) the call must be placed as an outside call. Message Waiting — Automatic will work for messaging services that service outside calls (AUDIX and Message Center Service).

Unified Messaging

Unified Messaging is fully compatible with the Message Waiting — Automatic feature. See the Unified Messaging feature for details.

Hardware Requirements

Message Waiting — Automatic requires the use the following special hardware.

For Message Waiting — Lamp

- A voice terminal with an integrated message waiting lamp:
  - Any 7500 Series BRI (Basic Rate Interface) telephone
  - Any 7400D Series digital voice terminal
  - Any 7300S or 7200H Series hybrid voice terminal
  - PT (Personal Terminal) 510D or 515 BCT (Business Communications Terminal
  - Any 7100A Series analog voice terminal
  - Any 2500 Series voice terminal with an integrated lamp.
- Or a Z34A Message waiting Indicator for 2500 sets without an integrated lamp.

For Audible Message Waiting

Any voice terminal that is compatible with the System 85 or DEFINITY Generic 2 can receive audible message waiting alerting.

Message Service Requirements

The Message Waiting — Automatic feature has no other hardware requirements of its own. This feature is dependent on the messaging services used, and its hardware requirements are those of the associated messaging features.

- AUDIX
- Call Coverage

- Display—Voice Terminal
- Leave Word Calling.

Feature Administration

Assignment of Message Waiting — Automatic is on a per-extension basis. Assignment of a messaging service and a message-storage destination is on a per-voice terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can fully administer this feature using the SMT (System Management Terminal). However, since the Audible Message Waiting service cannot be administered with TCM, the TCM feature can partially administer this feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

Message Waiting — Automatic can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
MESSAGE WAITING — AUTOMATIC

PROCEDURE	WORD	PURPOSE	SMT
000	2	Specifies the destination (switch, AUDIX adjunct, or AP) for LWC message storage for a voice terminal and designates Call Coverage message retrieval permission.	Yes
000	3	Assigns audible automatic message waiting to an extension number.	Yes
063	1	Assigns an automatic message waiting lamp to a voice terminal.	Yes
063	2	Displays Automatic Message Waiting assignments.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — MESSAGE WAITING — AUTOMATIC

PATH NAME	PURPOSE
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change terminal buttons	This screen is also used to assign the Automatic Message Waiting lamp to a voice terminal.

Message Waiting — Manual

Description

This feature enables a multiappearance voice terminal user to light the green status lamp at another preassigned multiappearance voice terminal by pressing a manual message waiting button. The status lamp lights at both terminals and can be turned off by either voice terminal.

The Message Waiting feature normally indicates a need to contact the activating party. The status lamp could also indicate to a secretary DO NOT DISTURB or NOT AVAILABLE while lighted. In these cases, the user should relabel the lamps.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Turn On the Message Waiting Lamp From a Controlling Voice Terminal:

1. Be sure the green status lamp is not lit.
2. Press **[MSG WAIT]** . [The green status lamp lights at both voice terminals.]

To Turn Off the Message Waiting Lamp From the Controlling or Signaled Voice Terminal:

1. Be sure the green status lamp is lit.
2. Press **[MSG WAIT]** . [The green status lamp goes out at both voice terminals.]

Considerations

Signaled Voice Terminal

A voice terminal can only be assigned as a signaled voice terminal once. It cannot receive a message waiting signal from more than one voice terminal.

MSG WAIT Button

The number of MSG WAIT buttons assigned to a voice terminal is limited only by the number of unassigned buttons.

Interactions With Other Features

None.

Hardware Requirements

Multiappearance voice terminals are required for this feature.

Feature Administration

Assignment of the Message Waiting feature is on a voice terminal basis for multiappearance voice terminals.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE — MESSAGE WAITING — MANUAL			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
053	2	Administers buttons for the controlling and signaled voice terminals.	Yes

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — MESSAGE WAITING — MANUAL	
PATH NAME	**PURPOSE**
terminal-change terminal buttons	Assigns the buttons for the Manual Message Waiting feature.

Modem Pooling

Description

Modem Pooling provides a "pool" of conversion resources that is available as needed to convert analog encoded data signals to the DCP (Digital Communications Protocol) format. Modem Pooling eliminates the need for a dedicated modem for every data terminal that requires access to analog facilities. This provides an economy of scale by allowing occasional modem users to share these resources.

Modem Pooling also allows off-premise users to access digital-interfaced facilities, such as host computers, over analog trunks using the same computer ports used by local users. Without Modem Pooling, a separate set of analog interfaced computer ports would be required for off-premise users that do not have access to digital trunk connections.

Feature History and Development

The Modem Pooling feature was first available on System 85 in Release 1. This feature has remained virtually unchanged until DEFINITY Communications System Generic 2.

Generic 2 enhancements include the following:

- New conversion resource insertion algorithm based on bearer capability
- Hayes Smart Modem conversion resources available
- On-Premises Modem Pooling support.

The Modem Pooling Requirement

Modems convert digital signals to analog signals and vice versa. Since both the System 85 and Generic 2 are digital switches, most of their data features use either DCP or the ISDN Integrated Services Digital Network) digital protocol. Many network facilities, both public and private are still analog; therefore, interpremises data communications frequently require a modem to perform the conversion between the analog and digital formats.

The System 85 and Generic 2 switches use data modules to provide protocol conversion between DCP or ISDN and the RS-232 signaling protocol used by most modems and data terminals. Modem Pooling pairs modems and data modules to form conversion resources.

Conversion Resources

A Modem Pooling conversion resource is a modem and a data module permanently paid to provide protocol conversion between an analog format data signal and the DCP data signal used by the switch.

> **NOTE:** Modem Pools convert between DCP and analog only. When necessary, the Interworking function *(see* Section 63, ISDN—PRI [Primary Rate Interface], and Appendix G, Integrated Services Digital Network) handles any conversion needed between DCP and ISDN.

With the Modem Pooling feature, conversion resources are grouped into pools for ***switched access*** when required. Since conversion resources are not permanently attached to the data terminals, the data terminals can also access digital facilities where a conversion resource is not needed. ***Switched access*** makes modem pooling virtually transparent to the user. When conversion is required, the switch automatically inserts the conversion resource into the call path.

Modem Pool Characteristics

Each modem pool is established to support a specific set or range of parameters. Modem pools can be set up to support synchronous or asynchronous, full- or half-duplex transmissions. Frequently, a modem pool will support only one data rate. However, modems are available that can respond to multiple data rates automatically (autobaud). With the introduction of the 7400A Data Module, Modem Pools can be set up to provide Hayes "Smart Modem" compatible support.

Modem pools use paired trunk groups (one digital and one analog). Within these trunk group pairs, individual trunk pairs are connected by a conversion resource. Each pair of trunk groups can support up to 99 conversion resources. The Route Advance feature can be used to access up to five trunk groups, for a maximum of 495 conversion resources, before the call must enter queue or fail for lack of a conversion resource.

Supported Calls

The Modem Pooling feature supports connections between analog trunks (for example, DID, ETN, WATS, FX and CO) and a local digital data endpoint. Modem Pooling also supports calls between digital trunks (such as, DS1 or ISDN) and local analog data endpoints and analog encoded data calls over digital trunks (for example, voice grade DS1) to digital endpoints.

Outgoing Calls

A typical outgoing call situation that requires Modem Pooling support is a local DCP (or ISDN) data terminal calling a distant host computer over an analog trunk Figure 85-1 shows such a typical outgoing call connection.

Incoming Calls

Incoming calls (from off-premises terminals) are also supported by the Modem Pooling feature. An incoming data call over an analog trunk to a *DCP* or *BRI* (Basic Rate Interface) data endpoint is recognized by the switch as needing a conversion resource. The switch reserves a conversion resource and then places it in the call path when the connection is completed. For System 85, Release 2 switches, the switch identifies the need for a Modem Pooling conversion resource based on translated characteristics of the incoming trunk group and the addressed end point.

For Generic 2 switches, the Bearer Capability feature provides an improved source of information for this decision. For ISDN calls, the Bearer Capability (BC) IE in the ***call setup message*** provides the needed information. For other types of calls, the BCCOS (Bearer Capability Class of Service) of the facilities involved provides the needed information (see the BCCOS feature).

Voice to Data Transfers for DCP Terminals

If the incoming call is answered at a DCP voice terminal and transferred to its associated data module using ***one button transfer*** (see the **Data Call Setup** feature), the conversion resource is inserted when the data transfer takes place.

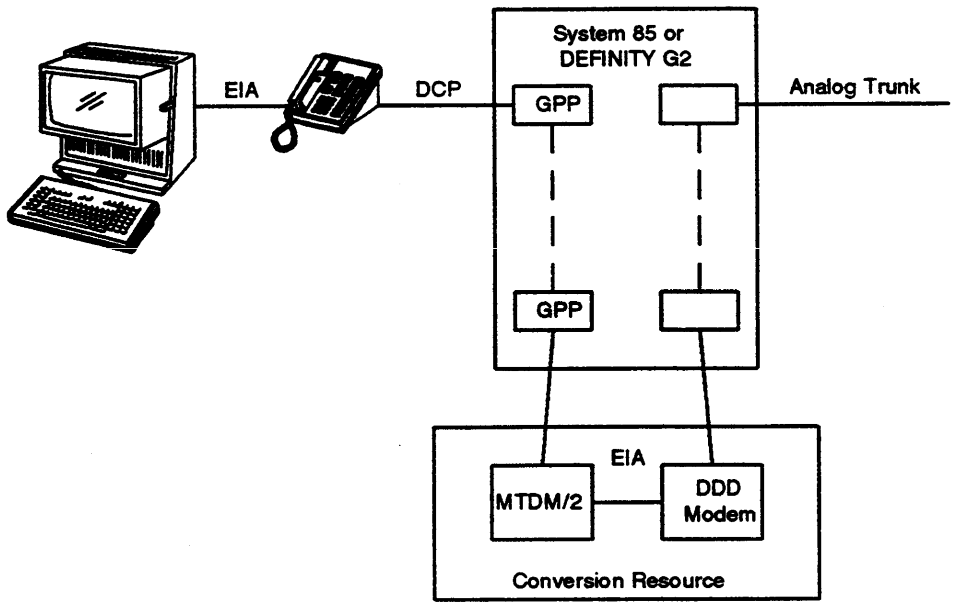

Figure 85-1. Modem Pooling Arrangement

Voice Calls to BRI Terminals

Incoming calls placed to BRI terminals work differently. The ISDN—BRI voice terminal does not have the ability to transfer an incoming call to the associated data extension ***(one button transfer*** is not available to BRI terminals). An incoming voice call placed to a BRI terminal must be placed to the data extension if it is to be converted to a data call. In this case, the switch recognizes the call as data because of the BCCOS of the extension addressed, and inserts a conversion resource. If the incoming call is placed to a voice only or to a voice / data extension number and answered by the voice terminal, the call cannot then be transferred to the data terminal. The caller must hang up and place the call again to establish a data terminal connection and receive Modem Pooling support. Note that if the BRI data terminal is served by a data appearance (a capability provided to BRI stations in Generic 2.1, 3.0) a voice call will never terminate to a specified data appearance or data prime line.

On-Premises Calls

On System 85, Release 2, Version 4 and earlier switches, Modem Pooling is not used for local calls (calls that both originate and terminate on the same switch). However, a local call that appears to the switch as an incoming trunk call can receive Modem Pooling support. For example, an on-premises user could dial off-premises (DOD) to a number provided for use by off-premises (DID) data callers. This call would receive Modem Pooling support. This practice is not recommended for general use since it places a heavy demand on CO trunks (two trunks are used by each call). In the long term, it would be more economical to convert analog data stations to digital or to provide analog computer ports *(see* the Data Communications Access feature).

With the Generic 2 switches, modem pooling support can be provided for local (on-premise) calls where one end of the call is administered as a line and the other as a trunk. Calls from local analog data terminals to digital HCA (Host Computer Access) ports or from local DCP data terminals to analog DCA (Data Communications Access) ports would receive modem pooling support. Figure 85-2 shows one type of connection that was not supported in System 85, Release 2, Version 4 and earlier, but is supported in Generic 2.

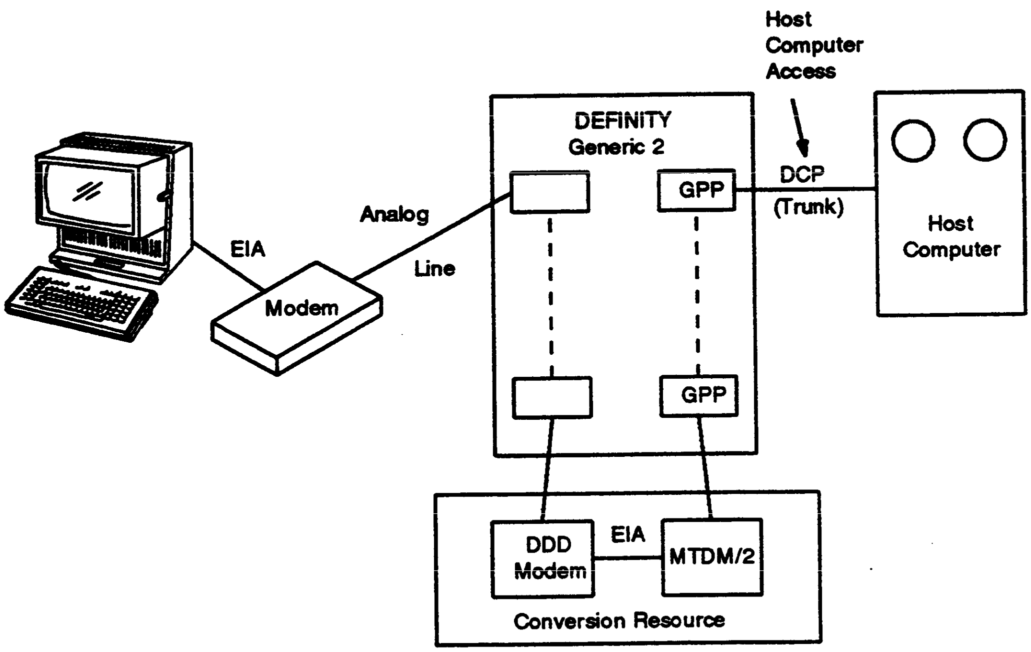

Figure 85-2. Local Modem Pooling Support in Generic 2

Host Computer Terminating Calls

Modem Pooling support is provided for local calling requirements when one end point is administered as a trunk and the other as a line. For example, an analog interfaced data terminal (line) can place a call to a HCA (Host Computer Access) port (DCP trunk) and a Modem Pooling conversion resource will be provided. Also, a DCP interfaced data terminal (line) can place a call to a DCA (Data Communications Access) port (analog trunk) and a Modem Pooling conversion resource will be provided.

Attendant-Extended Calls

Modem Pooling support is not provided for attendant-extended calls.

For ***System 85, Release 2, Version 4 and earlier switches,*** data calls that require both attendant assistance and Modem Pooling support can be extended to a DCP voice terminal with a data button assigned *(see* the Data Call Setup feature). The voice terminal user can then extend the call to the desired DCP data endpoint using ***one button transfer.***

For ***Generic 2 switches,*** attendant-seeking data calls for DCP or analog end points (HCA or DCA) can be handled in the same way as in System 85, Release 2, Version 4 and earlier switches. However, data calls to BRI data end points work differently. If a data call intended for a BRI data end point comes in on an attendant-seeking trunk *(See* the Multiple Listed Directory Numbers feature), the only way this call can be handled is to give the calling party (assuming the call comes in as a voice call) the connect number to dial the BRI data end port directly, and tell them to try again. Attendants cannot extend calls to BRI data end points.

Data Preindication

System 85, Release 2, Version 4 and Earlier Switches

Data preindication (Data Call Setup feature) reserves a modem pool member (conversion resource) when needed. For data calls requiring the use of Modem Pooling, ***one buttton transfer*** with ***data preindication*** or data terminal (keyboard) dialing techniques are recommended. Failure to preindicate can result in call failure after the connection is made if the needed conversion resource is not available. If, however, the conversion resource is available, the call will succeed even if data preindication is not used.

DEFINITY Communications System Generic 2 Switches

Using DCP (Digital Communications Protocol) Terminals

When *Voice Terminal Data Call Setup* is used with a DCP voice terminal, data preindication is required in Generic 2. The bearer capability of the call is the determining factor in whether or not a Modem Pooling conversion resource is inserted into the call path. DCP Voice terminals and data terminals have different extension numbers and distinctly different BCCOSs *(see* the **Bearer Capability** feature). If data preindication is not used, the BCCOS of the voice terminal is used in setting up the call rather than the BCCOS of the data terminal. If a Modem Pooling conversion resource is needed, it ***will not be inserted*** even if one is available. This is because the bearer capability indicates a voice call that would not have needed a conversion resource.

Data preindication is also required when placing a voice terminal data call setup call to an ISDN—BRI voice / data station, even though Modem Pooling is not required. Again, without data preindication the BCCOS of the voice terminal is used to set up the call and the BRI voice terminal, rather than the data terminial will receive the call. BRI voice terminals do not have one ***button transfer*** capabilities and cannot transfer the call to the associated data terminal.

Using ISDN—BRI Basic Rate Interface) Terminals

Data Preindication is not possible from an ISDN—BRI voice terminal. The BRI generates a Bearer Capability IE based on the way the call is initiated (either from the data terminal or using the Data / Send / Off button on the voice terminal). For ISDN calls, Modem Pooling action by the switch is based entirely on the Bearer Capability IE of the call.

User Operations

Generally, Modem Pooling requires no special user operation unless autobaud modems are used. Conversion resources are added to the connection automatically when needed and when applicable. The following user operations apply to the special situation where user intervention is required.

From DCP Terminals

For outgoing (off-premises) calls using Voice Terminal Data Call Setup from a DCP voice terminal, ***a conversion resource must be reserved*** (using data preindication) before the data call is made.

To reserve a conversion resource from a DCP voice terminal:

1. Go off-hook on the associated voice terminal. (Dial tone)
2. Press the **[DATA]** button. (The green LED associated with the DATA button flutters, indicating a conversion resource is reserved.)
3. Enter the dial access code for the desired trunk group,

   or

   Dial the network access code. (Second dial tone.)
4. Dial the 7- or 10-digit number for the off-premise destination. (Call progress tone, ringing tone.)
5. Press the **[DATA]** button again. (Call control is transferred to the data terminal. Call progress messages, RINGING, ANSWERED, appear on the data terminal.)

Considerations

Hard and Soft Processor Swaps

Stable calls using a Modem Pooling conversion resource endure a hard processor swap. However, calls cannot be placed through a conversion resource during a hard processor swap.

Hayes "Smart Modem" Modem Pools

The Hayes "Smart Modem"command mode of operations is popular with many data users, especially PC users. While Hayes users have always been supported by Modem Pooling much of the functionality was lost due to the inability of conversion resources to respond to command queries.

With the introduction of the 7400 series of data modules, end-to-end "Smart Modem" connections are now possible. This includes the Modem Pooling conversion resource if required. The 7400A Data Module can be used with a Hayes type modem to form a Hayes "Smart Modem" conversion resource. **Note** that the 7400B Data Module cannot be used for this purpose.

With a "Smart Modem" Modem Pooling conversion resource (and Smart Modem termination points), Hayes compatible end-to-end connections can be established, even if Modem Pooling support is required for the call. However, where a variety of modem pools are available (that is, conversion resources with other than Smart Modem components are also available on the switch), there is no way to assure that a Hayes Smart Modem compatible Modem Pooling conversion resource will be used for any given call.

While the 7400A data module was first introduced in the Generic 2 time frame, it can be used for Modem Pooling on any System 85, Release 2 switch.

Tandem Switched

Modem Pooling support ***is not provided*** at any tandem switch. That is, Modem Pooling support is provided only at the switch where a call originates and at the terminating switch. Any intervening switch that the call may tandem through will not insert a conversion resource.

Tie Trunk Calls

Dial repeating and automatic tie trunks can be used in interpremise connections requiring modem pooling (data calls). Both voice terminal and keyboard dialing (data terminal) users can access dial repeating tie trunks.

When an *automatic tie trunk* is used, the attendant is automatically involved, unless the trunk is set up as a Remote Access trunk Attendant-extended calls are not provided Modem Pooling support; therefore, only voice terminal users can use normal automatic tie trunks for data calls. Data terminal users (keyboard dialing) can use automatic tie trunks only if they are setup as Remote Access trunks.

Traditional Module Required

The Generic 2.1 Issue 1.0 switch required that there be at least one traditional module for the Modem Pooling feature. Modem Pooling requires that an ADFTC (Analog/Digital Facilities Test Circuit) or equivalent be present on the system. Initially, there was no equivalent circuit for the Universal Module.

With Generic 2.1 Issue 2.0, the TN771B MTCP (Maintenance Test Circuit Pack) provides an equivalent circuit to the ADFTC for the universal module. The TN771B can be retrofitted to earlier Generic 2 switch only if the common carrier universal bus interface is also upgraded to the UN154B.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

AAR (Automatic Alternate Routing)

On System 85 and DEFINITY Generic 2.1 switches, the Modem Pooling feature is fully compatible with the AAR feature. The specific way in which these features interact, however, changes between System 85, Release 2, Version 4 and Generic 2.

System 85, Release 2, Version 4 and Earlier

For System 85, Release 2, Version 4 and earlier switches, the Modem Pooling selection process is separate and distinct from the AAR route selection process. A Modem Pooling conversion resource is selected based on switch translation for the end points of the call, either before AAR route selection (if the call was preindicated) or after route selection.

DEFINITY Communications System Generic 2

For Generic 2 switches, the Modem Pooling selection process is part of the AAR or WCR route selection process and takes place concurrently, based on call setup message requirements (for ISDN calls) or BCCOS. The AAR feature is available on System 85 and Generic 2.1 switches. AAR is replaced by the WCR feature on Generic 2.2 features. The search algorithm uses the following steps:

1. After selecting the appropriate pattern, the AAR search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (such as, Mode 2 data, 1200 bps, restricted channel, etc.). If a match is found and a trunk is available, other factors such as FRL permitting, the search ends and the action taken is "circuit switch the call."

2. If a match with an available trunk is not found, the algorithm attempts to connect the call to a preference for which the action to take is not "block the call." With currently available options, this would be a preference where the action to take is "insert ***modem pool."***

   The "action to take" function is part of the new feature, Bearer Capability, and is discussed in more detail under that feature.

ARS (Alternate Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, the Modem Pooling feature is fully compatible with the ARS feature. The specific way in with these features interact, however, changes between System 85, Release 2, Version 4 and Generic 2.

System 85, Release 2, Version 4 and Earlier

For System 85, Release 2, Version 4 and earlier switches, the Modem Pooling selection process is separate and distinct from the ARS route selection process. A Modem Pooling conversion resource is selected based on switch translation for the end points of the call, either before ARS route selection (if the call was preindicated) or after route selection.

DEFINITY Communications System Generic 2

For Generic 2.1 switches, the Modem Pooling selection process is part of the ARS route selection process and takes place concurrently, based on call setup message requirements (for ISDN calls) or BCCOS. On Generic 2.2 switches, the ARS feature is replaces by the WCR feature.

1. The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (such as, Mode 2 data, 1200 bps, restricted channel, etc.). If a match is found and a trunk is available, other factors such as FRL permitting the action taken is "circuit switch the call."
2. If a match is not found the algorithm attempts to connect the call to a preference for which the action to take is not "block the call." With currently available options, this would be a preference where the specified action is "insert ***modem pool."***

Bearer Capability

The Bearer Capability feature has a dramatic effect on the operations of the Modem Pooling feature. On System 85, R2 V4 and earlier switches the Bearer Capability feature is not available. On these switches, Modem Pooling applies only to off-premise calls. Also, for outgoing calls, the reservation of a Modem Pooling conversion resource is a separate software action (either before or after call routing).

On the Generic 2 switch local Modem Pooling support can be provided for certain on-premise calling situations (Host Access and Data Communications Access). For outgoing calls, reservation of a Modem Pooling conversion resource is made as part of the call routing process *(see* the AAR, ARS, and WCR feature interaction).

Each extension, trunk group, and routing pattern is assigned a BCCOS. For data calls, switch actions are prescribed (in switch translations) for each possible combination of one BCCOS calling another BCCOS. One of the prescribed switch actions is "insert ***modem pool."*** In Generic 2, the use of Modem Pooling is prescribed for each calling situation by the switch actions called for by the Bearer Capability feature. These switch actions are user administerable.

Call Vectoring

The "route to" step of the Call Vectoring feature is compatible with the Modem Pooling feature. Whenever a conversion resource is needed to complete the call to a "route to" step’s destination, these conversion resources will be inserted.

Data Call Setup

System 85, Release 2, Version 4 and Earlier

The Modem Pooling feature is designed specifically to support the Data Call Setup feature by providing switched conversion resources between analog and digital data communications links. On System 85, Release 2, Version 4 and earlier switches, this support is provided only for off-premise calling situations (either incoming or outgoing calls).

DEFINITY Communications System Generic 2

In Generic 2, Modem Pooling support is extended to include certain on-premise connections such as terminal to local host computer.

Also, in Generic 2, the importance of ***data preindication*** for voice terminal data call setup using DCP voice terminals increases. Calls originated using DCP voice terminal data call setup will not receive Modem Pooling support unless data preindication is used.

Data Protection

Data Protection—Permanent should be assigned to all trunk groups associated with Modem Pooling to prevent any intrusions attempted by bridge-on feature users.

ISDN—BRI (Basic Rate Interface)

The Modem Pooling feature supports data calls to and from ISDN—BRI terminals as required. The only real difference in the way Modem Pooling functions with ISDN—BRI data calls as opposed to DCP data calls is in the way the switch determines the need for a conversion resource. With DCP data calls, need is determined from translated characteristics of the facilities involved (assigned BCCOS) or from the use of Data Preindication With ISDN—BRI, need is determined from information contained in the all setup message.

ISDN—BRI components (data modules) cannot be used to form a Modem Pooling conversion resource.

ISN (Information Systems Network) Interface

Modem Pooling provides 2-way connectivity between ISN endpoints and remote stations via either public or private network trunks. For example, a remote data station user can dial the number assigned to the ISN trunk group, receive the ISN dial prompt, and dial the ISN destination code. An ISN station user can also dial through to the remote data station. Use of Modem Pooling requires that extension number steering be available to access ISN Interface trunks.

ISDN—PRI (Primary Rate Interface)

The Modem Pooling feature works the same for calls using ISDN—PRI facilities as it does for other types of data calls. The switch at each end of the call must determine the need for a modem pooling conversion resource. This determination is made differently in System 85, Release 2, Version 4 switches than in Generic 2 switches.

System 85, Release 2, Version 4 Switches

A problem may occur if an analog interfaced data station places an ISDN call to a digital interface data end point on a remote switch. Because the call is coming in on an AVD type DS1 trunk facility, the receiving switch may not be able to recognize that a modem pooling conversion resource is needed.

Another problem could occur if data stations use the "ISDN Facilities Preferred" option in their class of service. If such calls start out on ISDN facilities and then are transferred to analog facilities at a tandem node, it will be too late to insert a modem pooling conversion resource at the originating switch.

DEFINITY Communications System Generic 2 Switches

In Generic 2, some of the signaling characteristics for PRI trunks change. The AVD bit is no longer used; instead, all data call characteristics are carried in the Bearer Capability Information Element of the call setup message. With a BCCOS (Bearer Capability Class of Service) assigned to all lines, trunks, and network routing preferences, this information can be provided even for analog originated data calls. "Unknown" type calls received over PRI trunks are evaluated (voice or data) basal on the BCCOS of the addressed extension.

Main/Satellite/Tributary

Data calls placed through main / satellite arrangements over analog tie trunks require Modem Pooling connections at each end of the call where digital facilities are used. A data call to or from a DCP-interfaced endpoint at a main or satellite switch must be supported by Modem Pooling. Modem Pooling at the main cannot provide conversion for a data endpoint at a satellite location.

Queuing

When a data call is queued, the modem pool conversion resource is held waiting along with the queued call.

Tenant Services

The Modem Pooling feature is not partitioned. When a digital trunk group, an analog trunk group, and a conversion resource are provided, access to this analog-to-digital conversion is provided for every partition in the switch.

Trunk Verification—Voice Terminal

The Trunk Verification—Voice Terminal feature can be used to test Modem Pooling conversion resources on System 85, Release 2, Version 3 and later switches and on the Generic 2 switch. This cannot be done on earlier switches, and the Trunk Verification—Attendant feature cannot be used for this purpose.

When keyboard dialing is used to place a data call, the switch automatically reserves a Modem Pooling conversion resource and, if necessary, inserts the resource into the connection. As the switch reserves a resource, the digital Modem Pooling trunk-group number and the individual trunk number of the selected conversion resource are displayed in the RINGING call-progress message. In response to trouble reports, these values (after converting the trunk-group number to the trunk-group access code) can be used to verify the Modem Pooling conversion resource.

The Trunk Verification—Voice Terminal feature can also be used to seize a specific conversion resource, thereby reserving that resource for a call. This action overrides automatic action by the switch software.

WCR (World Class Routing)

The Modem Pooling feature works with the WCR feature in the same way it did with the earlier networking features, AAR and ARS. If a Modem Pooling conversion resource is needed for a call, it is reserved for that call and inserted into the connection when needed.

Restricting Feature Use

Any restrictions applicable to trunk groups apply to the trunk groups used for Modem Pooling. Attendant Control of Trunk Group Access, however, should be avoided as this would render the modem pool useless. Care must be taken to apply appropriate restrictions to each side of a trunk group pair.

Hardware Requirements

For Traditional Modules:

Connecting Circuits

- SN270, GPP (General Purpose Port) Circuit Pack

  The data module connects to the switch using a GPP (General Purpose Port) circuit administered as a trunk.

- SN243 Data Port Circuit Pack

  The modem connects to the switch via a DCA trunk circuit.

Supporting Circuitry

- SN255B Tone Detector Circuit*

  The Tone Detector circuit detects call-progress tones for off-premises calls when data terminal (keyboard) dialing is used. This circuit also provides identifying signals used by the switch to determine when to put in a conversion resource for outgoing trunk calls.

- SN261C ADFTC (Analog/Digital Facilities Test Circuit)

  The Analog/Digital Facility Test circuit supports both automatic (self tests) and demand testing on modem pool members. This circuit can only be installed on a traditional module.

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* If Modem Pooling is being added to an existing switch the tone detector circuit pack improbably not already present. Care must be taken to ensure that this circuits is also added or the feature will not work.

For Universal Modules:

Connecting Circuits

- TN754, Digital Line Circuit Pack

  The data module connects to the switch using the digital line circuit pack administered as a trunk.

- TN742, Analog Line Circuit Pack

  The modem connects to the switch via the analog line circuit pack.

Supporting Circuitry

- TN748C Tone Detector

  The Tone Detector circuit detects call-progress tones for off-premises calls when data terminal (keyboard) dialing is used. This circuit also provides identifying signals used by the switch to determine when to insert a conversion resource for outgoing trunk calls.

- TN771B MTCP (Maintenance Test Circuit Pack)

  The Maintenance Test Circuit Pack supports both automatic (self tests) and demand testing on modem pool members.

- UN154B Universal Bus Interface

  If the TN771B is used, the port carrier must be equipped with the UN154B bus intrerface. This is an updated version from the UN154 that was used with initial releases of the Universal Module. Early Generic 2 switches (prior to Generic 2.1, Issue 2.0) can use the TN771B MTCP only if the UN154 is replaced by the UN154B.

For Either Module Type:

The Modem Pooling feature requires the use of a specially configured conversion resource. This consists of the following:

- Data Module

  A 7400A data module, or MTDM/2 (Modular Trunk Data Module). (The 7400A is Hayes-compatible and D-link compatible and supports asynchronous data at speeds from 300 bps to 19.2 Kbps.)

  The data module has a switch that selects the modem pooling mode.

- RS-232 Connecting Cable

- DDD Modem

  DDD Modems are required to provide analog connectivity. Table 85-A lists AT&T modems that are supported for use with the Modem Pooling feature.

TABLE 85-A. AT&T Modems Supported For Modem Pooling

Current AT&T Modems			
Modem Model	**Data Rate**	**Duplex**	**Synchronization**
208BR	4800	Half	Synchronous
212AR	300 / 1200	Full	Asynchronous
2224G*	300 / 1200 / 2400	Full	Asynch / Synch
2248A	4800	Full	Asynchronous
2296A	4800 / 9600	Full	Asynchronous

*The 2224G Modem will function in the Asynchronous mode at all data rates, however, it functions the Synchronous mode only at 1200 and 2400 bps.

Other AT&T Modems			
Modem Model	**Data Rate**	**Duplex**	**Synchronization**
103JR	Up to 300	Full	Asynchronous
202SR	1200	Half	Asynchronous
201CR	2400	Half	Synchronous
2224A*	300 / 1200 / 2400	Full	Asynch / Synch

*The 2224A Modem will function in the Asynchronous mode at all data rates, however, it functions in the Synchronous mode only at 1200 and 2400 bps.

Conversion resources can be rack mounted. The data module and modem are connected through their RS-232 interface circuitry.

Feature Administration

Assignment of the Modem Pooling feature is on a per-system basis. Assignment within the system is made on a per-trunk group basis.

For System 85 switches, the Modem Pooling feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the FM (Facilities Management) feature.

On the DEFINITY Communications System Generic 2 switch, the Modem Pooling feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — MODEM POOLING		
PROCEDURE	WORD	PURPOSE
051	1	Assigns data terminal type and data module translations to modem pool members and ADFTC or MTCPs.
052	1	Assigns line type characteristics to ADFTC or MTCPs.
100	1	Assigns trunk types and dial access codes to trunk groups. The applicable trunk-type encodes are as follows: 100 Tone detector 101 Analog data modem pool 102 Digital data modem pool.
100	2	For System 85 switches: Assigns data transmission characteristics to a trunk group. For Generic 2 switches Assigns the BCCOS (Field 2) and Modem Pooling Characteristics.
180	1	For the analog and digital portions of a conversion resource (Modem Pooling pair), assigns the equipment location of each portion to that portion's trunk-group number.

The following are the applicable FM path names used with the AP 16.

FM SCREENS — MODEM POOLING	
Path Name	Purpose
facilities-mgmt data-switching attributes	Assigns the DAC and trunk-group number to a trunk group (both analog and digital). Assigns the equipment locations to analog and digital modems.
facilities-mgt data-switching trunk-attributes	Assigns the transmission characteristics to digital trunk groups (such as, data rate, full or half duplex, synchronous or asynchronous). Turns the Data Restriction feature on or off. Assigns the trunk type to each group. Assigns an overflow sequence. A routing table can be set up to associate a modem pool with a list of other pools to be used for the overflow.
facilities-mgt data-switching modem-pairs	Pair each digital port used in Modem Pooling with an analog port. Assigns the modem pool pair as 2-way originating only, or terminating only.

Notes:

Multiappearance Preselection and Preference

Description

This feature provides multiappearance voice terminal users with options for placing or answering calls on selected appearances. These options include:

- Preselection—Allows voice terminal users to ***manually*** select an appearance by pressing the appropriate button before going off-hook. All appearance preference options are overridden.
- Ringing Appearance Preference—Automatically selects a ringing appearance for voice terminal users upon going off-hook.
- Calling Appearance Preference*—Automatically selects an appearance with an incoming call whether that line is ringing or not. Calling Appearance Preference or Preselection can be used to answer calls that would otherwise go unanswered from an appearance on another voice terminal.
- Idle Appearance Preference†—Automatically connects voice terminal users to an idle appearance upon going off-hook.
- Prime Appearance Preference†—Automatically selects the terminal's primary appearance for a voice terminal user going off-hook. Prime Appearance Preference can be used when a particular appearance is preferred for toll or interfacility calling. This operation serves to improve the recognition of calling costs.
- No Appearance Preference‡—The voice terminal user must ***manually*** select an appearance to place or answer a call.
- Last Appearance Preference†—The last appearance used is selected automatically (actually remains in use) when going off-hook to place a call or when a call terminates on the line.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature. Each operation assumes that the particular option has been assigned for the voice terminal.

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* "Terminating" preference.

† "Originating" preference.

‡ "Terminating" and "originating" preference.

Preselection

To override the preference assigned and be connected to another appearance:

1. Verify that the selected appearance is idle. [The green status lamp of the selected appearance is dark.]
2. Press the idle appearance button. [The red status lamp lights.]
3. Go off-hook. [The green status lamp lights, dial tone is heard, and the user is connected to the appearance.]
4. When the connection is no longer desired, go on-hook. [The green status lamp goes out. The red status lamp may or may not go out.]

Ringing Appearance Preference

To be automatically connected to an incoming call that is ringing:

1. Verify that the voice terminal is ringing. [Ringing is heard, the green status lamp of the ringing appearance is flashing and the red status lamp is lit.]
2. Go off-hook. [Ringing stops, the green status lamp lights, and the user is connected with the calling party.]
3. When the connection is no longer desired, go on-hook. [The green status lamp goes out. The red status lamp may or may not go out.]

Calling Appearance Preference

To connect to an incoming call which may or may not be ringing:

1. Verify that an appearance has an incoming call present. [Ringing may or may not be heard, the green status lamp is flashing, and the red status lamp is lit.]
2. Go off-hook. [If the voice terminal was ringing, ringing stops, the green status lamp lights, and the user is connected with the calling party.]
3. When the connection is no longer desired, go on-hook. [The green status lamp goes out. The red status lamp may or may not go out.]

Idle Appearance Preference

To be automatically connected to an idle appearance upon going off-hook:

1. Verify that at least one appearance is idle. [At least one appearance has a green status lamp that is dark.]
2. Go off-hook. [The red and green status lamps of the appearance selected light, dial tone is heard, and the user is connected to the idle appearance.]
3. When the connection is no longer desired, go on-hook. [The red and green status lamps go out.]

Prime Appearance Preference

To be automatically connected to the prime appearance:

1. Verify that the prime appearance assigned is not busy. [The red status lamp is lit and pointing to the prime appearance. The green status lamp of the prime appearance is dark.]
2. Go off-hook. [The green status lamp lights, dial tone is heard, and the user is connected to the prime appearance.]
3. When the connection is no longer desired, go on-hook. [The green status lamp goes out; red status lamp remains lit.]

No Appearance Preference

This feature does not automatically connect the voice terminal user to an appearance. The appearance must be selected manually using the Preselection function.

Last Appearance Preference

To be automatically connected to the last appearance used:

1. Verify that the last appearance used is not busy. [The red status lamp is lit and pointing to the last appearance used. The green status lamp of the last appearance used is dark.] (If the appearance is busy and the user goes off-hook, the user is bridged onto the 2-party connection. This occurs unless Manual Exclusion is active or unless another party is already bridged onto the connection.)
2. Go off-hook. [The green status lamp lights, dial tone is heard, and the user is connected to the last appearance used.]
3. When the connection is no longer desired, go on-hook. [The green status lamp goes out.]

Considerations

Reorder Tone and Idle Appearance Preference

For Idle Appearance Preference, the switch returns reorder tone whenever the user goes off-hook while every appearance is active.

Status of Red Lamp

For Preselection, Ringing Appearance Preference, and Calling Appearance Preference, the original appearance preference is restored when going on-hook. The status of the red lamp depends on what features have been assigned.

- If No Appearance Preference is assigned, the red status lamp goes out.
- If Idle Appearance Preference is assigned, the red status lamp goes out.
- If Last Appearance Preference is assigned, the red status lamp stays lit.
- If Prime Appearance Preference is assigned, the red status lamp of the prime appearance lights and the red status lamp of the selected lamp goes out.

Interactions With Other Features

The following System 85 and DEFINITY Generic *2* features affect or are affected by the operation of this feature.

Hold

Interactions between the Hold feature and the Multiappearance Preselection and Preference feature are as follows:

- If the HOLD button on the terminal is pressed while the terminal is on-hook, any appearance preference for call origination (Last Appearance Preference, Prime Appearance Preference, or Manual Preselection) is lost. The user, upon going off-hook, does not connect to any appearance (unless Idle Appearance Preference is assigned).
- If the Prime Appearance is placed on hold and the controlling terminal user goes on-hook, the I-use lamp is relit on the Prime Appearance button. If the controlling terminal user then goes off-hook, the terminal connects back to the Prime Appearance and that call is removed from hold.
- If the HOLD button is pressed while the terminal is in the on-hook ringing state and the Ringing Appearance Preference is active, the Ringing Appearance Preference feature becomes inoperative. That is, when the user goes off-hook, no appearance is selected (unless Idle Appearance Preference is assigned).

Intercom—Automatic, Dial, and Manual

At the originating voice terminal, the Multiappearance Preselection and Preference feature cannot automatically select an intercom appearance. However, at the receiving voice terminal, the Multiappearance Preselection and Preference feature can select the intercom appearance receiving the call.

Personal Central Office Line

The Multiappearance Preselection and Preference feature does not automatically select Personal Central Office Line appearances.

Hardware Requirements

A multiappearance voice terminal is required for this feature.

Feature Administration

Assignment of the Multiappearance Preselection and Preference feature is on a per terminal (equipment line location) basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
MULTIAPPEARANCE PRESELECTION AND PREFERENCE

PROCEDURE	WORD	PURPOSE	SMT
051	1	Specifies the desired type of originating and terminating preference for a voice terminal.	Yes
052	1	Assigns prime appearance (prime line) to an extension number of a voice terminal.	Yes

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — MULTIAPPEARANCE PRESELECTION AND PREFERENCE

PATH NAME	PURPOSE
terminal-change terminal equipment	Specifies the desired type of appearance preference for a voice terminal and assigns prime appearance (prime line) to a voice terminal.

Notes:

Multiple Listed Directory Numbers

Description

The Multiple Listed Directory Numbers feature helps the attendant identify the purpose of incoming public network calls. It does this through the ICI (Incoming Call Identification) display. With only a single LDN (Listed Directory Number), the ICI display shows only that the call is an LDN call. This situation does not provide much feedback for the attendant of a large, diversified System 85 or DEFINITY Generic 2. Using multiple LDNs, the attendant can then determine something about the purpose of an LDN call before the call is even answered.

Types of LDNs:

There are two types of LDNs. Each type uses a different type of CO (Central Office) trunk.

- DID (Direct Inward Dialing) LDN — This type of LDN uses a DID trunk group from the CO to the System 85 or DEFINITY Generic 2. The CO sends the dialed digits to the switch for determination of routing. Through this type of routing the trunk group can be shared by more than one LDN as well as by calls bound for other destinations. (The Multiple LDN feature provides as many as 999 DID LDNs, each with a unique ICI display.)

  These DID LDNs can also be dialed by System 85 or DEFINITY Generic 2 voice terminal users. These users need only dial the extension number digits to access an attendant.

- Non-DID LDN — This type of LDN uses a non-DID (such as "attendant completing") trunk group. Calls from over these trunk groups are routed directly to the attendant queue. No digits are passed from the CO to the switch. Thus, the trunk has no other use. This is like a private line at a person's home. (The number of non-DID LDNs is limited by the number of trunk groups that can be dedicated for this use.)

The decision to use DID verses non-DID LDNs is based on the present switch configuration and/or on the number of LDNs desired. If DID service is to be provided, LDN traffic should be engineered into the DID plan. If not, the number of LDNs to be provided may or may not warrant the use of DID service. There is a point at which DID service, that may require only a single DID trunk, becomes more economical than providing one non-DID trunk for each LDN.

Application Example

Figure 87-1 shows an example of how multiple LDNs might be used. In this example, the owner of an automobile dealership has divided the organization into two separate divisions; one for cars and one for trucks. There are also finance and parts divisions that support both the car and truck divisions. The car division and the truck division each have their own LDN. Since no distinction is made between car financing and truck financing, there is no need for two separate financing LDNs. The same is true for parts. These two divisions, along with other calls, can share a third LDN. In this example, calls using 456-2400 cause the ICI to display "CARS." If the caller asks for the service department, the attendant knows from the ICI that the caller wants car service. Calls using 456-2500 display "TRUC" on the ICI. When the ICI displays "LDN" (general information), the attendant knows the call is of a general nature.

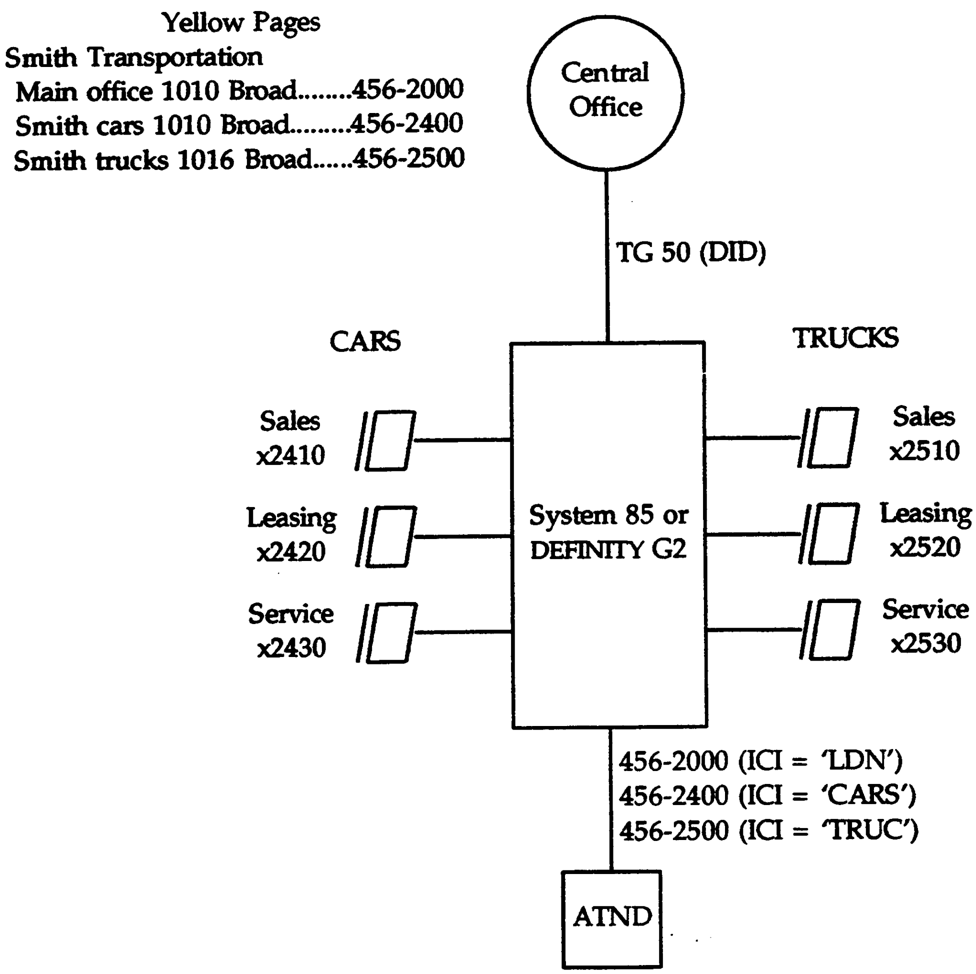

Figure 87-1. Sample Use of Multiple LDNs

Feature History and Development

This feature was first available on System 85 in Release 1. For Release 2, Version 3, the maximum number of DID LDNs was increased from four to nine. The DID LDNs were also made accessible by System 85 voice terminal users. A separate attendant access code is no longer required.

For Release 2, Version 4, the maximum number of DID LDNs was increased from 9 to 999. This limit of 999 LDNs is intended to provide enough capacity for partitioned switches. However, the full capacity of 999 DID LDNs is also available to switches that are not partitioned.

User Operations

None.

Considerations

Billing Provisions

For billing purposes, one listed directory number must be assigned to the attendant(s). When complete billing information is unavailable, the call is billed to this number.

Multiple Attendant LDNs

Each non-DID LDN must use a separate trunk group to provide a unique ICI display.

Published Numbers and Malicious Call Trace

For LDN applications of Malicious Call Trace, malicious calls to LDNs are usually placed by someone outside the organization. These people are likely to place malicious calls to numbers found in the public telephone directory. When this is the case, attendants answering calls directed to published numbers should be equipped with a feature button for convenient activation of Malicious Call Trace.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Call Vectoring

A vector directory number cannot be assigned as a system LDN in Procedure 204, Word 1. When this is attempted, an administration error will occur.

In order to route public network calls to the attendant queue with a VDN, the vector must contain a "route to" command that directs these calls to the attendant queue. However, vector treatment of calls within the attendant queue is not available. Vector steps (such as, recorded announcements, conditional go to steps, forced busy, and forced disconnect) can be applied to these calls ***before they enter the queue.*** However, once a call enters the attendant queue, vector processing stops.

Data Call Setup

Attendant seeking incoming calls cannot connect directly to a DCP (Digital Communications Protocol) interfaced data end point. To complete such a call (requiring a conversion resource), the attendant must extend the call to a multiappearance voice terminal that has a Data button assigned for the desired data end point. The multiappearance voice terminal can then transfer the call to the data end point using 1-button transfer procedures.

Dial Access to Attendant

Beginning with R2 V3 System 85, a voice terminal user can call the attendant group by dialing a DID LDN.

Extension Number (Multidigit) Steering

Normally, incoming exchange network calls to the System 85 or DEFINITY Generic 2, via the assigned local listed directory number, connect to either a local attendant or a Centralize Attendant Service attendant. With Extension Number Steering, it is possible for Direct Inward Dialing trunks to provide access to both types of attendants through the use of different telephone numbers. Each number is published (as Listed Directory Numbers), but the switch treats them differently. One number is treated as a normal LDN which is routed to the attendant. In this case, the call routes to the CAS attendant. The other number is treated as a DID LDN call. But instead of routing the call to the CAS attendant, the digits are assigned to Extension Number Steering for redirection to a local attendant (for example, CAS special service attendant).

Foreign Exchange Access

The Multiple Listed Directory Number feature can be used to direct incoming FX Access calls to the attendant queue.

Intercept Treatment

The Attendant Diversion to Recorded Announcement function overrides the Multiple Listed Directory Numbers feature. When an attendant activates Attendant Diversion to Recorded Announcement, a public-network caller cannot reach the attendant queue. Instead, these calls are diverted to the recorded announcement.

Main/Satellite/Tributary

If a Main / Satellite complex uses DAC (dial access code) steering, the LDN (Listed Directory Number) at the main should not begin with the same first digit that is assigned to a Main / Satellite a group. Furthermore, LDNs should not begin with a digit assigned to a feature dial access code. The exception is the attendant access code. The leading digit(s) of an LDN can be the same as the attendant access code. Likewise, an LDN can have the same leading digit(s) as local extension numbers.

Power Failure Transfer

During a power failure, a switch that uses incoming (or 2-way) CO trunks for non-DID LDN (Listed Directory Number) service can transfer these LDN calls to predesignated voice terminals.

Remote Access

When the switch is in the day mode and Remote Access has shared service with non-DID LDNs, users dialing the remote access number are routed to the attendant queue. The calls are then extended by an attendant rather than using the Remote Access feature.

When the switch is in the unattended console mode or the Remote Access is not shared with non-DID LDN service, a Remote Access user can dial an assigned LDN to reach the attendant group or the night terminal.

WATS Access

The Multiple LDN feature can be used to direct incoming 800 Service Access calls to the attendant queue.

Hardware Requirements

The Multiple Listed Directory Number feature requires the following additional or special hardware.

For Traditional Modules:

- Additional trunk circuits may be required.
  - Non-DID trunk circuits of an SN230 circuit pack (four circuits per SN230)
  - DID trunk circuits of an SN232 circuit pack (four circuits per SN232).

For Universal Modules:

- Additional trunk circuits may be required.
  - Non-DID trunk circuits of a TN747B circuit pack (eight circuits per TN747B)
  - DID trunk circuits of a TN753 circuit pack (eight circuits per TN753).

Regardless of the Module Type:

- A LORAIN* voice switched gain amplifier may be necessary to achieve sufficient transmission gain for non-DID connections.

Feature Administration

Assignment of the Multiple Listed Directory Numbers feature is on a system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel).

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* Trademark of Lorain Telephone Electronics.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

SYSTEM 85 RELEASE 2, VERSION 3 AND LATER ADMINISTRATION PROCEDURES — MULTIPLE LISTED DIRECTORY NUMBERS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
204	1	Administers the system's LDNs and designates the alphanumeric ICI display for each LDN.	No
275	2	Enables trunk-group sharing between the Remote Access and Multiple LDN features.	Yes

A DID LDN is administered by assigning the desired alphanumeric display and the LDN extension number to a call type (from LDN1 to LDN999) in Procedure 204, Word 1.

A non-DID LDN is administered by assigning the desired alphanumeric display to the trunk-group number of an attendant-completing trunk group.

The following are the applicable administration procedures for Release 2, Version 2, and earlier switches.

SYSTEM 85 RELEASE 2, VERSION 2 AND EARLIER ADMINISTRATION PROCEDURES — MULTIPLE LISTED DIRECTORY NUMBERS		
PROCEDURE	**WORD**	**PURPOSE**
204	1	Designates the alphanumeric ICI display for each LDN.
352	1	Administers the system's DID LDNs.

Music-on-Hold Access

Description

This feature provides music to a held party. Music assures the held party that the connection is still in effect. A nontalking or no-tone connection could cause the calling party to question if the connection is still in effect.

Feature History and Development

This feature was first available for System 85 in Release 1. For software packages up to R2 V3, Issue 1.3, only one music source can be provided per switch. Beginning with R2 V3, Issue 1.4 and R2 V4, the capability of providing one music source per module (up to 31 sources) is added.

User Operations

None.

Considerations

Music Source

Music sources or other audible indications are not provided with the switch-only access to these music sources.

Music Limit

As each local voice terminal places a call on hold, the Music-on-Hold software adds music to the held party's time slot.

Without a music source per module

Based on time-slot and TMS-blockage limitations, as many as 255 callers per module can listen to music at the same time. In practice, the limit is considerably lower.

With a music source per module:

When a music source is provided for every module, the practical limit more nearly approaches the theoretical limit of 255 callers per module. Music can simply be added to every held caller's time slot. (TMS-blockage is no longer a concern.)

Music Sources Provided for Some Modules

It is acceptable to provide a music source for some modules, but not for others. When this is done, the switch always provides music from ***one of two*** sources. When an extension in a certain module places a call on hold, music is added to the held party's time slot by that module's music source (if provided). Otherwise, music is added to the time slot using the music source attached to the ***lowest numbered module*** that does have a music source.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

When Attendant Call Waiting and Music-on-Hold Access are provided, a call placed by the attendant to a busy single-line voice terminal is connected to music until the called voice terminal is no longer busy or the attendant reconnects to the waiting call.

ACD (Automatic Call Distribution)

For ACDs using R2 V3 (beginning with Issue 1.4) and R2 V4, it is strongly recommended that each module in the switch where Music-on-Hold is desired be equipped with a music source. This will reduce the use of time slots and links between modules.

NOTE: Multiple music sources are not available in Release 2, Version 3 prior to Issue 1.4.

Call Park

When Music-on-Hold is implemented, music can be provided for a call on hold in an answer-back channel.

Call Vectoring

The music interface that can be provided with a "delay" step is functionally independent of the system-wide Music-on-Hold feature. To provide music for vector processing only, music source(s) should be provided and the Music-on-Hold software should be fully administered. Then, to turn off the ***regular*** Music-on-Hold feature, Field 11 of Procedure 275, Word 1 is set to "0."

Code Calling Access—Traditional and Universal

When Music-on-Hold is implemented, music can be provided for a call on hold in an answer-back channel.

Conference—Three Party

Music is provided to the first call put on hold when establishing a Conference—Three Party call. When an established three party conference call has been put on hold and one of the parties on hold goes on-hook, music is not given to the remaining party on hold.

Hold

If a conference call is put on hold, the conferees continue with the conference call and are not provided music, even if Music-on-Hold Access is active. Music is supplied to a nonconference party on hold.

Look-Ahead Interflow

At a Look-Ahead Interflow receiving switch, the music interface that can be provided with a "delay" step is functionally independent of the system-wide Music-on-Hold feature.

Loudspeaker Paging Access

When Music-on-Hold is implemented, music can be provided for a call on hold in an answer-back channel.

Personal Central Office Line

When assigned, Music-on-Hold is provided for held Personal Central Office Line appearances.

Malicious Call Trace

When Malicious Call Trace is activated on a switch that provides Music-on-Hold, the originator of the malicious call receives music whenever this caller is placed on hard hold. This effect usually occurs when Malicious Call Trace is activated from a multiappearance voice terminal using the dial access code. If this functionality is considered unacceptable, there are several ways to attenuate or circumvent the problem. To minimize the caller's alarm, the person activating Malicious Call Trace can advise the caller, "I need to put you on hold for a short time. I'll be right back." Other alternatives are to provide MCT ACTIVATE buttons for voice terminals that are likely to receive malicious calls, or to activate Malicious Call Trace from a neighboring voice terminal. If the previous alternatives are considered unacceptable, Music-on-Hold can be disabled using Procedure 275, Word 1.

Tenant Services

The Music-on-Hold Access feature is not partitioned. When a music source is provided and Music-on-Hold is enabled in Procedure 275, Word 1, music is provided to held parties in every partition.

Multiple music sources are also a system-side resource. If an extension (regardless of extension partition) in a certain module places a call on hold, music is added to the held party's time slot by that module's music source (if provided). Otherwise, music is added to the time slot using the music source attached to the ***lowest numbered module*** that does have a music source.

Transfer

When Music-on-Hold Access is provided, if a multiappearance terminal is transferring a single party, the party receives music after being put on hold. However, if transferring two parties, the parties do not receive music while on hold. Instead, the two held parties maintain a talking connection.

Hardware Requirements

The Music-on-Hold Access feature requires the following additional or special hardware.

For Traditional Modules:

- An auxiliary trunk circuit of an SN231 circuit pack for each music source (four circuits per SN231)

For Universal Modules:

- An auxiliary trunk circuit of a TN763C circuit pack for each music source (four circuits per TN763C)

Regardless of the Module Type:

- A 36A voice coupler
- A 2012D power transformer to supply power to the voice coupler
- Customer-provided music source(s).

Feature Administration

Access to the Music-on-Hold Access feature is assigned on a per-system basis. Whereas, music sources are assigned for the Music-on-Hold Access feature on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES MUSIC-ON-HOLD ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
054	3	Displays the Hold-With-Music button type for a multiappearance voice terminal.	Yes
100	1	Assigns the trunk type to the Music-on-Hold Access trunk group. The applicable encode is as follows: 62 Music-on-Hold Interface.	No
150	1	Assigns the SN231 or TN763C equipment location of a Music-on-Hold trunk (beginning with R2 V3, Issue 1.4, up to 31 trunks) to its trunk-group number.	No
275	1	Assigns Music-on-Hold to the system class of service (Field 11). This procedure also assigns Music-on-Hold for answer-back channels (Field 7). (The Call Vectoring feature can have Music-on-Hold without administering Field 11.)	Yes

Notes:

Off-Premises Data-Only Extensions

Description

The Off-Premises Data-Only Extension feature provides switched access to data equipment at a remote location (greater than 5000 feet) from System 85 or DEFINITY Generic 2 using analog or digital private line facilities that do not compete with voice traffic. This feature is used for data communications, when a significant volume of data is exchanged between the switch and a remote host computer or cluster of data terminals. Operation of this feature is transparent to all users. The users of the remote terminal cluster can access System 85 or DEFINITY Generic 2 data endpoints using data terminal (keyboard) dialing. unless specifially restricted, any terminal on the system that can originate data calls, can access a remote data endpoint as though it were on-premises.

Analog Service

The use of analog lines requires a dedicated modem and TDM for each private line facility to a remote data endpoint The analog facilities can be arranged on an individual line basis or on a multiplexed basis (Figure 89-1).

Digital Service

Digital private line service permits direct digital access to the switch and avoids the need for modems. This service is available with DATAPHONE® Digital Service. This service connects the System 85 or DEFINITY Generic 2 to Off-Premises Data-Only Extensions via DS1 carrier links that support data rates up to 56 Kbps on an individual basis or on a multiplexed basis (also shown in Figure 89-1). The DSU (Data Service Unit) provides the interface at each end of the DDS link. At the switch, the data module converts DCP to RS-232 for acceptance by the DSU. At the remote side, the output of the host computer or terminal cluster is RS-232, which connects directly to the DSU.

This service is not limited to local area support. The DATAPHONE Digital Service network uses T1 carrier links for connections under 100 miles and microwave radio links for distances more than 100 miles.

DATAPHONE Digital Service Supports user data rates of 2.4, 4.8, 9.6, and 56 Kbps. It can be interfaced with the system as individual lines or locally multiplexed to provide multiple lines per link. The best arrangement for a given application depends on the data rates and number of lines actually needed. In this way, the user can be provided with the level of service required by local needs. If 1200 bps is the rate needed for a group of off-premises work stations, several stations could be multiplexed over one DATAPHONE Digital Service link. Figure 89-1 shows a possible arrangement (using multiplexing and DATAPHONE Digital Service) to provide Off-Premises Data-Only Extension service for the System 85. The DATAPHONE Digital Service system does not provide multiplexing and demultiplexing. This must be locally engineered.

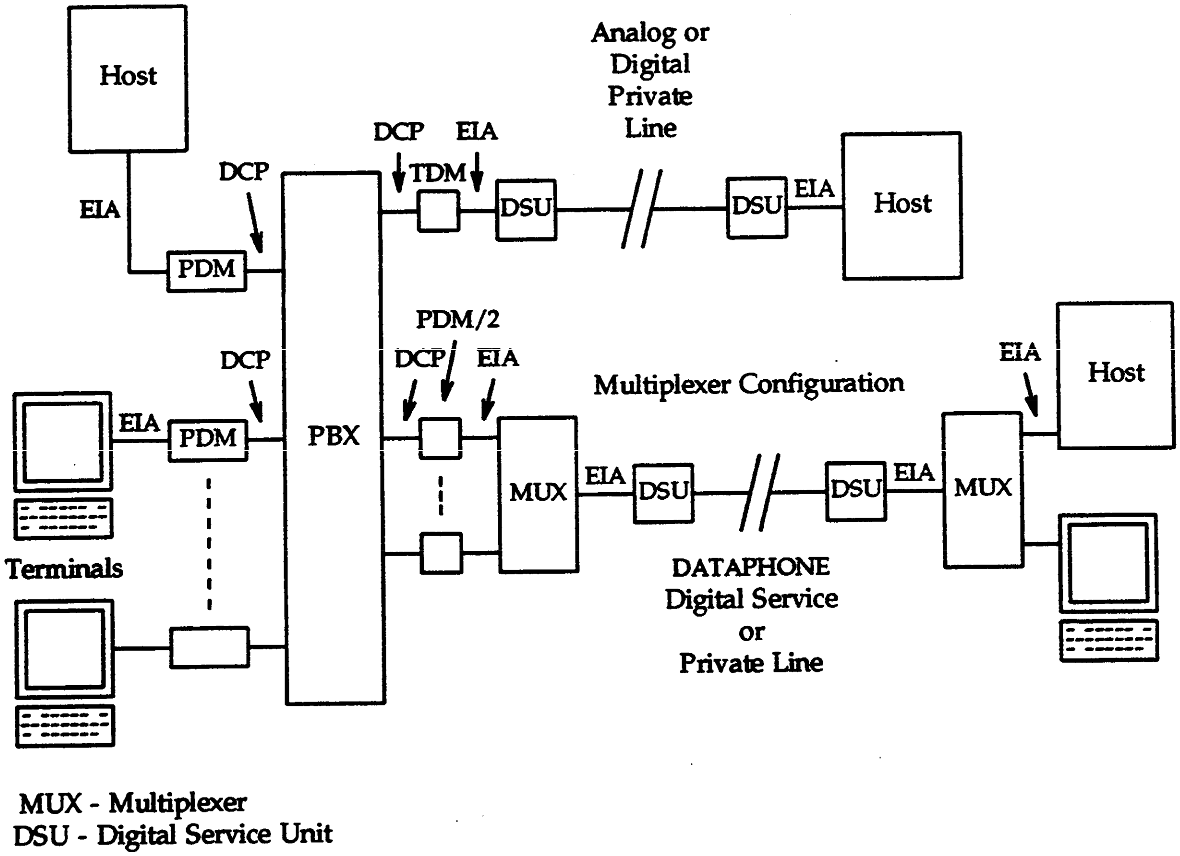

Figure 89-1. Off-Premises Data-Only Extension Arrangements

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The remote data endpoints (i.e., the terminal cluster or host computer) access endpoints on the switch using keyboard dialing. System 85 or DEFINITY Generic 2 users may use either voice terminal dialing or keyboard dialing. See Data Call Setup for specific procedures.

Considerations

Switch Appearance

The Off-Premises Data-Only Extension appears to the System 85 as an on-premises data module optioned for terminal dialing. As such, it has the same limitations and interactions as an on-premises data terminal (keyboard) dialing station.

Data Protection

These extensions should be assigned Data Protection—Permanent.

Interactions With Other Features

The following System 85 features affect or are affected by the operation of this feature.

The usual interactions that apply to features that involve trunks, such as Hunting and Queuing also apply to Off-Premises Data-Only Extensions.

Data Call Setup

The data module supporting the Off-Premises Data-Only Extension should be assigned Data Terminal (Keyboard) Dialing, an attribute of the Data Call Setup feature.

Host Computer Access

If the Host Computer Access feature uses TDMs to interface local host ports with the switch, off-premises data stations that also used a TDM for switch interface cannot access these ports or be accessed by the local host using these ports. This is because a TDM cannot handshake with another TDM. This problem can be avoided by using PDMs on Host Computer access ports (some or all) or by using the DMI (Digital Multiplexed Interface) feature for local host access.

ISDN—BRI

This ISDN—BRI feature provides line side service only in DEFINITY Generic 2 switches. ISDN—BRI terminals cannot be used for Off-Premises Data-Only Extensions.

Modem Pooling

If the off-premises data facilities access the local system using an MTDM, the Modem Pooling feature is not available. This is because an MTDM cannot handshake with the MTDM used in the Modem Pooling conversion resource.

Restricting Feature Use

Restriction capabilities that can be applied to any on-premises data module can be applied to the data module supporting the Off-Premises Data-Only Extension.

Hardware Requirements

The Off-Premises Data-Only Extension feature uses private line analog or digital facilities. For either traditional or universal modules, the specific hardware requirements depend on the arrangements used.

- If an analog private line is used, a dedicated modem and data module must be provided.

- If DATAPHONE Digital Service (DS1) is used, a Data Service Unit (DSU) is needed to interface the connecting line with the supporting data module.
- Multiplexer requirements can be customer engineered.

Feature Administration

Assignment of the Off-Premises Data-Only Extensions feature is on a per-line basis as a data module with terminal dialing capabilities.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can fully administer this feature using the SMT (System Management Terminal) and TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES OFF-PREMISES DATA-ONLY EXTENSIONS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns extension characteristics for off-premises extension.	Yes
010	3	Assigns data protection to an extension class of service.	Yes
051	1	Assigns characteristics of a TDM.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — OFF-PREMISES DATA-ONLY EXTENSIONS	
PATH NAME	**PURPOSE**
terminal-change terminal equipment	Assigns a data module to an equipment location, an extension number, and keyboard dialing.
terminal-change class-of-service attributes	Assigns Data Protection to an extension class of service.
terminal-change extension attributes	Assigns the attributes of an extension number (for example, assign an extension to a class of service).

Override

Description

The Override feature permits authorized multiappearance voice terminal users to interrupt other terminal users who are on a 2-party connection. Only terminal users who require the ability to contact other terminal users on a preemptive basis should be assigned this feature.

Warning Tone

The talking parties hear a warning tone before the third party enters the connection. The warning tone is a 4-second burst of 440-hertz tone.

An override call to a multiappearance voice terminal intrudes on a 2-party connection only when all appearances [including originating only appearances] are busy. If any appearance is idle, the override call terminates to the idle appearance with distinctive 3-burst ringing.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Override a Busy Voice Terminal

From a single-line voice terminal:

1. Go off-hook. If on a 2-way connection, momentarily press the switchhook, and dial the Call Hold access code. [Dial tone]
2. Dial the Override access code. [Second dial tone]
3. Dial the desired extension number. [Warning tone is heard by both parties on the active call and the overriding terminal user. The user is bridged onto the connection.]

From a multiappearance voice terminal (alternative 1):

1. Go off-hook. If on a 2-way connection, press **[TRANSFER]** or **[CONFERENCE]** .
2. Press an idle appearance button. [Dial tone]
3. Dial the Override access code. [Second dial tone]

4. Dial the desired extension number. [Warning tone is heard by both parties on the active call and the overriding terminal user. The user is bridged onto the connection.]

From a muitiappearance voice terminal (alterntaive 2):

1. Go off-hook. If on a 2-way connection, press **[TRANSFER]** or **[CONFERENCE]** .
2. Press an idle appearance button. [Dial tone]
3. Dial the desired extension. [Busy tone is heard. Special ringback is heard if Call Waiting is being used.]
4. Press **[OVERRIDE]** . [Warning tone is heard by both parties on the active call and the overriding terminal user. The user is bridged onto the connection.]

Considerations

Dark Status Lamp

When a 3-way connection is established and any party hangs up, the OVERRIDE status lamp goes dark.

Intercept Tone

Intercept tone results when:

- The calling voice terminal is not assigned this feature.
- An invalid access code or extension number is dialed.
- The calling voice terminal has voice terminal restrictions.
- The called voice terminal has voice terminal restrictions.
- The calling single-appearance voice terminal has a call on soft hold.
- The calling single-appearance voice terminal has a call on both soft hold and hard hold.
- An override call is attempted toward a connection involving a precedence call with a precedence level higher than Routine.

Busy Tone

Busy tone results when:

- The called voice terminal has a party in soft hold.
- The called voice terminal is maintenance busy.
- The called voice terminal is in any state other than a stable talking state.
- The called voice terminal is in a talking state with a auxiliary trunk. Auxiliary trunks are loudspeaker paging, etc.
- The called voice terminal is in a talking state with the attendant.

Hard and Soft Processor Swaps

If an Override user has entered a stable connection when a hard processor swap occurs, this 3-party connection will endure the hard swap.

A voice terminal user cannot override a line during a hard processor swap.

The Override feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

Override has precedence over any call waiting (via the Attendant Call Waiting feature) on the called extension. However, Override is denied toward the line that is waiting.

ACD (Automatic Call Distribution)

When Override is activated toward the individual extension number of a busy ACD agent, the override call intrudes into the agents active call. When Override is activated toward the individual extension number of an idle ACD agent, the override call alerts the agent with 3-burst ringing.

When Override is activated toward an associated extension number of an ACD split and the split supervisor is busy, the call intrudes into the split supervisor's active call. When the split supervisor is idle, the call alerts the supervisor with 3-burst ringing. (The Override call does not enter the split's queue.)

Bridged Call

The Override feature can be used toward a terminal that is in a normal talk state with another terminal, even if a third terminal is bridged onto the connection.

Busy Verification of Lines

Override is denied when the called extension is being busy verified.

Call Coverage

When Override is used to place a call to an extension with Call Coverage active, the call does not redirect to coverage. The call rings an idle appearance (with 3-burst ringing) or enters an active appearance (with warning tone, when every appearance is active) at the originally called extension.

Call Forwarding—Busy and Don't Answer

When a called terminal has activated Call Forwarding—Busy and Don't Answer, an override call does not forward. Three-burst ringing is provided for an idle forwarding terminal, and the override call enters the conversation of a busy forwarding terminal.

Call Forwarding—Don't Answer

When a called terminal has activated Call Forwarding—Don't Answer, an override call does not forward. Three-burst ringing is provided for the idle forwarding terminal.

Call Forwarding—Follow Me

An Override call does not forward when Call Forwading—Follow Me is active. Three-burst ringing is provided for an idle forwarding terminal, and the override call enters the conversation of a busy forwarding terminal.

Call Park

The switch denies Override toward an extension in call park.

Call Vectoring

Override calls cannot be placed to VDNs. When this is attempted, the switch returns intercept tone.

Call Waiting

Override has precedence over any call waiting (via the Call Waiting feature) on the rolled extension. However, Override is denied toward the line which is waiting.

Centralized Attendant Service

When a backup terminal for Centralized Attendant Service is handling a release link trunk call, the switch denies an attempt to enter the conversation by another terminal using the override feature.

Code Calling Access—Universal

Override is denied to a voice terminal line that has accessed code calling.

Conference—Attendant Five Party

Any attempt to use Override toward a terminal that is connected to a attendant conference is denied.

Conference—Attendant Six Party

Any attempt to use Override toward a terminal that is connected to a attendant conference is denied.

Conference—Three Party

Override is denied when directed toward a line connected to a three party conference call unless the line appears on a multiappearance terminal with more than one appearance assigned. In this case, the override attempt routes to an idle appearance of the line, if any.

Data Protection

Override is denied when the called extension or the trunk to which the extension is connected has the Data Protection feature active.

Distributed Communications System

The Override feature works only on the local switch in a DCS. If directed toward an extension on a distant DCS node, the caller attempting to use override receives intercept treatment.

EUCD (Enhanced Uniform Call Distribution)

When Override is activated toward the individual extension number of a busy EUCD agent, the override call intrudes into the agent's active call. When Override is activated toward the individual extension number of an idle EUCD agent, the override call alerts the agent with 3-burst ringing.

When Override is activated toward an associated extension number of an EUCD split and the split supervision is busy, the call intrudes into the split supervision active call. When the split supervisor is idle, the call alerts the supervisor with 3-burst ringing. (The Override call does not enter the split's queue.)

> **NOTE;** If the called agent (or split supervisor) is using a multiappearance terminal, an override call (in preference to intruding into the active call) will terminate to an idle appearance (if available) with 3-burst ringing. When no idle appearances are available, the override call will intrude into the active call.

Hold

The Hold feature interacts with the Override feature as follows:

- Override is allowed toward a voice terminal that has a call on hold.
- Override of a voice terminal's appearance that is on hold is denied. Busy tone is heard.

Hunting

An Override call does not hunt. The Override call completes to the called extension.

IPA (Interpartition Access)

A voice terminal user (in a partition other than Extension Partition 0 is allowed to place Override calls to extensions in the same partition group or in Extension Partition 0 When the user ties to place an Override call to an extension in any other partition group, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can place an Override call to any other extension in the switch.

Last Number Dialed

When a user places an Override call, the Override access code is not stored and dialed by the LND (Last Number Dialed) feature. The extension number dialed after the second dial tone is stored in LND memory.

Line Lockout

Busy tone is returned to a calling party who is attempting override toward a voice terminal in lockout.

Loudspeaker Paging Access

Override is denied to a voice terminal line connected to Loudspeaker Paging (making a page or waiting for answer-back).

Malicious Call Trace

If an Override call is attempted toward a line involved in a Malicious Call Trace, the calling party receives busy tone.

Precedence Calling

The Override feature is denied when directed toward any connection involving a Precedence Calling call with a precedence higher than ROUTINE.

Priority Calling

Override has precedence over any call waiting (via the Priority Calling feature) on the called extension. However, Override is denied toward the line which is waiting.

Privacy—Manual Exclusion

The Override feature allows a terminal user to enter a call even though Privacy—Manual Exclusion is active.

Queuing

Override is denied when the called extension is in an off-hook queue waiting for an idle trunk.

Restriction—Attendant Control of Voice Terminals

The use of Override is denied toward a terminal with Attendant Control of Voice Terminals—Termination restriction active.

Restriction—Voice Terminal Restrictions

An Override call to a terminal line with Termination or Manual Terminating Line restriction in effect is denied. If the restricted terminal is in a 2-party connection with an unrestricted voice terminal, override directed toward the unrestricted terminal is permitted.

Serial Calls

The Override feature is denied toward a line or trunk involved in a serial call.

Tenant Services

A voice terminal user (in a partition other than Extension Partition 0) is allowed to place Override calls to extensions in the same partition or in Extension Partition 0. When the user tries to place an Override call to an extension in any other partition, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can place an Override call to any other extension in the switch.

Trunk Verification—Attendant

If a 3-way connection has been established using the Override feature, Trunk Verification Attendant is denied.

Trunk Verification—Voice Terminal

If a 3-way connection has been established using the Override feature, Trunk Verification—Voice Terminal is denied.

Hardware Requirements

None.

Feature Administration

Assignment of the Override feature is on a per-terminal class-of-service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administer using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — OVERRIDE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the voice terminal class of service to an extension number.	Yes
010	1	Assigns Override to a voice terminal class of service.	Yes
054	2	Assigns the OVERRIDE button to a multiappearance voice terminal. The applicable encode is as follows: 5 Override.	Yes
350	1	Assigns the first digit of the dial access code (if required).	No
350	2	Assigns the Override dial access code. The applicable encode is as follows: 11 Override.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — OVERRIDE	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns Override to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change terminal buttons	Assigns the OVERRIDE button to a multi-appearance voice terminal.

Personal Central Office Line

Description

The Personal Central Office Line feature provides a multiappearance voice terminal user with direct access to dedicated CO (Central Office) trunks. Typical users are executives, dispatchers, or buyers with a high volume of calls going outside the switch. Businesses with specialized incoming traffic (such as a service department) can also benefit from these dedicated circuits.

Feature History and Development

This feature was first available on System 85 in Release 1. In Release 2, Version 2, the ability to transfer Personal CO Line calls was added.

User Operations

The following are the user operating procedures for this feature.

To Access a Personal Central Office Line:

1. Press the **[CO LINE PICKUP]** appearance button associated with the Personal CO Line. [Associated status I-use button lights.]
2. Go off-hook. [CO dial tone is heard, and the green status lamp lights.]
3. Dial the desired number.

To Release a Personal Central Office Line:

Go on-hook.

To Answer an Incoming Call on a Personal Central Office Line:

1. Press the **[CO LINE PICKUP]** appearance button associated with the Personal CO Line.
2. Go off-hook.

Considerations

Trunk Types

Only public network access trunks can be assigned to this feature. These trunks include:

- CO (Central Office)
- FX (Foreign Exchange)

- 800 Service
- WATS (Wide Area Telecommunications Service).

Maximums

A maximum of 150 trunks can be designated as Personal Central Office Lines. These 150 trunks can all be assigned to one trunk group or spread out among as many as 150 trunk groups. This arrangement depends on trunk type, requirements of the supporting central office, and local requirements.

The same Personal Central Office Line can appear on from 1 to 16 different voice terminals.

DROP Button and Personal Central Office Lines

Beginning with R2 V2, the user of a Personal Central Office Line who has established a 3-Party reference involving this line can press the DROP button to disconnect the third party in the conference.

Currently, however, the user of a Personal Central Office Line who is using this line for a 2-party call cannot press the DROP button to discconnect the call and to receive new dial tone on the same appearance.

Unassigned Personal Central Office Lines

An incoming call to a Personal Central Office Line that is not assigned to a voice terminal button routes to the attendant.

Voice Features

Most voice terminal features are not available on a Personal Central Office Line. Those features that can be used are:

- Abbreviated Dialing*
- Conference—Three Party.
- Hold
- Last Number Dialed*
- Music-on-Hold Access
- Transfer

* Available on Personal CO Lines administered as rotary out.

Intercept on Outgoing Calls

Intercept tone is received if:

- The user goes off-hook on an idle Personal Central Office Line that is a WATS 1-way incoming trunk
- The number dialed is invalid
- The terminal is restricted.

Single-Appearance Terminals and Straight Line Sets

The Personal CO Line feature cannot be assigned to single-appearance terminals. Also, single-appearance terminals (assigned as straight line sets in Procedure 051) cannot share a Personal CO Line appearance.

Hard and Soft Processor Swaps

Stable calls over Personal CO Lines endure a hard processor swap. However, calls cannot be placed over Personal CO Lines during a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

The Abbreviated Dialing feature can be used to place calls over Personal CO Lines that are administered as "rotary out."

The Abbreviated Dialing feature cannot be used to place calls over Personal CO Lines that are administered as "touch-tone out." In this case, the serving CO scans Personal CO Lines to detect an off-hook signal. After the CO recognizes an off-hook, the CO returns dial tone to the calling party. System 85 or DEFINITY Generic 2 software is not involved in this call-origination process. So, during the process, the switch does not invoke an originating register. Without this register, the switch cannot outpulse the stored digits.

CDR (Call Detail Recording)

The CDR feature does not record calls made on a Personal Central Office Line appearance.

Extension Number Portability

A user's extension number can be ported to another node, but their personal CO Line number may have to change.

Last Number Dialed

The Last Number Dialed feature can be used to redial calls over Personal CO Lines that are administered as "rotary out."

The Last Number Dialed feature cannot be used to redial calls over Personal CO Lines that are administered as "touch-tone out." In this case, the serving CO scans Personal CO Lines to detect an off-hook signal. After the CO recognizes an off-hook, the CO returns dial tone to the calling party. System 85 or DEFINITY Generic 2 software is not involved in this call-origination process. So, during the process, the switch does not invoke an originating register. Without this register, the switch cannot outpulse the stored digits.

Multiappearance Preselection and Preference

The Multiappearance Preselection and Preference feature does not automatically select Personal Central Office Line appearances.

Music-on-Hold Access

When assigned, Music-on-Hold is provided for held Personal Central Office Line appearances.

Power Failure Transfer

The Power Failure Transfer feature can be applied to Central Office trunks used as Personal CO Lines. However, the multiappearance voice terminal with the Personal CO Line cannot also be the designated station during the power failure.

Privacy—Manual Exclusion

Privacy—Manual Exclusion cannot be used to prevent bridging onto a Personal Central Office Line call.

Restriction—Attendant Control of Voice Terminals

Personal Central Office Line calls are not affected when an attendant activates a restriction toward an extension on a voice terminal.

Restriction—Voice Terminal Restrictions

Incoming calls on a Personal Central Office Line are not affected when the Voice Terminal Restrictions (Termination) feature is assigned to an extension class of service.

Tenant Services

There are no tests in Procedure 057, Word 1 to ensure that every voice terminal (from 1 to 16) sharing a Personal CO Line belongs to the same extension partition. It is the responsibility of the system manager to ensure that these shared lines do not cross partition boundaries.

A Personal CO Line trunk group can be dedicated to each extension partition that uses the personal lines. (There is a maximum of 150 Personal CO Lines that can be spread among as many as 150 trunk groups.) When this is done, each trunk group can be assigned (in Procedure 270, Word 5) to the specific extension partition to which the trunk group terminates.

To minimize the consumption of Personal CO Line trunk groups in a partitioned System 85 or DEFINITY Generic 2, these discrete personal lines can also be set up ***to converge to*** one (or a few) large trunk groups at the partitioned switch. When this is done, the several trunk groups can be assigned to Extension Partition 0 in Procedure 270, Word 5.

> **NOTE:** This specific trunking configuration is not a "shared trunk group" in the legal sense. The calling activity over Personal CO Line trunk groups would not be averaged. Rather, each CO Line in the trunk group is assigned to terminate to a specific voice terminal (or set of voice terminals), and can only be used by the assigned terminal(s). Each trunk facility is dedicated to a partition; the ***trunk-group numbers*** are shared.

Transfer

When a call on a Personal Central Office Line is transferred to another party, the owner of the personal line temporarily loses control of it. The owner cannot place or receive a call on that line until the Transfer call is finished. This can be avoided by setting up a 3-party conference and then placing the conference on hold. Then, at any time, the owner can reenter the connection and request use of the personal line.

WATS (Wide Area Telecommunications Service) Access

Using incoming (or 2-way) WATS trunks, the Personal Central Office Line feature can be used by a caller in the 800 Service area to reach a specific station on the System 85 or DEFINITY Generic 2 switch.

Hardware Requirements

The Personal Central Office Line feature requires the following additional or special hardware.

For Traditional Modules:

- Personal Central Office trunk circuits of an SN230 circuit pack (four circuits per SN230).

For Universal Modules:

- Personal Central Office trunk circuits of an TN747B circuit pack (eight circuits per TN747B).

Regardless of the Module Type:

- Multiappearance voice terminals.

Feature Administration

Assignment of the Personal Central Office Line feature is on a per-voice terminal and a per-appearance button basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

The Personal CO Line feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — PERSONAL CENTRAL OFFICE LINE			
PROCEDURE	WORD	PURPOSE	SMT
057	1	Assigns the trunk to an appearance button and sets the alerting option.	Yes
100	1	Assigns the dial access code and trunk type of a Personal CO Line trunk The applicable trunk-type encodes are as follows: 19 2-way automatic incoming attendant-completing/DOD 24 2-way automatic incoming attendant-completing/DOD 26 1-way automatic incoming attendant-completing 27 1-way outgoing DOD.	No
101	1	Administers the characteristics for the trunk group administered in Procedure 100, Word 1.	No
150	1	Assigns the SN230 or TN747B equipment location of a Personal CO Line trunk to its trunk-group number.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — PERSONAL CENTRAL OFFICE LINE	
PATH NAME	PURPOSE
terminal-change terminal personal-line	Displays or prints the Personal CO Line assignments.
terminal-change terminal buttons	Assigns the trunk to a button on a multiappearance voice terminal.

PC Interface

Description

The PC (Personal Computer) Interface feature consists of the PC/PBX and PC/ISDN product family referred to as *Data Interface Products.* These products are used with the System 85 and DEFINITY Generic 2 switches to provide users of AT&T PCs and other IBM* compatible PCs fully integrated voice and data work station capabilities.

Configurations

Several different configurations are available for the PC Interface feature. For convenience in referencing these break down into three groups. Groups 1 and 2 use the DCP (Digital Communications Protocol) while Group 3 uses the ISDN—BRI (Basic Rate Interface) protocol.

Configuration Group 1

Group 1, consists of those DCP PC/PBX configurations that use a PC Cartridge in a 7404D voice terminal (manufacture discontinued) to communicate with the switch. This group was formerly called package 1 (now Release 3.0X) and package 2. Group 1 is shown in Figure 92-1.

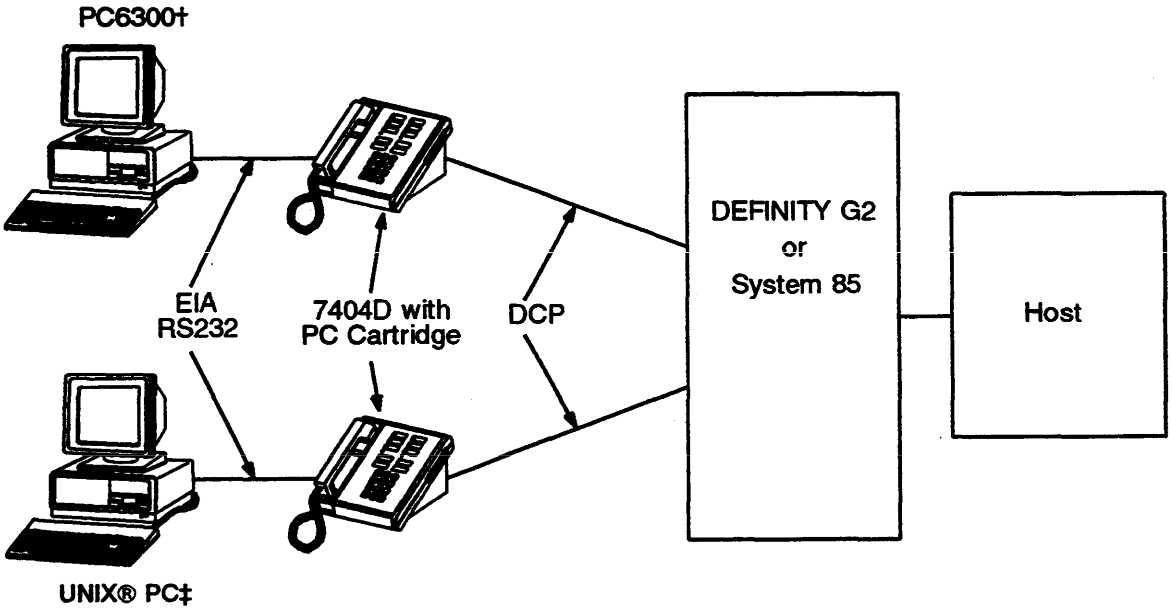

Figure 92-1. DCP PC Interface Configuration (Group 1)

* Registered trademark of the IBM Corporation.

† Other IBM-compatible PCs can be used, as well as the PC 6300.

‡ UNIX PCs were originally introduced but are no longer supported by the PC Interface feature.

Configuration Group 2

Group 2, consists of those DCP PC / PBX configurations that use the PC / PBX Interface Card (formerly DCP expansion card) in the PC itself to provide the communications interface with the switch. Group 2 uses the DCP protocol and is described in detail in PC / PBX Platform Installation and Reference manual, 555-016-101. Group 2 was formerly called packages 3, 4, 5, and 6, now combined as Release 3.0X Group 2 is shown in Figure 92-2.

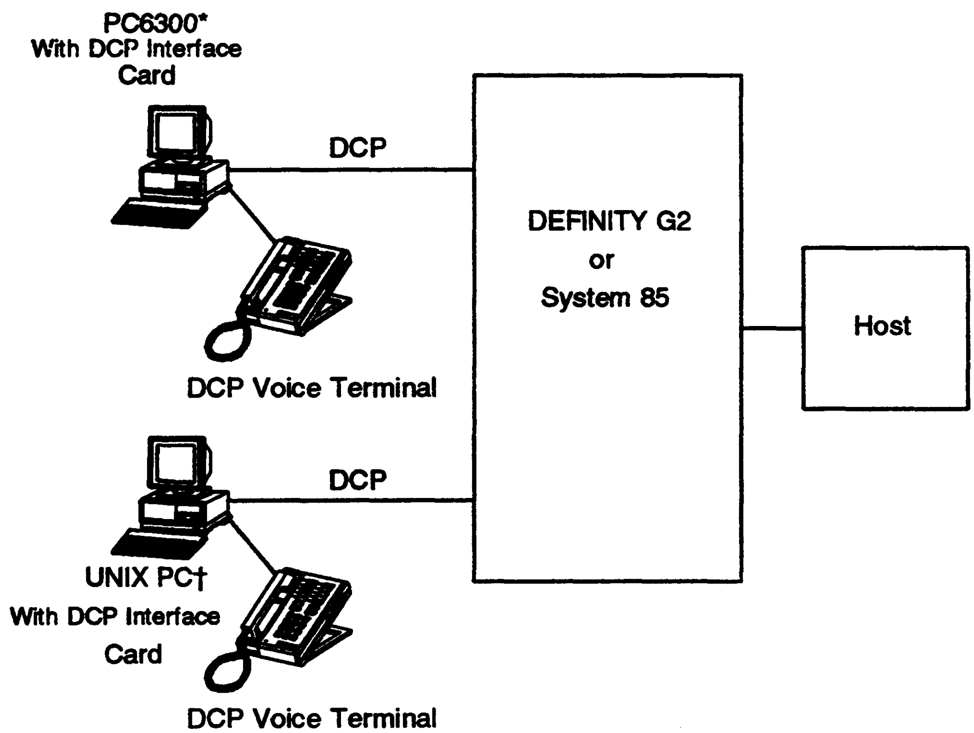

Figure 92-2. DCP PC Interface Configuration (Group 2)

Configuration Group 3

Group 3 consists of those configurations that use the ISDN—BRI (Basic Rate Interface) for communications. Connectivity to the switch is provided by the PC / ISDN Interface Card installed in the PC itself. Possible arrangements in this group include the PC as a stand alone terminal (PC only), or with from one to four voice terminals, hand sets, or headsets. Group 3 is available on the DEFINITY Generic 2 switch but not on previous System 85 switches. Group 3 is shown in Figure 92-3.

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* Other IBM-compatible PCs can be used, as well as the PC 6400.

† UNIX PCs were originally introduced but are no longer suppoted by the PC Interface feature.

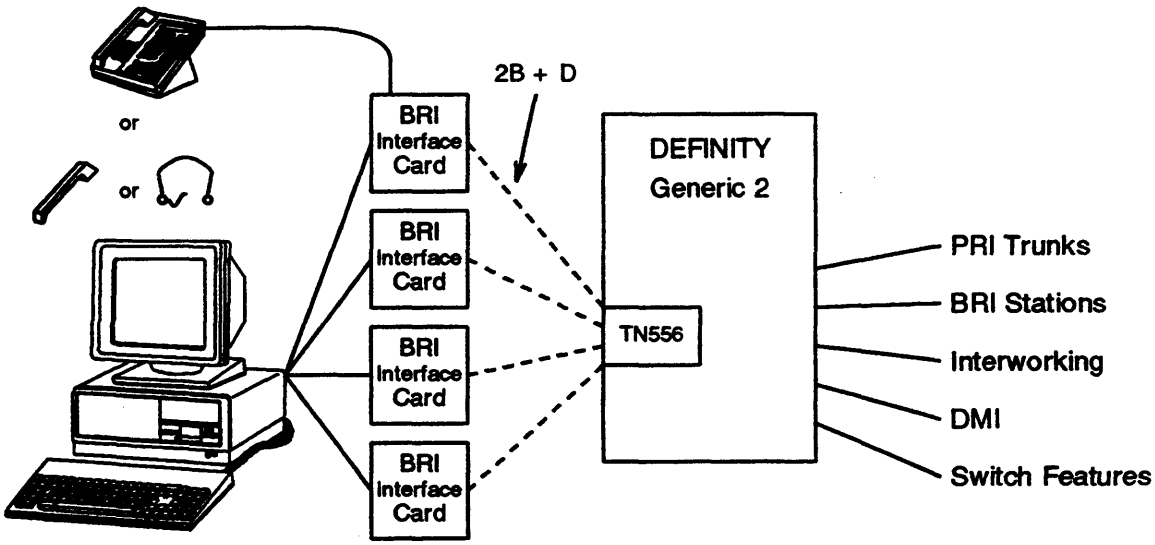

Figure 92-3. ISDN—BRI PC Interface Configuration (Group 3)

Applications and Benefits

PC Interface users receive benefits provided by both the PC and the features and services of the System 85 or DEFINITY Generic 2 switch.

Switch Features and Services

PC Interface users have multiple appearances (depending on the software used) of their assigned extension. One or more of these appearances can be designated for use with data calls. With the ISDN-BRI version, up to four separate PC/ISDN Interface Cards can be used on the same PC. Each interface card used can be assigned a separate extension number, and each of these extension numbers have multiple appearances. The availability of specific features depends on the class of service of the assigned extension and the system class of service of the switch. Modem Pooling should be provided to ensure general availability of off-net calling services.

PC Features and Services

Features and services provided by the PC are a function of the PC software in place. With the PC/PBX Connection software, they include PC based Directory Services (on line), a Last Number Dialed feature (separate from these features as provided by the switch), and basic asynchronous terminal emulation.

Other options include such functions as synchronous 3270 Emulation and Hayes Smart Modem Emulation. The E78 Plus/ISDN software package provides enhanced software based 3270 emulation with high speed (64Kbps) connections (through a 3270 data module) to a local or remote 3270 cluster controller. This includes bulk file transfer capability over switched links including either ETN (Electronic Tandem Network) or ISDN circuits. Expensive fixed coaxial connections can be avoided. The user interface with the

E78 Plus/ISDN software is the same as with the widely used E78/IRMA™ hardware Configurations.

See the appropriate installation and reference manual (555-016-101 for Group 2 and 555-016-102 for Group 3) for further details on PC software options and features.

The PC/ISDN Platform offers local custom applications programming capability. A separately orderable publication, The PC/ISDN Software Developer's Guide, 555-016-103, provides the information that users will need to customize existing applications software or to develop new applications software to meet local needs.

Feature History and Development

The DCP versions of this feature (Groups 1 and 2) were first supported in System 85 R2 V4, as the PC/PBX Connection feature. For System 85 R2 V4 and later switches, PCS have their own terminal type (PC). On earlier versions of the Release 2 System 85 switch, PC/PBX configuration Groups 1 and 2 can be used by administering them as an alias. Use terminal type BCT 510D for Release 2, Versions 2 and 3. Use 7405D with display module for Release 2, Version 1 switches.

With DEFINITY Generic 2, the feature name is changed to PC Interface to avoid confusion with the PC/PBX Connection software package which is part of the overall offering. PC Interface Group 3, the ISDN—BRI version of this feature, is first available on DEFINITY Generic 2.1, Issue 2.0. The PC/ISDN Platform (Group 3) cannot be used with System 85 switches.

User Operations

User operations depend on the software used with the PC. For specific user operations, refer to the appropriate PC Interface documentation set as follows:

- 555-016-715, PC/PBX Connection

  Documentation Set for the DCP versions (Groups 1 and 2) using the PC/PBX Connection software package.

- 555-016-101, PC/PBX Platform

  Installation and Reference manual for PC/PBX Platform.

- 555-106-102, PC/ISDN Platform

  Installation and Reference Manual for the PC/ISDN Platform (Group 3).

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Considerations

Use of Speakerphones

Most System 85 and DEFINITYGeneric 2 speakerphones can be used with the PC Interface.

Function Key Module

The Function Key Module of the 7405D can be used with the PC Interface feature.

SPD (Service Profile Identifier)

On the DEFINITY Generic 2 switch, BRI terminals are normally "initializing terminals" and require that a SPID be assigned *(see the* ISDN—BRI feature).

The PC/ISDN Platform (Group 3), in a stand alone configuration (no associated voice terminal), is "non-initializing BRI terminal." A non-initializing terminal does not require a SPID. When the PC/ISDN Platform is assigned as a stand alone non-initialing terminal, it is administered using a locally defined terminal type with General Terminal Administration, Procedure 50, Word 1 *(see* the Administration Procedures Manual, 555-105-506). When this is done, the terminal type is defined as a non-initializing terminal that does not support MIMs (Management Information Messages). This requires a value of "0" in field 6 of Procedure 50, Word 1. Other specific characteristics of the locally defined terminal type will depend on the telephone manager applications software used with the PC.

The PC/ISDN Platform can also be assigned with an (initializing) ISDN—BRI voice terminal (such as an ISDN 7505) that uses a SPID. In this case, the station should also be assigned using a locally defined terminal type to take full advantage of the capabilities of the PC Interface. However, this terminal type should be initializing with support of MIMs.

Internal Features

Specific internal features available depend on the application package used. The PC/PBX products provide extensive local directory capability (up to 32,000 entries). They also provide a Last Number Dialed capability that is separate from the Last Number Dialed feature on the System 85 or DEFINITY Generic 2. Configuration Group 3 (the ISDN—BRI version) also provides a separate Last Number Dialed capability that can be used from either the voice terminal or the PC keyboard.

Data Modules

Voice terminals with data modules are not recommended for use with the PC Interface feature (except for 3270 Data Modules when 3270 emulation is used). If a DCP data module (such as a DTDM) or ISDN data module (such as an ADM-T) is attached to the voice terminal used in conjunction with the PC Interface card (either DCP or ISDN—BRI), the data module will be bypassed (not used). All interface functions are performed by the interface card even if a data module is present.

Display Modules and Terminals

A voice terminal with display capabilities is not recommended for use with the PC/PBX Connection package. If one is used, administer the voice terminal as a 7405D plus display module and data module.

Call Appearances

The DCP PC Interface configurations (Groups 1 and 2) support five appearances, one of which must be dedicated to data use. This confirmation limit may not be optimal for users (such as attendants) who need many call appearances. The PC/ISDN Platform supports many more call appearances. On the DEFINITY Generic 2, the PC/ISDN Platform is limited by switch administration to a maximum of 208 call appearances (52 call appearances per interface card X a maximum of four interface cards = 208 appearances.). Note that each extension number is limited to 12 appearances. This means that each interface card can accommodate four and one third extension numbers (without any bridged appearances). Special applications software must be installed on the PC to provide access to all these appearances.

7404D Voice Terminal

The 7404D voice terminal with messaging cartridge cannot be used with the PC Interface. The 7404D is manufacture discontinued and may no longer be ordered.The 7404D with PC cartridge is used only with Group 1.

Interactions With Other Features

The PC Interface feature interacts with most other features in the same way as other DCP and ISDN—BRI voice and data terminals.

Data Communications Access

The PC Interface feature uses a digital interface (either DCP or ISDN—BRI) and is not directly compatible with the Data Communications Access feature which uses an analog interface Modem Pooling like conversion must be applied if these two features are to be used together.

Data Protection

The PC Interface feature is used for data communications, therefore Data Protection—Permanent should be assigned.

Host Computer Access

Both the PC Interface feature and the Host Computer Access feature use digital interfaces. These features are directly compatible (no Modem Pooling conversion needed).

ISDN—BRI (Basic Rate Interface)

The ISDN—BRI feature must be active on the switch to use the PC/ISDN Platform (Group 3). With ISDN—BRI configurations, up to four PC/ISDN interface cards can be installed in one PC. When multiple cards are used, each card is assigned to a separate and distinct interface on the switch. Separate interface cards cannot share the same ELL (Equipment Line Location). Each separate interface card can have its own separate voice terminal or voice-calling device. When a voice terminal is used, special applications software is not required on the PC. However, to use a handset or headset alone, special applications software is needed.

Modem Pooling

Modem Pooling is needed if the PC Interface feature is used to place calls to, or receive calls from, off-premises stations over analog trunks.

Restricting Feature Use

The voice and data appearances used by the PC Interface are subject to the same general restrictions as other like appearances. These can be applied either as fixed restrictions through the extension class of service, or as temporary restrictions through the Attendant Control of Voice Terminals feature. Other restrictive measures, such as the FRL (Facilities Restriction Level) and Attendant Control of Trunk Group Access features also apply to stations using the PC Interface feature.

Hardware Requirements

The specific hardware required by the PC Interface varies depending on the configuration group used.

Group 1

Configuration Group 1 consists of those configurations that use the DCP PC interface cartridge in the 7404D voice terminal.

- 7404D VDS (Voice Data Station) — ***Manufacture Discontinued***
- 31815 PC cartridge
- Standard, EIA RS232 connecting cables
- One of the following Personal Computers
  - — AT&T PC 6300 and compatibles, with MS-DOS* Version 2.11 or later

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* Trademark of the Microsoft Corporation.

† Trademark of the IBM Corporation.

‡ Registered trademark of the IBM Corporation.

§ Trademark of the IBM Corporation.

— AT&T PC 6300 Plus, with MS-DOS Version 3.1 or later

— IBM PC and PC/XT†, with PC-DOS‡ Version 2.0 or later existing versions

— IBM PC/AT§, with PC-DOS Version 3.1 or later.

- PC Accessories:

  — Serial asynchronous communications port (standard on AT&T PC 6300 and 6300 Plus)

  — 256K RAM minimum (PC/PBX Connection application package)

  or

  384K RAM recommended minimum for integrated software applications such as concurrent Lotus 1-2-3. ∝ .

Group 2

Configuration Group 2 consists of those DCP configurations that use the PC Interface card in the PC itself.

- Most Sytem 85 or DEFINITY Generic 2 7400-series (DCP) digital voice terminals can be used.
- PC/PBX Interface Card
- One of the following Personal Computers:

  — AT&T PC 6300 series and compatibles, including PC 6300,6300 WGS, 6300 Plus, 6310, 6312 WGS, 6286 WGS, and 6386 WGS, with MS-DOS Versions** up to 4.X (except 3.0)

  — IBM PC, PC/AT, PC/XT, and Personal System/2 Model 30 and Model 30/286 with PC-DOS Versions up to 4.X (except 3.0).

- Memory Configurations (depending on application being run):

  — 320K RAM minimum

  or

  — 448K RAM recommended for large integrated software applications such as concurrent Lotus 1-2-3.

Group 3

Configuration Group 3 consists of BRI configurations that use a PC expansion card in the PC itself (there is no BRI cartridge option).

∝ Trademark of the Lotus Development Corporation.

** DOS releases beginning with 1 (for example 1.XX) cannot be used.

- Voice Terminal or Voice Calling Devices
  - Any DEFINITY Generic 2 ISDN—BRI Voice Terminal (7500 Series) can be used with the Group 3 configurations.

    or

  - With appropriate applications software, an AT&T R-Type replacement Handset (PEC X10150) can be plugged into the ISDN—BRI PC expansion card instead of a BRI voice terminal.

    or

  - A headset can be used instead of either a BRI voice terminal or the R-Type Handset. The following Plantronics headsets can be used:

    StarSet* Series Communications Headset model StarMate* MH0228-3

    Supra* Series Communications Headset model MH0528-3

    Supra Series Communications Headset model MH0529-3.

- PC/ISDN Expansion Card (up to four per PC).
- One of the following personal computers:

  AT&T Personal Computers

  - AT&T PC 6300 (and compatibles)
  - AT&T PC 6300 Plus
  - AT&T 6300 WGS
  - AT&T PC 6310
  - AT&T PC 6312 WGS
  - AT&T PC 6286 WGS
  - AT&T PC 6386 WGS

  IBM Personal Computers

  - IBM PC
  - IBM PC/XT
  - IBM PC/AT
  - PS/2† Model 30

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* Registered trademark of the Plantronics Corp.

† Trademark of the IBM Corporation.

  — PS/2 Model 30—286.

Compaq‡ Personal Computers

  — Compaq DeskPro286

  — Compaq DeskPro386.

- PC Accessories:

  — For AT&T, Compaq and compatible PCs, 2.0 and all later releases (except 3.0) of MS-DOS

  — For IBM PCs, 2.0 and all later releases (except 3.0) of PC-DOS.

System Connections:

Group 1

- The RS232 port of the 7404D voice/data set is connected via an EIA RS232 cable to the RS232 port of the PC.
- The 7405D voice/data set has a modular digital port connection to the System 85 or DEFINITY Generic 2.
- Standard operating distance of the 7404D from the PBX is 5,000 feet maximum for 24-gauge wire or 4,000 feet for 26-gauge wire.

Group 2

- The PC Interface card plugs into an expansion slot on the PC. The card has two standard, 8-pm modular jacks (line and phone).
- The digital phone plugs into the phone jack on the PC Interface card.
- The line jack on the card provides a digital port connection to System 85 or DEFINITY Generic 2.
- Standard operating distance of the PC Interface card from the PBX is 5,000 feet maximum for 24-gauge wire or 4,000 feet for 26-gauge wire.

Group 3

- The PC/ISDN Interface card (from one to four cards) plugs into an expansion slot on the PC. The card provides two standard 8-pin modular jack connections for both a line connection (to the switch) and a phone connection. A standard 4-pin modular jack is also available for use with a hand-set or head-set rather than a voice terminal.
- Each expansion card provides separate and distinct line and phone connections.
- Standard and maximum operating distances for ISDN—BRI terminals are a function of a number of factors including loop signal loss and power. The factors affecting

‡ Registered trademark of Compaq Computer Corporation.

operating distance limits are discussed in detail under Distance Specifications in the **DEFINITY Generic 2 System Description,** 555-105-201.

Feature Administration

Administration guidelines in this section pertain to the System 85 Release 2 and DEFINITY Generic 2 switches. For additional information and for information on administering the PC itself, refer to the **PC Interface Installation and Reference Guide** (part of documentation set 555-016-715), **PC/PBX Platform Installation and Reference** manual, 555-016-101, or **PC/ISDN Platform Installation and Reference manual,** 555-016-102.

The PC Interface feature is administered on a per-station basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can also administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — PC INTERFACE			
PROCEDURE	**WORDS**	**PURPOSE**	**SMT**
050	1 & 2	Used for General Terminal Administration to locally define new terminal types (PC/ISDN Platform).	N/A
051	1	Assigns and removes terminal types.	Yes
051	2	Assigns the SPID (Service Profile Identifier) used with the voice terminal portion of the PC/ISDN interface.	Yes
052	1	Assigns extension numbers and images to an ELL and to specific buttons.	Yes
052	2	Assigns additional extension and appearance (image) characteristics to ELL and buttons.	Yes
054	1	Assigns miscellaneous feature buttons, including the *Wait For Principal* button to a multiappearance voice terminal. The Wait For Principal button is used only with the PC Interface feature in a Message Center role.	Yes
054	2	Assigns and removes Custom Calling feature buttons.	Yes
054	4	Assigns and removes Display feature buttons.	Yes
055	1	Assigns and removes Terminal Busy feature buttons.	Yes
055	2	Assigns and removes One Button Transfer feature. buttons.	Yes

(Continued)

ADMINISTRATION PROCEDURES — PC INTERFACE *(Continued)*			
PROCEDURE	WORDS	PURPOSE	S M T
056	1	Assigns and removes Intercom buttons.	Yes
057	1-3	Assigns and removes CO line buttons.	Yes
		Performs a station swap between two equipment locations.	Yes
059	1-5	Assigns and removes Abbreviated Dialing buttons.	Yes
063	1	Assigns and removes Automatic Message Waiting buttons.	Yes
070	4	Displays information about the terminal equipment assigned to an equipment location.	Yes

Power Failure Transfer

Description

The Power Failure Transfer feature uses an emergency transfer panel that provides five circuits for bypassing the switch if the common control loses power. Each circuit directly connects a telephone to a CO trunk. Typically, these telephones are placed with people who need to keep receiving incoming calls or place outgoing calls should the switch happen to be out of service

During a commercial power failure the common control in a standard power-option switch will operate off of batteries for a brief period of time; common controls in switches equipped for extended power holdover will operate as long as the battery plant can maintain adequte voltage. In either case, once battery-provided voltage is too low, battery power is shut off and the emergency transfer panel connects designated voice terminals directly to designated co trunks.

The model 808A emergency transfer panel provides five emergency transfer circuits. During normal switch operation, -48V DC power from the alarm panel keeps the 808A's power failure detection relays open and lights a green LED on the 808A emergency transfer panel. During a power failure or major system failure the 808A operates as follows (see Figure 93-1):

- Upon failure, the power failure detection relays switch to bypass mode.
- Each emergency transfer circuit directly connects a designated model 8110, 8102, 7102, 2500-type, or other FCC-registered analog voice terminal to a central office (CO) trunk. The switch is completely bypassed.
- When a voice terminal connected to the 808A goes off-hook during emergency transfer, circuitry inside the panel places signaling on the CO trunk causing the CO to return dial tone. Each 808A bypass circuit can be optioned for either loop-start or ground-start signaling.

Should the -48V DC feed from the alarm panel come back on-line while a call connected through the 808A is in progress, the 808A will maintain the connection until the user goes on-hook. This is an improvement over the model 574-5 emergency transfer panel which simply dropped all calls in progress when power was restored. Unlike the older 609A transfer panel, the 808A panel doesn't require the voice terminal user to operate a signaling key.

NOTE: Acceding to FCC requirements, the local exchange (telephone) company must be notified before installing an emergency transfer panel. Furthermore, the local exchange company must be notified when the panel is removed.

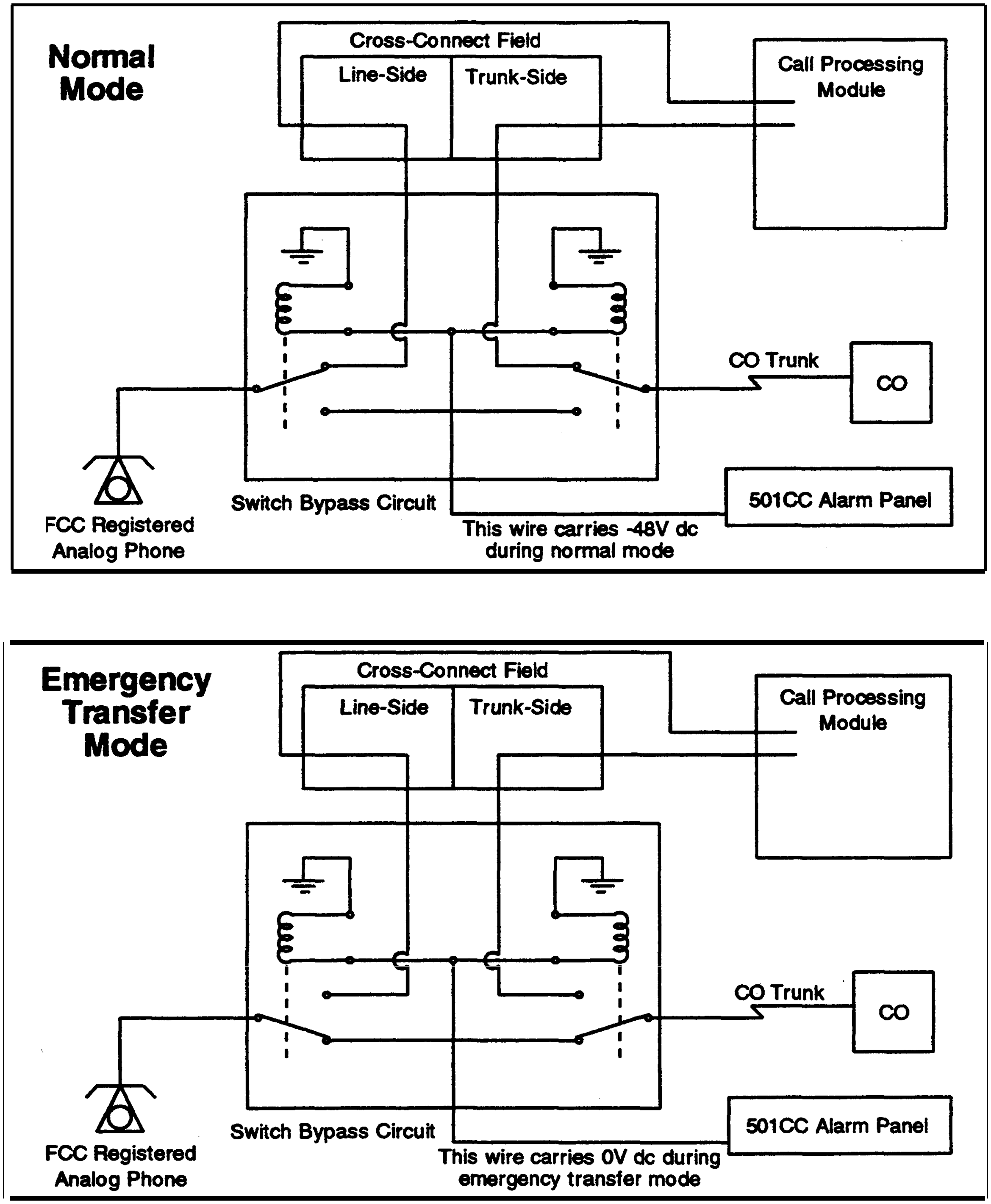

Figure 93-1. Emergency Transfer Circuit

In addition to power loss, the following conditions also cause emergency transfer:

- Common control failure. When the system is equipped with duplicated common controls, both common controls must fail.
- Fuse failure in the alarm panel circuit supplying the -48V to the emergency transfer panel.
- The EMERGENCY TRANSFER switch on the alarm panel being set to the ACT (activate) position.
- Seventy-five percent of the system's modules being out of service.
- TMS (Time Multiplexed Switch) being out of service.
- The TRANSFER TEST SWITCH on the 808A being activated.

When an emergency transfer occurs, the System 85 or DEFINITY Generic 2 switch is bypassed. Since the switch is bypassed, switch features and restrictions do not apply to telephones using activated emergency transfer circuits. When emergency transfer occurs due to TMS or module failure, terminals that have not been transferred by the panel may still work; however, calls between extensions or facilities located in different modules may not be possible.

Feature History and Development

This feature was first available for System 85 in Release 1. It was initially supported by the 609A panel, then by the 574-5 panel, and now by the model 808A panel.

Model 574-5 operation is essentially the same as 808A operation except that the 574-5 immediately drops all calls in progress when power is restored — the 808A does not.

With the 609A a ground-start key is required at each designated emergency transfer telephone to signal the CO that the telephone user wants to place a call. The 609A panel provides ten emergency transfer circuits.

Power Failure Transfer for Remote Modules

Remote modules can be equipped for power failure transfer. An 808A (or other model) emergency transfer panel is connected to a ZAEY1 circuit pack in the remote module fan assembly. The ZAEY1 provides the -48V DC feed to the panel. When an emergency transfer condition exists at the central locale, the common control sends a message to the remote module's ZAEY1 circuit pack instructing it to turn off the -48V DC feed which will, in turn, cause the panel at the remote locale to go into emergency transfer mode.

In addition to central-locale emergency transfer criteria, the following conditions also cause emergency transfer at a remote module.

- Remote module power failure
- Remote module link failure — the communications link between the remote module and the central locale fails
- Remote module processor failure.

User Operations

The Power Failure Transfer feature goes into effect automatically.

To Receive a CO Call on a Power Failure Transfer Terminal:

The emergency transfer telephone rings using current from the CO — the user answers. Distinctive ringing patterns from the switch are not available.

To Originate a Call From a Power Failure Transfer Terminal

When an 808A or a 574-5 panel is used:

1. Go off-hook. [CO Dial tone]
2. Dial a destination number.

When a 809A Panel is used:

1. Go off-hook.
2. Press the **[GROUND START]** button. [CO Dial tone]
3. Dial a destination number.

Considerations

Lost Calls

When power to the common control is lost, all calls are lost when emergency transfer occurs. When emergency transfer occurs due to the previously mentioned major alarm conditions, calls whose connection paths use only unaffected portions of the switch will stay up. Other calls may or may not stay up depending on the exact nature of the alarm and the connection paths.

Emergency Transfer Connections and Recovery

When the switch recovers from the condition that caused the emergency transfer, the -48V DC feed from the alarm panel to the emergency transfer panel is restored. Emergency transfer connections using the 574-5 panel or the 609A panel are immediately dropped once the -48V DC feed comes back on-line. When the 808A emergency transfer panel gets back the -48V DC feed, it maintains each existing connection until the connection's user goes on-hook.

Trunk Usage

This feature is not normally used on FX (Foreign Exchange) or WATS (Wide Area Telecommunications Service) trunks. However, when these trunks are analog and powered from an outside source such as the CO, they may be used for power failure trunks. Such arrangements must be made with the local CO.

Dial "0" Calls

When "0" is dialed from an emergency transfer telephone while emergency transfer is active, the local exchange company's operator will answer.

Hard Processor Swaps

A hard processor swap will ***not*** cause an emergency transfer. A "hard processor swap" is an event that can take place in a switch equipped with duplicated processors. For more information about processor swaps, see the **Duplication** section in Chapter 2 of the *DEFINITY Generic 2 and System 85 System Description (555-105-201).*

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

DID (Direct Inward Dialing)

A DID call cannot be received during the power failure transfer mode of operation

Multiple Listed Directory Numbers

During a power failure, a switch that uses incoming (or 2-way) CO trunks for non-DID LDN (Listed Directory Number) service can transfer these LDN calls to predesignated voice terminals.

Personal Central Office Line

Central Office trunks used for Personal Central Office Lines can also be used for power failure transfer. However, if multiappearance voice terminals are used with the Personal Central Office Line, these terminals cannot be used as a designated station in a power failure transfer.

Unattended Console Service — Preselected Call Routing

When full (flexible) night station service is activated and a power failure occurs, the terminal line(s) assigned by the attendant is lost from memory. When commercial power is restored and reinitialization occurs, night station service calls are routed to the default terminal.

Hardware Requirements

Emergency transfer panel

The latest emergency transfer panel offering is the AT&T model 808A. The 808A provides five emergency transfer circuits and should be used for all new switches and additions. It is a direct replacement for the 574-5 panel. Model 574-5 and model 609A still work on all versions of System 85 and Generic 2. Each 609A emergency transfer circuit requires a ground start key for CO trunk signaling. The 609A provides ten emergency transfer circuits.

The emergency transfer panel is usually installed at the cross-connect field.

FCC-registered telephones

Each emergency transfer circuit requires an FCC-registered analog telephone such as the AT&T model 8110, 8102, 7102, or a 2500-type set. This requirement exists since the telephone is cut through directly to the public network during emergency transfer.

CO Trunks

Each emergency transfer circuit requires an analog CO trunk circuit such as the SN230B on a traditional module, or the TN747B on a universal module.

Feature Administration

The Power Failure Transfer feature is provided by hardware and requires no administration.

Precedence Calling

Description

Precedence Calling operates in the AUTOVON (Automatic Voice Network). However, Precedence Calling can be adapted for use within any private network that uses a "STAR" or hub-like configuration. The Precedence calling feature provides two capabilities: ***preemption*** and automatic ***diversion to attendant assistance.*** These capabilities help to ensure the rapid completion of important calls. Precedence Calling is specifically designed for national defense and emergency calling situations. On System 85 or DEFINITY Generic 2, the Precedence Calling feature enables the switch to function effectively in the AUTOVON environment and can extend this ability to an associated DCS (Distributed Communications System) network.

Feature History and Development

Precedence Calling was first available on System 85 with Release 1. In its initial form, special voice terminals and attendant consoles were required.

Gateway type AUTOVON access service for a DCS was introduced with System 85, Release 2, Version 1. Hardware independent Precedence Calling using software routines to provide Precedence Calling capabilities from standard voice terminals and attendant consoles was also introduced in System 85 with Release 2, Version 1.

On switchs prior to the DEFINITY Communications System, Generic 2.2 switch, the AAR feature is required for the Precedence Calling feature to work. With Generic 2.2 this is no longer required; however, the Standard Network option is still required.

The AUTOVON Network

The AUTOVON network itself is part of the Defense Communications System. It provides private network services between US. Government Offices and Installations with defense or defense-related missions and functions. Certain local government and private organization sites (Civil Defense, Red Cross, contractors, etc.) also may be granted AUTOVON access.

AUTOVON Access

A System 85 or DEFINITY Generic 2 switch can access the AUTOVON either as an isolated switch (single subscriber switch on the network) or as a gateway for a private-network configuration.

Gateway Configuration

The private-network configuration consists of a central or ***Hub*** switch that provides access (gateway service) to the AUTOVON for other switches in the network. The central switch is individually connected to the other nodes in a "STAR" network configuration (dilink between each node and the Hub or central node). Both these configurations (single subscriber switch and gateway service arrangement) are depicted in Figure 94-1.

NOTE: For maximum functionality, it is recommended that the private network be set up as a DCS network. Tie trunks that are to be used for Precedence Calling (precedence capable trunks) must be dedicated to AUTOVON access; however, the DCIU links do not need to be dedicated to AUTOVON use.

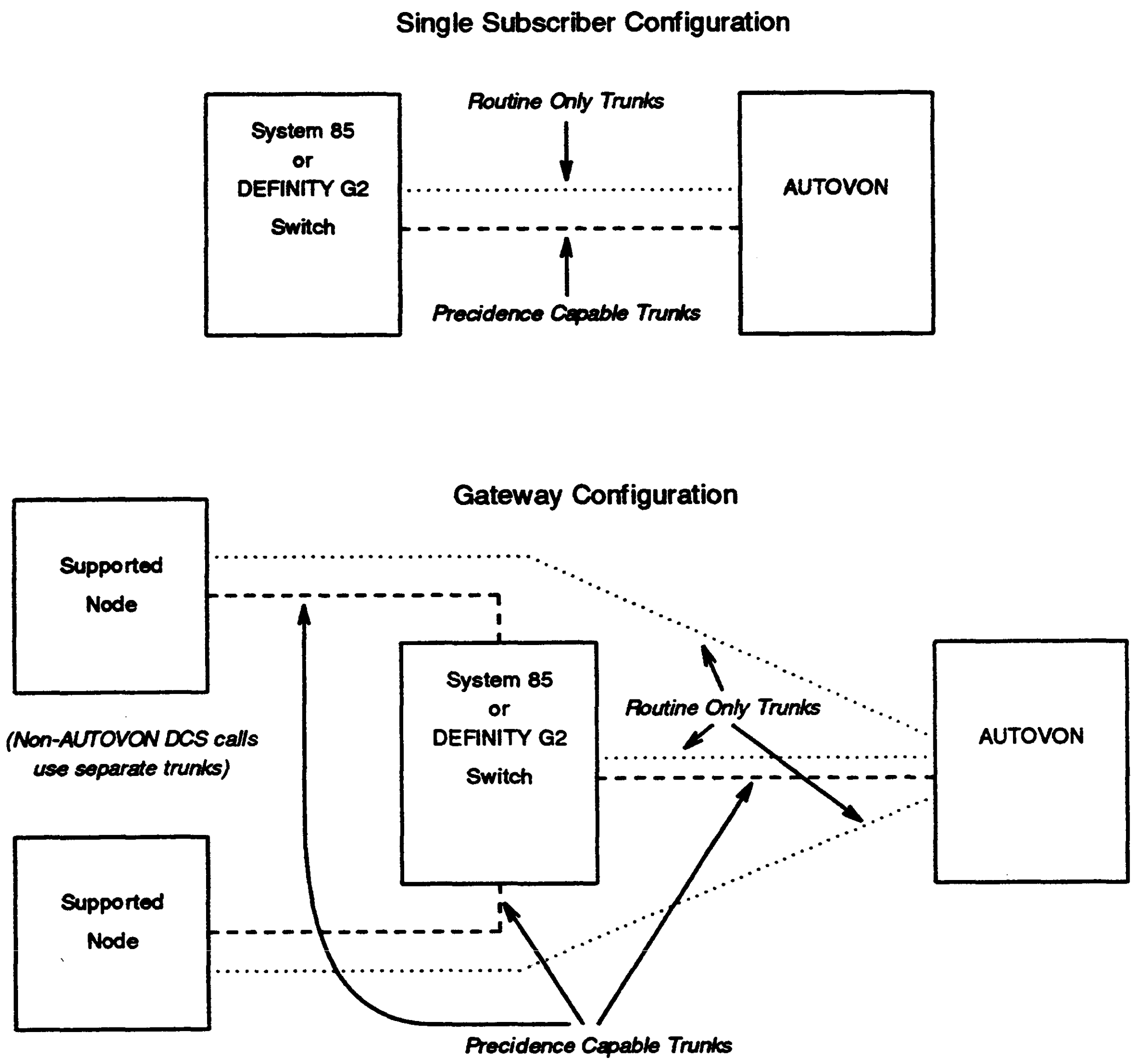

Figure 94-1. AUTOVON Access Configurations

Precedence Levels

In the AUTOVON system, all calls are assigned a precedence. This precedence is then Used to determine what calls can preempt what other calls. The following five precedence levels (in descending order of precedence) are available:

- Flash Override
- Flash
- Immediate
- Priority
- Routine.

NOTE: The term "Priority" in this feature has no connection with the separate and distinct feature Priority Calling. Priority Calling and Priority Precedence are not related in any way.

AUTOVON Access Trunking

As shown in Figure 94-1, the System 85 or Generic 2 switch can be connected to the AUTOVON by both precedence capable and routine only trunk groups. It is important to understand the differences between these types of trunk groups and their significance to precedence calls.

Routine Only Trunk Groups

As the name implies, routine only trunk groups are capable of carrying only routine precedence calls. This means that high precedence calls (priority or higher) will not route over routine only trunks. It also means that a call placed over a routine only trunk group will never be preemptd by a higher precedence call (routine precedence calls will never preempted an existing connection). From the perspective of the Precedence Calling feature, these trunks are just like standard tie trunks. In a Precedence Calling arrangement, routine only trunks are non-precedence capable. Calls processed over routine only trunks do not carry a precedence level digit. The precedence level digit is a DTMF (Dual Tone Multifrequency) signal that is prefixed to the address sting of a precedence call and indicates the precedence level of the call *(see* Table 94-A, *Precedence Level Signaling).* The primary justification for using routine only trunk groups is that the majority of calls will be routine and routine only trunks are less expensive than precedence capable trunks.

Precedence Capable Trunk Groups

Precedence capable trunk groups carry both routine and higher precedence calls. Calls established over precedence capable trunks can preempt lower precedence calls if needed, and can be preempted by higher precedence calls. There can be more than one kind of precedence capable trunk group. Precedence capable trunk groups are assigned a maximum precedence level for outgoing calls in switch translation (Procedure 305, Word 1). This maximum precedence level sets the upper limit on the precedence level of outgoing calls that can use each trunk group. For example, if the maximum precedence level for a trunk group is set at Priority, an outgoing call with a precedence level of Immediate will not route over that trunk group. This is similar to routine only trunk groups in that calls with a precedence equal to the maximum level set for the trunk group can not have the trunk preempted by outgoing calls. Actually, routine only trunk groups are simply trunk groups that have been assigned a maximum precedence level of routine in Procedure 305, Word 1.

Incoming Precedence Calls

The precedence level limit assigned to AUTOVON access trunk groups in switch translation applies to outgoing calls only. Precedence calls coming into the switch (or DCS network) from the AUTOVON are not restricted by these translations. That is, a trunk group with a maximum precedence level of Priority established for outgoing calls can be used to route an incoming call with a precedence level of Immediate or higher. In this case, an established call with a precedence level of Priority, could be preempted if necessary by the incoming call although that call would never be preempted by an outgoing call.

Methods of Access

A caller can initiate Precedence Calling and access the AUTOVON using a dial access code. Callers can also receive attendant assistance for AUTOVON or precedence calls by dialing the attendant access code. The maximum precedence level allowed is assigned to extensions though the extension class of service (Procedure 010, Word 4). If AUTOVON access is not authorized or the precedence level attempted is not authorized to the terminal being used the call is automatically routed to the attendant for assistance.

Precedence Calling is limited to off premises calls using trunk groups assigned specifically for precedence calls (in Procedure 305). Precedence Calling cannot be used for station-to-Station calls.

Preemption

Preemption is one of the capabilities that make Precedence Calling work. The AUTOVON is designed to serve the needs of the defense community under both normal and emergency conditions. Under emergency conditions, the undue delay or failure of a high precedence call is an unacceptable condition. For this reason, when no idle facilities are available, preemption permits a higher precedence call (calls with an established precedence level other than ROUTINE) to arbitrarily disconnect a trunk or line being used by a call of lesser precedence, and then seize that facility for its own use.

Two forms of preemption are available:

Automatic

The system uses the precedence of a call being attempted, and if an existing connection with a lower precedence is using needed facilities, preempts the established call to complete the higher precedence call.

Manual

The attendant can change the precedence level if justified, manually preempt an established connection, or attempt other methods available (such as paging) to complete the call to the desired party or an acceptable substitute.

Incoming Precedence Calls:

For a high precedence call (other than ROUTINE), the switch will automatically preempt an established connection if it can determine that the established connection is of a lower precedence than the incoming call. Otherwise, the incoming precedence call is routed to the attendant for completion assistance (see Diversion). The call may have already preempted a trunk connection to reach the System 85 or DEFINITY Generic 2 switch.

Outgoing Precedence Calls:

For a high precedence call (other than ROUTINE), the switch will automatically preempt a precedence capable tie trunk:

- If all trunks are busy, and
- The switch can determine that one or more of the calls in progress are of lower precedence than the call being placed.

Otherwise, the outgoing precedence call is routed to the attendant for completion assistance.

Once in the AUTOVON system, the call retains its precedence level, and this precedence level continues to be used to preempt circuits as required to complete successfully.

Diversion to Attendant Assistance

The other capability that makes Precedence Calling work is diversion to attendant assistance. If a high precedence call (PRIORITY or higher) cannot be completed automatically within a reasonable period of time, it is routed to the attendant for completion assistance. This applies to precedence calls that cannot be completed for any reason, including calls that reach the called extension but are not answered within a reasonable period of time (30 seconds).

The Attendant Console

A standard 8-character ICI (Incoming Call Identification) display console with DXD (Direct Extension Selection) and BLF (Busy Lamp Field) is used with the Precedence Calling feature. However, to properly handle diverted high precedence calls the console should be administered to meet certain AUTOVON oriented requirements. The 8-character ICI display shows call identification in the left field and the precedence level of the call in the right-hand field.

Trunk Group Selection Area

The trunk group selection area of the attendant console should be set up to provide ready access to precedence capable trunks. The 24 trunk group buttons and associated lamps should be assigned to include precedence capable trunk groups. Figure 94-2 shows an example of how this can be done.

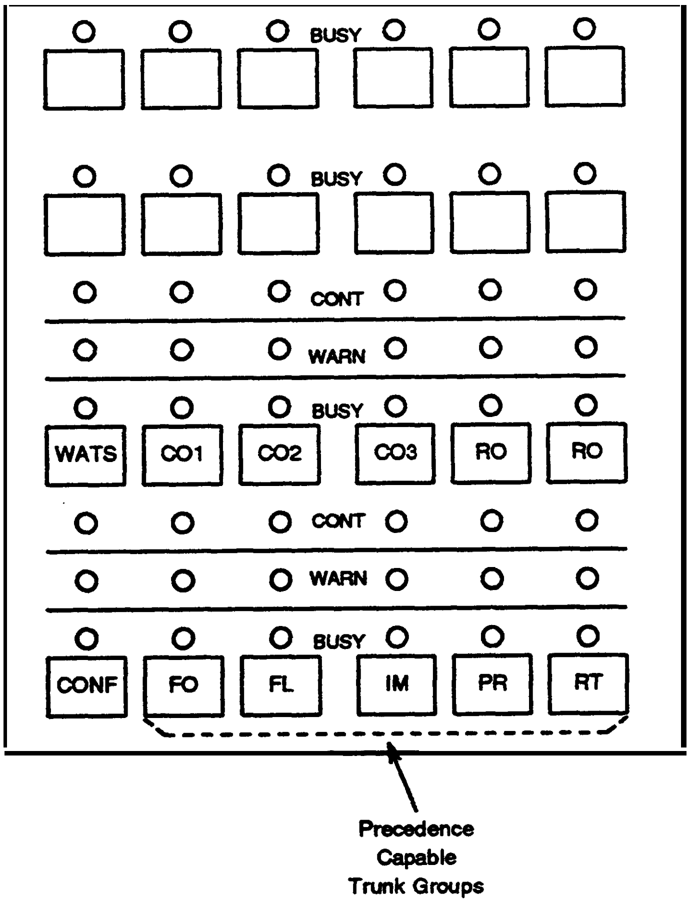

Figure 94-2. Attendant Console Trunk Group Selection Area

Attendant Control Area

In the attendant control area, it is desirable to assign specific precedence oriented control buttons such as:

- PRE (Preempt)

  Used to force preemption of a call when Busy Verification of Liners or Trunks Verification by Attendant feature activation is denied (attendant receives intercept tone). Also used to force the verified trunk or line to hang up when you are bridged onto the conversation in an attempt to complete a precedence call.

- DPL (Display Precedence Level)

  Displays the precedence level of an AUTOVON call and when pressed will redisplay that level any time while the call is on the switched loop of the console.

- FDSA OP (Flash Dial Service Assistance Operator)

  Recalls the DSA operator while the attendant is active on an outgoing AUTOVON call. (The outgoing AUTOVON call must have been routed to the DSA operator by the attendant or the call must have received prior assistance from the DSA operator.)

Figure 94-3 shows an example of the attendant console control area administered for Precedence Calling.

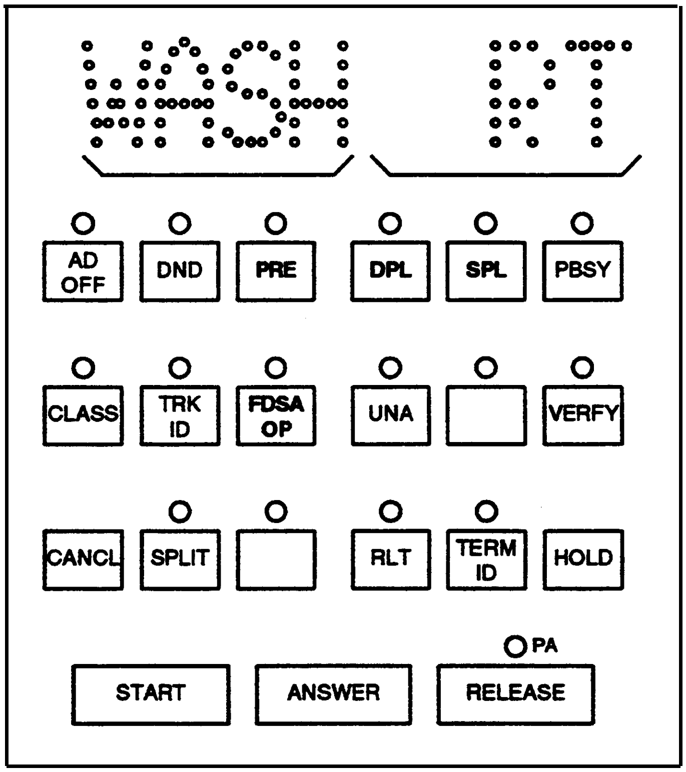

Figure 94-3. Attendant Console Control Area AUTOVON Buttons

DXS (Direct Extension Selection) with BLF (Busy Lamp Field) Area

The DXS/BLF area is also modified for handling precedence calls. In this case the middle six hundreds group buttons (bottom row) take on the additional function of precedence level button. One precedence level is assigned to each button in ascending order from left to right. This arrangement is depicted in Figure 94-4. These buttons are used to establish the precedence level of a call or to display the busy/idle status of trunks (with a corresponding or lower precedence level) in a selected trunk group when the Trunk Verification by Attendant feature is active.

The last (right most) of these button is designated "ALL." This is not a precedence level. The attendant uses the ALL button only during Trunk Verification, to display the busy/idle status of all of the trunks in the trunk group being verified, regardless of precedence level.

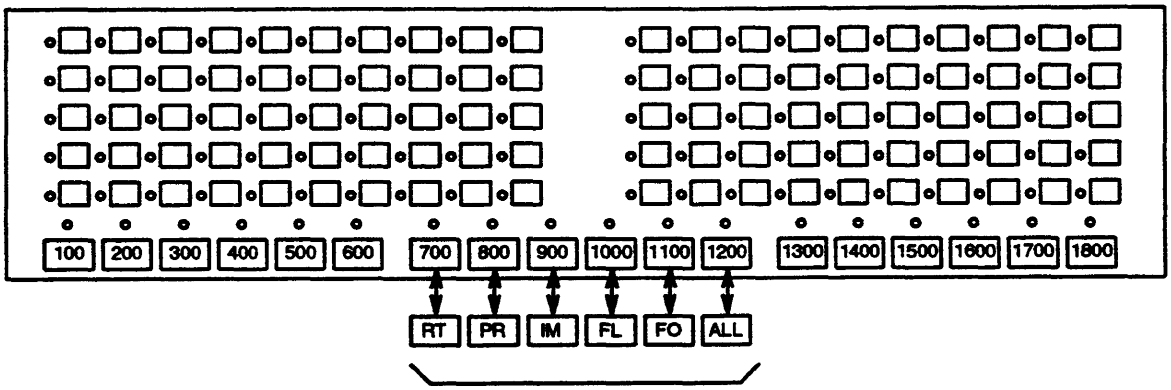

Figure 94-4. Attendant Console DXS/BLF Area with Precedence Calling

Precedence Signaling

The precedence level of an outgoing call is identified by a DTMF signal prefixed to the address digits. The DTMF signal is sent to the AUTOVON over precedence capable trunks only. Routine only trunks do not carry this signal (there is not DTMF signal for routine precedence calls). Calls carried over routine only trunks are assumed be routine precedence calls. Also, routine calls carried over precedence capable trunks do not have a DTMF precedence level signal ahead of the address digits. In this case, the absence of a precedence level digit is used to infer that the precedence level of the call is routine. The precedence level DTMF signals used and their corresponding precedence levels and digit value are shown in Table 94-A.

TABLE 94-A. Precedence Level Signaling

Precedence	Frequencies	Digit
Flash Override	697 Hz+ 1633 Hz	0
Flash	770 Hz+ 1633 Hz	1
Immediate	852 Hz+ 1633 Hz	2
Priority	941 Hz+ 1633 Hz	3
Routine	NONE	4

Incoming calls from the AUTOVON use dial pulse signaling. For these calls, the digits 0 through 4 indicate the precedence level of the incoming call.

The DTMF digits sent between switches to indicate the precedence level of a call are translation independent. That is, they are not affected by the administered digits (Procedure 356, Word 1) used by the caller to specify the precedence level of the call.

Preemption Warning Tone

A Preemption Warning Tone (440-hertz + 620-hertz for 3 seconds) is heard by all parties to a connection that is about to be preempted. If the call is a normal 2-party connection, the connection will be torn down after a 2.5-second timeout unless the parties go on-hook first. If the call is a conference call, only the party being called will be removed, and the remaining conferees can continue normally.

User Operations

The following are the user operating procedures for this feature.

Voice Terminal Procedures

Receiving Precedence Calls at a Voice Terminal

To answer a precedence call received at an idle station:

1. Distinctive ringing (3 bursts of ringing tone) is heard.
2. Go off-hook to answer.

To answer a precedence call from an active 2-party connection:

Preemption warning tone (440-hertz + 620-hertz for 3 seconds) is heard by both parties.

1. Go on-hook. [Station receiving the precedence call hears distinctive ringing.]
2. Go off-hook to answer.

Precedence call received while active on a conference call:

Preemption warning tone (440-hertz + 620-hertz for 3 seconds) is heard by all parties to the connection.

1. All parties remain off-hook. [Called party is automatically disconnected from the conference after 2.5 seconds]
2. When disconnected, go on-hook. [Party receiving the precedence call hears distinctive ringing.]
3. Go off-hook to answer.
4. Remaining conferees can continue with the conference call.

Placing a Precedence Call From a Voice Terminal

Using Direct Outward Dialing:

Precedence calls can be dialed directly from any voice terminal whose class of service is authorized to use the required precedence level.

1. Go off-hook. [Dial tone]
2. Dial the Precedence Calling feature access code. [Recall (second) dial tone]
3. Dial the call precedence level code. [Recall dial tone is silenced.]

   If precedence ringback tone is heard (rapid ringback), the call is being diverted to an attendant for completion assistance.
4. Dial the destination address. [Call-progress tones]

With attendant-assisted calling:

Outgoing precedence calls can be routed to an attendant either by dialing the local attendant directly for assistance *(see* Dial Access to Attendant feature), dialing the AUTOVON Access to Attendant code, or through Diversion. Diversions on an outgoing call takes place when:

- A precedence code is used that is higher than that authorized to the class of service of the voice terminal being used.
- No precedence capable circuits are available, and the precedence level of the call is insufficient to preempt circuits currently in use.

When the attendant answers, provide necessary instructions for placing the call.

Attendant Procedures

A *diverted* precedence call is identified at the console by the PRIO lamp lighting and the ICI display. If the PRIO lamp flashes (at 30 pulses per minute), more than one call is waiting in the priority queue.

To use automatic preemption:

1. Answer the alerting loop by pressing the appropriate loop button or by pressing **[ANSWER]** . [ICI shows current precedence level of the call if it was initiated as a precedence call.]
2. Obtain necessary instructions from calling party.
3. If the precedence level of the call needs to be set (not previously established) or raised:
   a. Press the **[SPL]** (Set Precedence Level) button.
   b. Press the appropriate precedence level button. [Dial tone]

   NOTE: The precedence level buttons can be used at any time before the outgoing circuit has been seized.

Precedence Level Keys are used to:

Establish a precedence level for a call that was not initiated over a precedence capable facility,

or

Raise the precedence level of a call as appropriate

4. Press **[START]** .

5. To extend the call to the final destination either:

   Dial the extension number or trunk-group access code, or press the appropriate DXS or Direct Trunk Group Selection buttons,

   or

   Press the **[RELEASE]** button to allow the calling party to finish dialing,

   or

   Press the **[HOLD]** button to remain connected to the call (if the Serial Calling feature is to be used).

To use manual preemption for an outgoing call:

Like the preceding automatic preemption, a call is received at the attendant console.

1. Press **[START]** .
2. Press **[VERFY]** .
3. Dial the appropriate trunk-group access code.
4. Press a precedence level button lower than the precedence of the call being placed (generally start with ROUTINE). [Busy trunks being used by a corresponding or lower precedence level call are identified by a lighted lamp.]

   NOTE: Operating a precedence level button after the trunk group has been selected does not change the precedence level of the call being placed.

5. Select the trunk to be preempted by pressing the DXS button associated with a lighted lamp.
6. Press the **[PREEMPT]** console button.
7. Complete the call by:

   Dialing the destination number,

   or

   Pressing **[RELEASE]** and allowing the calling party to finish dialing.

8. To remain connected to the call (for Serial Calling or later assistance) press **[HOLD]** .

To recall the AUTOVON DSA (Dial Service Assistance) operator while receiving AUTOVON dial tone:

Press the **[FDSA OP]** button.

NOTE: To recall the DSA operator, the call must have previously received DSA operator assistance.

Considerations

Points of Access to AUTOVON

The Precedence Calling feature is designed around the assumption that there will be a single point of connection (routing pattern) with the AUTOVON. Call routing algorithms used are based on this assumption. While other arrangements are possible (multiple access points), such arrangements are not specifically supported and must be locally engineered.

Precedence Call Processing

The switch will preempt an existing connection when necessary if it can determine that the precedence of the existing call is lower than that of the call it is attempting to complete. The following are examples of how selected call situations will be treated by the switch.

- Station-to-station Connection

  When a call with a precedence of priority or higher is directed toward a station involved in a station-to-station connection, the existing call is assumed to be of routine precedence and will be preempted.

- Station-to-trunk Connection with a Precedence Capable Trunk

  When a priority or higher precedence call is directed toward a station involved in a station-to-trunk connection and the trunk is precedence capable, the switch can determine the precedence level of the existing call. If the existing call is routine precedence the switch will preempt the existing connection. If the existing call is priority precedence or higher, the switch will preempt the connection if the call being attempted is of a higher precedence. Otherwise the switch will divert the new call to the attendant for assistance or redirect the call to a station if one of the call redirection features (Call Forwarding or Coverage is active. A precedence call (other than routine) can not be redirected to a que (other than the attendant priority que) or to a VDN.

- Station-to-trunk Connection with a Nonprecedence Capable Trunk

  When a priority or higher precedence call is directed toward a station involved in a station-to-trunk connection and the trunk is not precedence capable, the switch cannot determine the precedence level of the existing call. In this case, the switch will divert the new call to the attendant for assistance unless one of the redirection features (Call Forwarding or Coverage) is active. This is true for existing DCS calls as well as other types of station-to-trunk connections.

- Incoming Routine Precedence Calls

  When an incoming call has a precedence of routine, the switch will not preempt an existing connection regardless of type. A routine precedence call directed toward a busy station will be treated like a nonprecedence call (redirected if appropriate or busy signal).

AUTOVON Access Trunks

As described earlier, two types of trunks can be used to connect nodes or a DCS gateway (hub) switch to the AUTOVON system. These are ROUTINE Only trunks and Precedence Capable trunks. Calls connected over ROUTINE Only trunks cannot have these trunks preempted by other outgoing calls, nor can they use precedence calling to preempt other calls (in effect, the Precedence Calling feature does not apply). High precedence calls (priority precedence or higher) connected over precedence capable trunks can use the Precedence Calling feature to preempt calls of a lower precedence and can be preempted by calls of a higher precedence.

AUTOVON Trunk Signaling

An additional distinction exists between precedence capably trunks and routine only trunks. Calls sent and received over precedence capable trunks require eight digits. The first digit indicates the precedence level of the call while the last seven digits are address digits. Calls sent or received over routine only trunks use only the seven address digits. If a precedence call is erroneously sent over a routine only trunk, the precedence level digit will be read as part of the address and the last address digit will be chopped. This will result in a failed or misrouted call.

Interactions With Other Features

Abbreviated Dialing

The Abbreviated Dialing feature is fully compatible with the Precedence Calling feature. That is, Precedence Calling dialing sequences (including dial access codes and precedence level codes) can be stored in abbreviated dialing list locations and used to place Precedence Calling calls.

AAR (Automatic Alternate Routing)

On System 85 and DEFINITY Generic 2.1 switches, the AAR feature must be active for the Precedence Calling feature to work properly. To ***enable*** the routing of incoming precedence calls within a DCS, the Standard Networking field (in Procedure 276) and the AAR dial access code (in Procedure 350, Word 2) must be assigned for every node in the AUTOVON access DCS network. The incoming prefix digit (Procedure 103) must be the AAR access code for all AUTOVON and precedence capable intermachine trunks.

However, AUTOVON (Precedence Capable) trunk groups ***must not*** be included in AAR patterns. AAR is not used to route incoming precedence calls within a DCS. AUTOVON routing patterns are separately (from AAR routing patterns) assigned in Procedure 305, Word 1 and Word 2.

ACD (Automatic Call Distribution)

The precedence level of AUTOVON calls that are directed to or forwarded to an ACD split are checked. If the precedence level is higher than ROUTINE, the call does not enter the split's queue. Instead, the call is redirected to the attendant queue.

Attendant Control of Trunk Group Access

This feature functions normally for AUTOVON trunks. When in effect, Precedence Calling calls are routed to the attendant priority queue.

Busy Verification of Lines

The Busy Verification of Lines feature is used by the attendant when attempting to manually preempt an existing connection for a precedence (priority or higher) call.

Call Coverage

The Call Coverage feature functions normally for an incoming precedence call except where group coverage points are used. The incoming call routes to coverage if the established coverage criteria are met. However, an incoming precedence call does not route to a group coverage point (such as Message Center, ACD split, or AUDIX).

Call Vectoring

The precedence level of AUTOVON calls that are directed to or forwarded to a VDN are checked. If the precedence level is higher than ROUTINE, the call does not terminate to the vector. Instead, the call is redirected to the attendant priority queue.

CAS (Centralized Attendant Service)

The Precedence Calling feature is compatible with the CAS feature. That is, these features can be co-resident in the same network. However, the interface to the AUTOVON or DSN switch must be on the CAS main. Using this arrangement, precedence calls that cannot be completed as dialed are routed to a CAS attendant for subsequent handling. Also, the main switch sets timers for incoming precedence calls passing through the CAS main to a CAS branch so that unanswered calls can be diverted to an attendant at the main location. (Unanswered calls to a CAS branch do not inter the CAS queue at the branch locations.)

Once a precedence call has been answered at a branch location, the user at the branch can transfer the call to the CAS queue for subsequent rerouting by a CAS attendant. When this is done, however, these transferred calls lose their preemption capability. These calls are no longer able to override other active calls.

Conference—Attendant Five Party

If a conferee in an attendant 5-party conference is preempted by the Precedence Calling feature, all parties hear preemption warning tone. Then, all conferees must go on hook, and if necessary, the attendant reestablishes the conference.

Conference—Attendant Six Party

If a station or trunk being used on an attendant conference is preempted by the Precedence Calling feature, all parties to the conference will hear preemption warning tone. The affected station or trunk will then be made idle, and the conference call will remain in effect minus the preempted trunk circuit.

Conference—Three Party

If a station or trunk being used on a 3-party conference is preempted by the Precedence Calling feature, all parties to the conference will hear preemption warning tone. The affected station or trunk will then be made idle, and the conference call will revert to an otherwise normal 2-party connection.

Data Call Setup

Imcoming Precedence Calling calls cannot preempt calls connected using the Data Call Setup feature. Precedence calls directed to data call setup connections are routed to the attendant.

Data Protection

The Data Protection feature takes precedence over the Precedence Calling feature. That is, a Precedence Calling call does not preempt a call with Data Protection active. Precedence calls directed to extensions with data protection active are diverted to attendant assistance. The attendant cannot override data protection by using attendant preemption. Data protection permanent should not be administered to an AUTOVON trunk as it invalidates the Precedence Calling feature.

DCS (Distributed Communications Service)

While the AUTOVON access link cannot be a DCS link, and Precedence Calling is not a transparent feature within a DCS, the DCS feature is compatible with Precedence Calling. In a gateway configuration (where one DCS node provides AUTOVON access for the other nodes of the DCS), DCS software provides messaging services that can be useful to attendants and callers. The nodes of a DCS network interfaced to AUTOVON must use direct connections *(see* the DCS feature) to the gateway switch. DCS nodes that do not have direct links to the gateway switch will not have AUTOVON access through the gateway. However, other nodes in the DCS can have direct access to the AUTOVON if they are attached node (setup as single subscriber nodes to the AUTOVON).

Hot Line

The Hot Line feature is fully compatible with the Precedence Calling feature. The Hot Line feature uses the Abbreviated Dialing feature for its automatic dialing function. Therefore, any call that can be placed by using Abbreviated Dialing can be assigned as a Hot Line destination number. When the Hot Line feature is used with Precedence Calling, the precedence level code is part of the automatically dialed digit string and cannot be changed by the caller.

ISDN—BRI (Basic Rate Interface)

While Precedence Calling is not directly compatible with ISDN, BRI terminals can initate and receive precedence calls in the same way as other digital terminals. This is accomplished through the interworking function and precedence calls function normally for BRI stations.

ISDN—PRI (Primary Rate Interface)

The ISDN—PRI feature is not compatible with the Precedence Calling feature. AUTOVON Access trunk groups use APLT trunk types (trunk types 12 to 15). ISDN—PRI trunk groups cannot be assigned APLT trunk types. Also, on switches prior to DEFINITY Generic 2.2, the AAR or ARS feature is required to route ISDN—PRI calls while the AAR feature does not route AUTOVON calls.

Intercept Treatment

Incoming Precedence Calling calls that receive intercept treatment will be given Attendant Intercept, regardless of other options that may be administered.

Last Number Dialed

The Last Number Dialed feature completely stores and redials the digits dialed during a Precedence Calling call.

Look-Ahead Interflow

The Look-Ahead Interflow feature is not compatible with the Precedence Calling feature. AUTOVON Access trunk groups are APLT trunk groups (with Trunk Types 12 to 15). ISDN—PRI trunk groups (required for Look-Ahead Interflow) cannot be assigned as APLT trunk types. Also, precedence calls (other than routine) cannot be routed to a vector.

Override

The Override feature is denied when attempted toward any connection involving a precedence call higher than ROUTINE.

Queuing

Queuing (except attendant queue) is denied on a Precedence Calling call. Precedence calls that cannot preempt the needed facility are directed to the attendant rather than being placed in a facility queue.

Remote Access

Remote Access can be used in conjunction with the Precedence Calling feature if precedence calling is administered to class-of-service 31.

Restriction—Attendant Control of Voice Terminals

This feature functions normally for Precedence Calling calls. Calls to or from a restricted voice terminal route to the attendant priority queue for processing.

Restriction—Miscellaneous Trunk Restrictions

The Miscellaneous Trunk Restrictions feature does not restrict access to Precedence Capable trunk groups within the AUTOVON Access star configuration. However, when trunk-group dial access codes are used to access ROUTINE Only trunk groups, this access can be limited by Miscellaneous Trunk Restrictions.

Route Advance

Route Advance is only used with AUTOVON trunk groups to route Routine Only trunk calls (accessed using a trunk group dial access code) to Precedence Capable trunks when all Routine Only trunks are in use. The absence of the precedence digit is used to infer a precedence level of routine for these calls.

A route advance like function is provided by the Precedence Calling feature when more than one trunk group is assigned the same maximum precedence level (Procedure 305, Word 1). However, in this case the Route Advance feature as such is not used.

Tenant Services

The Precedence Calling feature cannot be used in a partitioned switch.

Trunk Verification—Attendant

The Trunk Verification—Attendant feature is used by the attendant when attempting to manually preempt a busy trunk circuit.

Unattended Console Service—

Alternate Console Position

The Unattended Console Service—Alternate Console Position can be used with the Precedence Calling feature.

The alternate console must be configured with the appropriate precedence calling buttons to allow for effective handling of precedence calls.

Call Answer From Any Voice Terminal

If the Unattended Console Service—CAAVT feature is used for a precedence call that has been diverted to attendant assistance, the service that can be provided to the precedence call is limited to the transfer of the call to some alternate extension or Paging if that feature is available. This feature is not recommended for switches that can expect to receive precedence calls during periods when a regular attendant is not on duty.

Preselected Call Routing

If the Unattended Console Service—Preselected Call Routing feature is used for a precedence call that has been diverted to attendant assistance distinctive (3-burst) ringing is heard at the designated voice terminal. The service that can be provided to the precedence call is limited to the transfer of the call to some alternate extension or Paging if that feature is available. This feature is not generally recommended for switches that can expect to receive precedence calls during periods when a regular attendant is not on duty.

WCR (World Class Routing)

On DEFINITY Communications System Generic 2.2 switches, the WCR feature replaces the AAR feature. The WCR feature performs the same functions for AUTOVON calls as were previously performed by the AAR feature. However, with WCR the standard network bit (in Procedure 276) is not required.

Precedence calls **do not use** WCR routing patterns. However, to ***enable*** the routing of incoming precedence calls within a DCS, a WCR network dial access code (from Procedure 350, Word 2) must be assigned as the trunk group prefix (Procedure 101, Word 3) for all AUTOVON and precedence capable intermachine trunks.

Restricting Feature Use

The basic voice terminal restriction features can be used to restrict access to or from the AUTOVON and the use of the Precedence Calling feature. These features are:

- Restriction—Attendant Control of Voice Terminals
- Restriction—Voice Terminal Restrictions.

Additionally, Precedence Calling (with a maximum precedence level) must be specifically allowed in a voice terminal class of service. Precedence Calling must also be assigned to a specific trunk group.

The Data Protection feature can be used to deny Precedence Calling preemption toward a specific terminal or trunk.

Hardware Requirements

The Precedence Calling feature requires the use of the following specific hardware items.

For Traditional Modules:

- SN233 tie trunk circuit packs (administered as APLT tie trunks; four trunk circuits per SN233)
- SN251 touch-tone dialing receiver/register circuit packs (four circuits per SN251)
- SN252 touch-tone calling sender circuit pack (four circuits per SN252)

For Universal Modules:

- TN760C tie trunk circuit packs (administered as APLT tie trunks; four trunk circuits to a circuit pack)
- TN748C tone detector circuit packs (four touch-tone receivers and two touch-tone senders per TN748C)

Regardless of the Module Type:

- The 8-character display attendant console with DXS/BLF.

Feature Administration

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer has limited capabilities using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using CSM (Centralized System Management).

The following are the applicable administration procedures.

Administration Procedures — Precedence Calling

Procedure	Word	Purpose	SMT
000	1	Assigns the class of service to an extension number.	Yes
010	4	Assigns the maximum precedence level for an extension class of service.	Yes
100	1	Assigns the trunk group characteristics, including trunk type and Route Advance pattern (R2V3 and earlier). The applicable APLT trunk-type encodes include: 12 Delay dial in/(wink start or delay dial) and dial tone out 13 Wink start in/(wink start or delay dial) and dial tone out 14 Delay dial in/dial tone out 15 Wink start in/dial tone out.	No
100	4	Assigns Route Advance (R2V4 and Generic 2).	N/A

(Continued)

Administration Procedures — Precedence Calling *(Continued)*			
Procedure	**Word**	**Purpose**	**SMT**
101	1	Assigns APLT feature allowed to precedence capable trunks.	No
103	1	Displays and assigns FRL and network association for trunk groups. Must set AAR prefix digit (on switches prior to DEFINITY Generic 2.2).	Yes*
103	3	For Generic 2.2 switches, assigns trunk group prefixing.	N/A
150	1	Assigns trunks to trunk groups.	No
203	1	Assigns the AUTOVON buttons to the attendant console. The applicable encodes areas follows: 28 Trunk ID 43 Manual Preemption button 44 Display Call Precedence Level 45 Display Called Number 46 Flash button 47 Set Precedence.	No
204	1	Assigns AUTOVON precedence identification and ICI for trunk groups to attendant console ICI field. Applicable precedence levels and message numbers areas follows: Precedence — R2V3 & Earlier — R2V4 & G2 Flash Override — Msg 300 — Msg 2300 Flash — Msg 301 — Msg 2301 Immediate — Msg 302 — Msg 2302 Priority — Msg 303 — Msg 2303 Routine — Msg 304 — Msg 2304	No
275	3	Assigns system class of service characteristics including local switch number and home NPA. Must be coordinated with the AUTOVON dialing plan when AUTOVON access is involved.	Yes
275	4	Assigns system class of service characteristics including maximum preemption levels for incoming calls, station originated calls, and attendant calls to the system class of service. Also specifies the AUTOVON Interface switch number (use DCS node number).	Yes
276	1	Assigns the AUTOVON feature to the switch. Also, activates standard network (required on System 85 and Generic 2.1 switches).	No
* Display only procedure for the SMT.			

(Continued)

Administration Procedures — Precedence Calling *(Continued)*			
Procedure	**Word**	**Purpose**	**SMT**
305	1	Displays and assigns the AUTOVON destination node (routing) for a a group and specifies the maximum precedence level (0 to 4, or dash) of each AUTOVON trunk group.	Yes*
305	2	Displays and assigns AUTOVON destination node numbering including home office code (NNX). A thousand's digit selects 1000 numbers with the office code to which AUTOVON calls can route, or a "-" allows call routing to all 10,000 numbers.	Yes*
350	1	Assigns the first digit of the AUTOVON dial access code (if required).	No
350	2	Assigns the AUTOVON dial access codes. The applicable encodes are: 82 AUTOVON Precedence Calling 83 AUTOVON Attendant Assistance.	No
356	1	Assigns a dial digit to each precedence level.	No
* Display only procedure for the SMT.			

The following is the applicable TCM path name used with the AP 16.

TCM Screen — Precedence Calling	
Path Name	**Purpose**
terminal-change class-of-service attributes	Assigns Precedence Calling to an extension class of service.

Notes:

Priority Calling

Description

The Priority Calling feature allows a caller to use distinctive 3-burst ringing to place a call. Priority Calling can be used toward an idle single-appearance terminal or an idle appearance of a multiappearance terminal. Priority calling can also be used toward a busy single-appearance voice terminal.

NOTE: There is no correlation between the term "Priority" as used in this feature and "Priority Precedence" as used in the Precedence Calling feature.

Single-Appearance Voice Terminals

Priority Calling allows the caller to signal a busy single-appearance terminal or straight line set. When Priority Calling is used toward a busy single-appearance voice terminal, the called party receives a special 3-burst tone. This tone is three, 0.1-second, 400-hertz beeps. The called party can answer the priority call immediately by using the Hold feature or by finishing the active call and going on-hook.

Priority Calling reduces the chances of a busy single-appearance terminal missing an important call. It also identifies an important call that the called party might otherwise let go unanswered or let go to coverage.

Multiappearance Voice Terminals

For multiappearance voice terminals, Priority Calling provides 3-burst (distinctive) ringing at an idle appearance [including originating (only) appearance]. However, if no idle appearance is available, the switch returns busy tone to the calling party. Unlike the single-appearance voice terminal, a Priority Calling call does not wait on a multiappearance voice terminal.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Make a Priority Call to a Busy Single-Appearance Voice Terminal:

1. Go off-hook or press an idle appearance. [Dial tone]
2. Dial the Priority Calling access code or press **[PRIORITY].** [Second dial tone]
3. Dial the desired extension number. [The called party hears a 3-burst tone. The calling party hears a special ringback tone.]

To Answer a Priority Call When 3-Burst Ringing Is Heard at an Idle Voice Terminal:

Go off-hook or press the ringing appearance. [A 2-party connection is made.]

To Answer a Priority Call While on a 2-Way Connection

Without ending the current 2-way connection:

1. Momentarily press the switchhook,

   or

   Press **[RECALL].** [The second party is put on soft hold. Recall dial tone received.]
2. Dial the answer-hold access code. [The second party is put on hard hold. The called party is connected to the incoming priority call.]

By ending the current 2-way connection:

1. Go on-hook. [The second party is disconnected. The switch rings the called party.]
2. Go off-hook. [A 2-party connection is made.]

To Answer a Priority Call While on a Call at a Multiappearance Voice Terminal Without Ending the Current 2-Way Connection:

1. Press the **[HOLD]** button. [The second party is put on hold. The HOLD green status lamp lights.]
2. Press the appropriate appearance button. [The called party is connected to the incoming priority call.]

Considerations

Attendant Console Calls

Priority Calling is a voice terminal feature and cannot be used from the Attendant Console.

Canceling Priority Calling

If the calling party goes on-hook, priority calling is canceled.

Internal Calling

Priority Calling applies only to internal calls.

Intercept Tone

Intercept tone is heard by the caller when the Priority Calling access code is dialed, and when Priority Calling is either restricted or not assigned.

Busy Tone

Busy tone is heard if Priority Calling is denied. Priority Calling is denied if the *called terminal* is in an unstable talking condition Possible unstable or transient conditions include

- Receiving dial tone
- In a talking state with an attendant
- Dialing
- Receiving ringback, busy, reorder, or intercept tone
- Momentarily pressing the switchhook to put another call on soft hold
- Receiving ringing.

Priority Calling to a busy terminal is also denied if the *calling terminal* is holding a call in the soft hold state. However, Priority Calling to an idle terminal is allowed if the calling terminal has a call on soft hold or to a busy terminal if the calling terminal has a call on hard hold.

Hard and Soft Processor Swaps

If a priority call is waiting on a busy single-appearance terminal when a hard processor swap occurs, the waiting call does not endure the hard swap.

The Priority Calling feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Automatic Callback

If a busy party has a priority call waiting and another party tries to call the busy party, the switch returns busy tone to the calling party. The calling party can now activate

Automatic Callback toward the busy party. However, the callback sequence is delayed until there are no calls waiting.

If a calling party (with a single-appearance terminal) activates Automatic Callback toward a busy terminal and then becomes busy, the switch does not allow activation of the Priority Calling feature ***by another party*** toward the original calling party. However, when the calling party (that originally activated Automatic Callback) is using a multiappearance voice terminal, the switch uses Priority Calling for the callback call when the called party goes on-hook. In this case, if the activating party (the original calling party) is busy, the callback call provides 3-burst ringing and rings on an idle originate only appearance if all other appearances are busy.

If the original calling party is using an analog set (single appearance voice terminal), the switch does not use priority calling, rather it waits for the caller to go on-hook.

ACD (Automatic Call Distribution)

When Priority Calling is activated toward a busy individual terminal in an ACD split, the call waits on that terminal. When Priority Calling is activated toward a busy ACD split, the call always waits on the controlling terminal (the call does not enter the split's queue).

Priority calls cannot wait on an ACD agent's single-appearance voice terminal while an observer (using agent override) is connected to the agent's call.

Bridged Call

Priority Calling is partially allowed toward straight line sets. Straight line sets are single-appearance voice terminals that share their extension with a multiappearance voice terminal and the multiappearance voice terminal has only one appearance of the shared extension. When only the straight line set is active on this type of shared extension, the switch allows a Priority Call to wait. However, when the multiappearance terminal is active on the shared extension, the incoming Priority Call is denied. Busy tone is returned to the calling party.

While a priority call is waiting on a straight line set, the multiappearance voice terminal can be used to bridge onto the active call. However, the waiting call cannot be retrieved until the multiappearance voice terminal leaves the connection.

Priority Calling is partially allowed toward shared extensions with more than one appearance. Terminating priority calls are routed to an idle appearance (if available), and 3-burst ringing is provided. When every appearance is busy, the switch returns busy tone to the calling party.

Busy Verification of Lines

A verification attempt made via the Busy Verification of Lines feature has precedence over a priority call on the line being verified. However, if the attendant attempts to busy verify a terminal line that is waiting for another line, the busy verification attempt is temporarily denied and routed to reorder tone.

Call Coverage

When using Priority Calling to place a call to an extension that has Call Coverage active, the call does not redirect to coverage. The call rings (with 3-burst ringing) or waits (with 3-burst tone) at the originally called extension. When every appearance of a called multiappearance voice terminal is busy, the switch returns busy tone to a priority caller.

The Consult / Return option of the Call Coverage feature enables covering users to consult with and return calls to principals. Consult / Return calls are treated as priority calls that override coverage. Like other priority calls, Consult / Return calls can terminate to an originating (only) appearance.

Call Forwarding—Busy and Don't Answer

When Call Forwarding is in effect, the forwarding operation occurs before Priority Calling is allowed. There are four possible operations. If the originally called voice terminal is busy and the forwarded-to voice terminal is idle, calls forward to and ring at the forwarded-to voice terminal. If the originally called voice terminal is busy and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice teminal is idle, calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal.

Call Forwarding—Don't Answer

When Call Forwarding is in effect, the forwarding operation occurs before Priority Calling is allowed. There are two possible operations. If there is no answer at the originally called voice terminal and the fowarded-to voice terminal is idle, calls forward to and ring at the forwarded-to voice terminal. If there is no answer at the originally called voice terminal and the forwarded-to voice terminal is busy, calls forward to and wait on the forwarded-to voice terminal.

Call Forwarding—Follow Me

When Call Forwarding is active at the called voice terminal, the forwarding operation occurs before Priority Calling is allowed There are two possible operations. Calls forward to and ring at the forwarded-to voice terminal (if this voice terminal is idle). Otherwise, calls foward to and wait on the forwarded-to voice terminal (if this voice terminal is busy).

Call Park

A Priority Call is denied when the Call Park feature is active at the called voice terminal.

Call Vectoring

Priority calls cannot be placed to VDNs. When this is attempted, the switch returns intercept tone.

Call Waiting

The Call Waiting feature is an ***independent*** companion feature to the Priority Calling feature. Call Waiting allows an incoming call to a busy single-appearance voice terminal to wait for the called terminal to go on-hook. The called party receives one or two 400-hertz beeps when a call is waiting. At one time (on the DIMENSION® PBX), Priority Calling was called *Call Waiting - Originating.*

Code Calling Access—Universal

A call is not allowed to wait on a single-appearance voice terminal that has accessed code calling.

Conference—Attendant Five Party

Priority Calling is denied when the called terminal is involved in an attendant established conference call.

Conference—Attendant Six Party

Priority Calling is denied when the called terminal is involved in an attendant established conference call.

Conference—Three Party

When calling a single-appearance voice terminal in a conference created by the Conference—Three Party feature using the Priority Calling feature, the call is denied.

When calling a multiappearance voice terminal in a conference created by the Conference—Three Party feature using the Priority Calling feature, the call completes to an idle appearance. If no appearance is idle, the call is denied.

Data Protection

Priority Calling is denied when the Data Protection feature is active on a call.

DDC (Direct Department Calling)

When Priority Calling is activated toward a busy individual DDC group terminal, the call waits on that terminal. When Priority Calling is activated toward a busy DDC group, the call always waits on the controlling terminal (the call does not enter the group queue).

EUCD (Enhanced Uniform Call Distribution)

When Priority Calling is activated toward a busy individual terminal in an EUCD split, the call waits on that terminal. When Priority Calling is activated toward a busy EUCD split, the call always waits on the controlling terminal (the call does not enter the split's queue).

Hold

Priority Calling is denied when either soft hold or hard hold is active at a called single-appearance voice terminal.

Priority Calling is also denied if the called terminal is busy and the calling terminal has a call on soft hold. However, Priority Calling is allowed when the called teminal is idle and when the calling terminal has a call in the hard hold state.

Hunting

Hunting is not performed when Priority Calling is activated toward a terminal line in a hunting group. The priority call either waits on a single-appearance voice terminal or terminates to an idle appearance on a multiappearance voice terminal.

ISDN—BRI (Basic Rate Interface)

Priority Calling works for ISDN—BRI terminals in essentially the same way as it does for other multiappearance voice terminals. The only exception is if a callled BRI terminal has a *Prime Data Line* or *Data Appearance* that is idle. A Priority Call will not ring at a Prime Data Line or Data Appearance, even if all other appearances are busy.

IPA (Interpartition Access)

A voice terminal user (if an extension partition other than Extension Partition 0) is allowed to place priority calls within the user's partition group or to Extension Partition 0. When the user tries to activate Priority Calling toward an extension in any other partition group, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can place a priority call to any other extension in the switch.

Whenever Priority Calling is allowed, the switch provides 3-burst ringing or 3-burst waiting tone for the called voice terminal.

Last Number Dialed

When a user places a priority call, the Priority Calling access code is not stored and redialed by the LND (Last Number Dialed) feature. The extension number dialed after the second dial tone is stored in LND memory.

Line Lockout

Busy tone is returned to a calling party who is attempting a priority call toward a voice terminal in lockout.

Loudspeaker Paging Access

A call is not allowed to wait on a line that has accessed Loudspeaker Paging.

Malicious Call Trace

If a priority call is attempted toward a line involved in a Malicious Call Trace, the calling party receives busy tone.

Override

Override is not allowed toward a line which is waiting but is allowed toward the 2-party call that has a call waiting.

Queuing

The callback sequence associated with Queuing is delayed until there are no waiting calls.

Recorded Telephone Dictation Access

Priority Calling is denied when Recorded Telephone Dictation Access is active at the called terminal.

Restriction—Attendant Control of Voice Terminals

If an attendant restricts call termination to a voice terminal, Priority Calling is also resticted to that terminal.

Restriction—Voice Terminal Restrictions

A voice terminal that is assigned the Termination Voice Terminal Restriction cannot receive Priority Calling calls.

Tenant Services

A voice terminal user (in an extension partition other than Extension Partition 0) is allowed to place priority calls within the user's partition or to Extension Partition 0. When the user ties to activate Priority Calling toward an extension in any other partition, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can place a priority call to any other extension in the switch.

Whenever Priority Calling is allowed, the switch provides 3-burst ringing or 3-burst waiting tone for the called voice terminal.

Transfer

Priority Calling is denied toward a single-appearance terminal with a Transfer in progress.

When the Priority Calling feature is activated toward a multiappearance terminal with a Transfer in progress, the call completes to an idle appearance. If no appearance is idle, the priority call is denied.

Trunk Verification-Attendant and Voice Terminal

If a trunk is being held or answered by a terminal using the answer-hold code of the Priority Calling feature, Trunk Verification is denied.

UCD (Uniform Call Distribution)

When Priority Calling is activated toward a busy individual UCD group terminal, the call waits on that terminal. When Priority Calling is activated toward a busy UCD group, the call always waits on the controlling terminal (the call does not enter the group's queue).

Hardware Requirements

None.

Feature Administration

Assignment of the Call Waiting feature enables the Priority Calling feature within the switch. Assignment of the Priority Calling feature is on a per-terminal class-of-service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — PRIORITY CALLING			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the extension class of service to an extension number.	Yes
010	1	Assigns Priority Calling to an extension number class of service.	Yes
054	2	Administers the Priority Calling button to a multiappearance voice terminal. The applicable encode is as follows: 2 Priority Calling.	Yes
275	1	Assigns Call Waiting to the system class of service.	Yes
350	1	Assigns the first digit of the dial access code (if required).	No
350	2	Assigns the dial access code for the Priority Calling feature. The applicable encode is as follows: 7 Priority Calling.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — PRIORITY CALLING	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns Priority Calling (Call Waiting Originating) to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change terminal buttons	Assigns the Priority Calling button (cwo) to a multiappearance voice terminal.

Privacy — Attendant Lockout

Description

This feature prevents the attendant from reentering (bridging on to) a 2-party connection held on the console, unless recalled by a voice terminal user.

Feature History and Development

This feature was fit available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Initiate Attendant Lockout:

1. Calling party is connected to an attendant [ATND lamp lights.]
2. Press **[START]** . [Calling party is split away from the attendant, the SPLIT lamp lights, and dial tone is heard.]
3. Dial the called party's extension. [Ringback tone is heard, and the RING lamp lights.]
4. If the attendant presses **[HOLD]** before the called party answers: [HOLD lamp lights. The ATND lamp goes out.]
   a. The attendant is released from the call. [Calling party hears ringback tone.]
   b. Called party answers the call. [The calling and called parties are connected together, the ANS lamp lights, the RING lamp goes out, and Attendant Lockout is activated.]
5. If the attendant does not press **[HOLD]** before the called party answers:
   a. The attendant and the called party are connected [ANS lamp lights.]
   b. Attendant announces the call.
   c. Attendant presses **[HOLD]** . [The called and calling parties are connected, the HOLD lamp lights, and Attendant Lockout is activated.]

To Recall the Attendant:

1. Be sure there is a 2-party connection. [HOLD and ANS lamps are lit on the console]
2. Press the **[RECALL]** button,

   or

   Momentarily press the switchhook. [The ANS lamp flashes, and ringback tone is heard by both parties.]

To Reenter a Previously Locked-Out Call

— After one party has gone on-hook: [ANS lamp goes out]

Press the appropriate loop button.

— After one party has pressed **[RECALL]** or momentarily pressed the switchhook: [ANS lamp flashes.]

Press the appropriate loop button.

Considerations

When Attendant Lockout Applies

The Attendant Lockout feature applies when:

- The two parties are on the same attendant console switched loop.
- One of the parties is on a local line port (as opposed to a trunk).
- The voice terminal user has not gone on-hook.
- The voice terminal user has not momentarily pressed the switchhook causing attendant recall.

Reentering Calls

The attendant can reenter a call before the called party has answered. If the called party answers while the attendant is on the line, the attendant is split away. The attendant then presses the HOLD button and is released. Attendant Lockout is then activated.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Auto—Manual Splitting

Attendant Recall Privacy works with the Splitting feature to cause automatic splitting. Manual splitting / unsplitting is disabled. The Privacy feature overrides the Attendant Auto—Manual Splitting feature, and the SPLIT button is not fuctional. Privacy is essentially a form of automatic splitting that prevents the attendant from bridging onto a talking connection. The Attendant Lockout feature does not override splitting; however, it denies the attendant the ability to reenter an established connection held on the console, unless recalled by a voice terminal.

Busy Verification of Lines

The attendant is prevented from busy verifying an extension that is connected to a loop held on the attendant console.

Conference—Attendant Five Party

When the attendant reenters a conference connection in answer to a recall, the attendant connects to all conferees. Privacy is denied.

Conference—Attendant Six Party

When the attendant reenters a conference connection in answer to a recall, the attendant and all conferees are connected Attendant Recall Privacy is denied.

Serial Calls

When Attendant Lockout is assigned, the attendant cannot reenter a trunk-to-terminal/terminal-to-trunk serial call without being recalled. Attendant Lockout is automatically disabled for trunk-to-hunk serial calls.

Tenant Services

The Attendant Lockout feature is not partitioned. This feature is assigned to the entire switch in Procedure 200, Word 1. If assigned, no attendant in the switch can reenter a call held on the console. If not assigned, every attendant in the switch can reenter a call held on the console.

Timed Recall on Outgoing Calls

If Attendant Lockout is provided, the attendant cannot release after a timed recall by pressing the HOLD button.

Trunk-to-Trunk Connections

Attendant Lockout is automatically disabled for trunk-to-trunk calls.

Hardware Requirements

None.

Feature Administration

Assignment of the Privacy—Attendant Lockout feature is on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE — PRIVACY—ATTENDANIT LOCKOUT		
PROCEDURE	**WORD**	**PURPOSE**
200	1	Administers the Privacy-Attendant Lockout feature.

Privacy — Manual Exclusion

Description

The Privacy—Manual Exclusion feature prevents other terminal users with an image of the same apppearance from bridging onto an active call on that appearance. Activating Manual Exclusion does not interrupt the call in progress.

Privacy—Manual Exclusion is useful at voice terminals where the user has an occasional need for assured privacy but a general need for other users to pick up (bridge onto) calls. This feature is transient in that it applies only to the specific call for which it has been activated. When that call ends, the feature activation is canceled.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Activate Manual Exclusion:

1. Be sum there is a 2-party connection and Manual Exclusion is not activated. [Green status lamp is not lit.]
2. Press **[XCLUSION]** .[Green status lamp lights. Manual Exclusion is activated.]

To Deactivate Manual Exclusion:

1. Be sure there is a 2-party connection and Manual Exclusion is activated. [Green status lamp is lit.]
2. Go on-hook or press **[XCLUSION]. [Green** status lamp goes out.]

Considerations

Separate Exclusion Buttons

Each extension number on a given terminal requiring the Manual Exclusion feature must be assigned a separate XCLUSION button. Manual Exclusion is limited to one appearance of an extension. It cannot be assigned to the same extension number on another terminal.

Activating Manual Exclusion

Privacy—Manual Exclusion can be activated for the originating appearance, the terminating appearance, or both.

If both the originating and the terminating appearances are shared with another terminal, Manual Exclusion must be activated by both the calling party and the called party to ensure complete privacy for the call.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Bridged Call

Bridging onto an appearance that has Manual Exclusion active is not allowed. The switch returns reorder tone.

If a party is already bridgd onto a call when Manual Exclusion is activated, the switch removes the bridged party from the connection and returns reorder tone.

Busy Verification of Lines

Activating Manual Exclusion on an extension does not prevent an attendant from busy verifying the line.

Hold

An appearance which is placed on hold while Manual Exclusion is active remains in Manual Exclusion.

Intercom—Automatic and Intercom—Manual

An automatic or manual intercom cannot be manually excluded.

Override

The switch allows an Override call to complete even if Manual Exclusion is active for the called extension. The Override call terminates to an idle appearance (if available). If an idle appearance is not available, the Override call enters the connection and warning tone is provided.

Personal Central Office Line

Manual Exclusion cannot be applied to a Personal CO Line appearance.

Hardware Requirements

Multiappearance voice terminals are required for this feature.

> **NOTE:** The Manual Exclusion feature, requiring an XCLUSION button, cannot be assigned to a straight line set.

Feature Administration

Assignment of the Privacy—Manual Exclusion feature is on a per-terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can fully administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE — PRIVACY—MANUAL EXCLUSION			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
054	1	Assigns the XCLUSION button to a multiappearance voice terminal. The applicable recode is as follows: 8 Manual Exclusion	Yes

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — PRIVACY—MANUAL EXCLUSION	
PATH NAME	**PURPOSE**
terminal-change terminal buttons	Assigns the XCLUSION button to a multiappearance voice terminal.

Notes:

Queuing

Description

When all the accessible routes (trunk groups) for an outgoing call are busy, Queuing allows the switch to hold the call waiting for a trunk to become available. A queue is an odered sequence, in this case of outgoing calls, waiting to be processed. During periods of high call activity, queuing improves caller efficiency by reducing or eliminating repeat dialing attempts. It can also improve trunk utilization by maintaining high trunk-group occupancy, allowing fewer trunks to carry the same call volume. Queuing can be assigned to any external trunk type including tie trunks, central office trunks, host access, and Wide Area Telecommunications Service trunks.

Feature History and Development

This feature was first available on System 85 in Release 1. In System 85, Release 2, Version 3, pattern queuing was introduced.

Types of Queuing

Each trunk group can have its own queue that is either ringback or off-hook. Based on administration, a specific trunk group can have more than one queue *(see* Queue Combinations).

Ringback Queuing

With ringback (some times called on-hook) queuing, the caller goes on-hook (hangs up) and waits for a ringback call from the switch when an idle trunk becomes available. If a main or satellite location has ringback queuing, the caller must dial his / her extension number to receive the ringback call.

Off-Hook Queuing

With off-hook queuing, the calling party waits off-hook for the next available trunk. As an option, music or a recorded announcement can be provided while the calling party waits off-hook.

Queuing Parameters

The Queuing feature, including type of queuing and queuing parameters are administered using Procedure 330. The administrable parameters for the Queuing feature include the following.

Priority Queueing

For off-hook queuing, a trunk group can be assigned two queues: a ***priority queue*** and a ***nonpriority queue.*** The switch serves calls in the nonpriority queue only when the priority queue is empty. Ringback queuing is always nonpriority.

Pattern Queuing

Pattern queuing was added in System 85, R2 V3. Pattern queuing applies only to calls placed through one of the networking features (AAR, ARS, or WCR). With these features, a call may have access to a series (pattern) of trunk groups (preferences). The pattern queuing option allows a specified number of preferences (trunk groups) in the pattern to be checked for an idle trunk during the entire queuing process. The switch may check the first preference only (ignoring pattern queuing), or the first two or more preferences (up to a maximum of 16). Pattern queuing is assigned using field 7 of Procedure 330, Word 1. This field sets the depth of pattern queuing allowed. If this field is set to "0," no queuing is allowed The depth of preferences that can be queued on is "0." In this case, the Queuing feature should be turned off (a "0" in field 1). If field 7 is set to "1," standard queuing is in effect (as opposed to pattern queuing). If field 7 is set to some number between 2 and 16, pattern queuing is in effect and the number in this field represents the maximum number of preferences (trunk groups) in the queuing pattern. That is, if pattern queuing indicates that three of ten preferences are to be checked, the first three preferences are screened for an idle trunk during the queuing process. However, the switch checks all ten preferences during the "last try." When a call is placed in queue, it queues on the first accessible preference (or best choice preference) and is restricted by that trunk group's queuing parameters (queue length and time-in-queue limits).

CAUTION: *Care must be exercised when setting the number of preferences to be included in Pattern Queuing. Increasing the number of preferences to be checked means an increase in processing time. If this added processing time does not produce a significant increase in calls served, queues could begin to overflow.*

FRL (Facility Restriction Level) Raising

The FRL raising function applies only to calls that are placed using one of the networking features, AAR, ARS, or WCR. Just before the "last try," when the time-in-queue limit elapses for the queued call, the switch can raise the call's FRL to help provide an allowable trunk for the call. FRL Raising is assigned on a per-system basis in Procedure 330, Word 1, field 4.

FRL Raising first compares the timed-out call's current FRL with the assigned ***Threshold FRL*** (Field 3). If the call's current FRL is equal to or greater than the Threshold FRL, the call is qualified for FRL Raising. At this time, the switch considers substituting the assigned Raised FRL (Field 4) for the timed-out call's current FRL. Substitution is made if the Raised FRL will be higher than the current FRL.

Queue Length Limit

The queue length limit is set for specific trunk groups in fields 3 and 4 of Procedure 330, Word 2. This limit is set separately for priority and nonpriority queues. That is, the priority queue for a specific trunk group has a separate queue length limit from the nonpriority queue. In both cases, the length limit can be set anywhere between "0" and "63." Setting a queue length limit to "0" has the effect of turning queuing off for that queue for that trunk group.

Time-in-Queue Limit

The amount of time a call can remain in queue can be either unlimited or limited by a time-out value. Trunk group queues are assigned a time-out value in fields 5 through 7, based on type of queue. The minimum wait time is 0.1 minutes for off-hook queuing and 1 minute for ringback queuing. Time-in-queue limits can be set in increasing increments up to 7.9 minutes (79 increments of 0.1 minutes each) for off-hook queues or 60 minutes (60 increments of 1 minute each) for ringback queues. Beyond the time-out values of 7.9 minutes for off-hook queues and 60 minutes for ringback queues, the wait time becomes indefinite (no limit). The queue time-out options are as follows:

- Ringback queue with no time-in-queue-limit available for terminals only
- Ringback queue with a time-in-queue limit of 1 to 60 minutes in 1-minute increments available for terminals and tie trunks; not available for attendants
- Off-hook queue with time-in-queue limit of up to 4 minutes in 0.1-minute increments available for terminals and trunks. Time-in-queue limits do not apply to attendant calls (attendants do not time out).

Trunk Group Dial Access Calls

For calls that are originated using a trunk group dial access code, the switch drops a call from queue when it exceeds the time-in-queue-limit.

Network Calls

For calls that are originated using one of the networking features (AAR, ARS, or WCR), the switch makes one final attempt to find an available trunk before tipping the call. Any toll restriction which may have been applied is removed and the call becomes toll-allowed. The switch compares the caller's default FRL with a preset threshold and raises the FRL to a higher (also preset) level if it is greater than or equal to the threshold. Calls in a ringback queue are dropped without notice to the caller. Callers waiting in an off-hook queue receive a tone warning them that their call is about to be dropped.

Serving the Queue

The switch processes calls in queue on a first-in, first-out basis. For off-hook queuing, the switch always serves the head-of-queue (the first call to enter the queue) first. This, however, is not always true for ringback queuing.

A person with a call in a ringback queue can place and receive other calls while awaiting ringback. If the head-of-queue is busy on another call or does not answer the ringback call, the switch tries to ring the next station with a call in queue. If the switch finds an idle line and services the call, the next caller in queue becomes the starting point for the next search. Callers who are busy or do not answer the ringback lose their place in queue. The switch removes a call from queue if a terminal is busy during two ringback attempts in a 3-minute period.

User Operations

The following are the user operating procedures for this feature.

Off-Hook Queuing

Off-hook queuing is provided automatically by the switch. The only operation required of the user is to remain off-hook.

To Enter a Ringback Queue

Using an Access Code (Trunk-Group, AAR, ARS or WCR):

1. Go off-hook. [Dial tone]
2. Dial the appropriate access code (Trunk-Group, AAR, ARS, or WCR). [If a trunk group DAC is used, confirmation tone is head If a network DAC is used, Second dial tone is heard.]
3. Dial the desired telephone number. [Confirmation tone]
4. If calling from a subtending switch, dial your voice terminal's extension number.
5. Go on-hook within 4 seconds.

To Complete a Queued AAR, ARS or WCR Call:

1. Wait for the queuing callback. [The switch is scanning for an idle trunk.]
2. Receive the queuing callback (3-burst ringing for a local terminal).
3. Go off-hook before the system-wide Don't Answer Timing Interval elapses.

   NOTE: If reorder tone is heard upon going off-hook, the reserved trunk was already seized at the distant end. To return to the head of queue, go on-hook.

4. Wait for the switch to outpulse the previously dialed digits. [Ringback tone]

   NOTE: If reorder tone is heard after the switch outpulses the digits, the call was blocked during routing. Go on-hook, and try the call later.

To Complete a Queued Trunk-Group Access Call:

1. Wait for the queuing callback. [The switch is scanning for an idle trunk.]
2. Receive the queuing callback (3-burst ringing for a local terminal).

3. Go off-hook before the system-wide Don't Answer Timing Interval elapses. [Dial tone]

   NOTE: If reorder tone is heard upon going off-hook, the reserved trunk was already seized at the distant end. To return to the head of queue, go on-hook.

4. Dial the destination telephone number. [Ringback tone]

   NOTE: If reorder tone is head after the switch outpulses the digits, the call was blocked during routing. Go on-hook, and try the call later.

To Remove Yourself From a Ringback Queue:

1. Go off-hook, while the switch is scanning for an available trunk. [Dial tone]
2. Dial the Cancel Outgoing Trunk Queuing access code. [Confirmation tone]
3. Go on-hook.

Flow Diagram

To help conceptualize this complex feature, a flow diagram describing the Queuing feature is provided in Figure 98-1. This diagram does not show all of the decisions made in the Queuing software. The diagram does, however, unify many of the different Queuing functions.

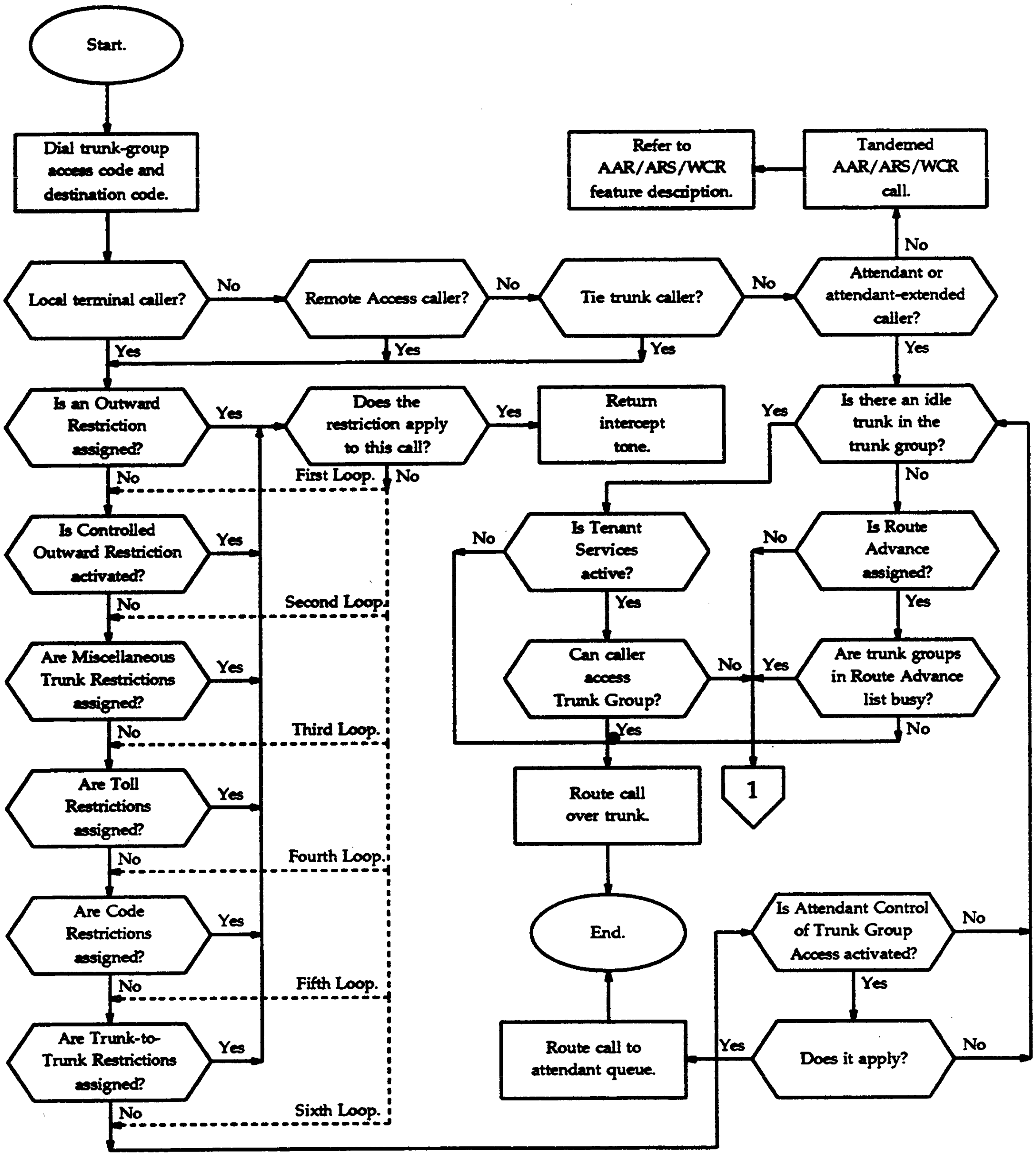

Figure 98-1. Queuing Process Flow (Sheet 1 of 3)

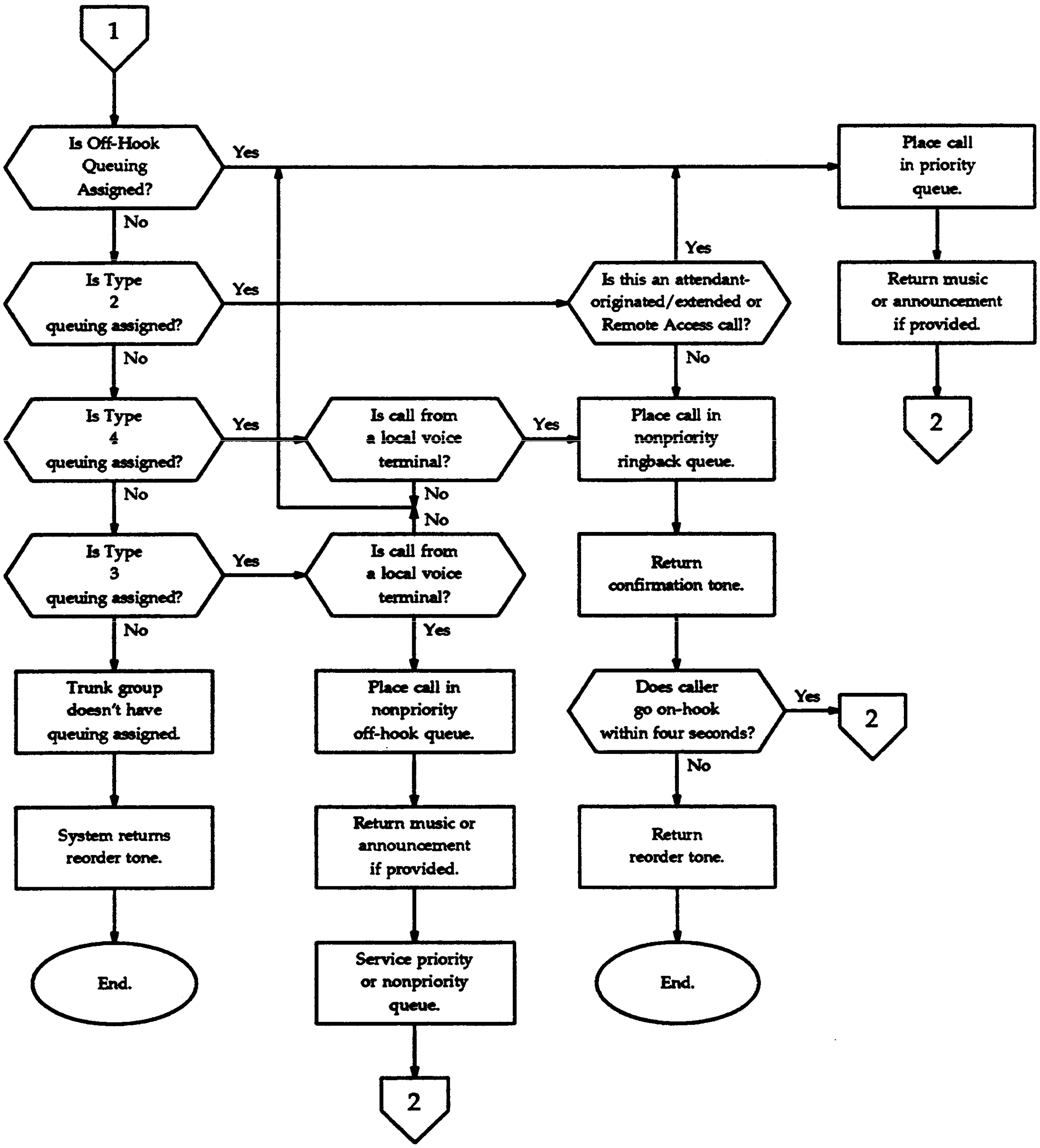

Figure 98-1. Queuing Process Flow (Sheet 2 of 3)

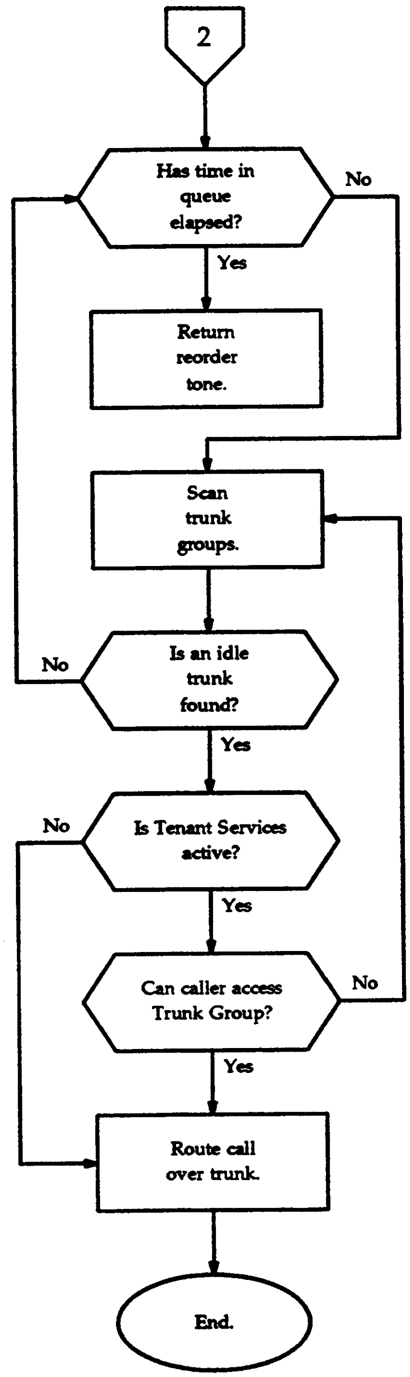

Figure 98-1. Queuing Process Flow (Sheet 3 of 3)

Considerations

Attendant Originated/Assisted Calls

Attendants cannot use ringback queuing and there is no time-in-queue limit for attenant calls of an off-hook queue.

ETN Queuing

In general, all tandem switches within an ETN (Electronic Tandem Network) have the same type of queuing. However, combinations are possible For example, one node of an ETN can have both off-hook and ringback queuing, while another node may have only off-hook queuing.

Network Call Queuing

An AAR or ARS call can only queue on the first trunk group in a routing pattern. A WCR call can queue on the "best" choice preference within the routing pattern. The best choice preference will depend on the characteristics of the call (including such factors as Call Category, FRL, and BCCOS), and may not be the first preference in the routing pattern.

A call cannot be placed in queue if the caller has another call on hold, has another call queued, or has a call waiting. Also, the switch does not queue IDDD (International Direct Distance Dialing) calls.

Ringback Queuing Restrictions

A call from a local terminal cannot enter a ringback queue if any of the following features or restrictions are active at the calling terminal:

- Attendant Control of Terminal Access
- Manual Terminating Line
- Automatic Callback
- Termination Restriction.

When the switch attempts to ringback a caller waiting in queue on a 2-way trunk group, it reserves a trunk for the ringback call. However, this reservation only applies on the switch setting up the reservation. There is a chance the reserved trunk may be seized from the far end (before the local switch places the ringback call), leaving no trunks for the ringback call. If this happens, the call is requeued at the head of the queue. To prevent this, a 2-way trunk group can be assigned a ringback restriction so that ringback queuing is not allowed, or ringback queuing is changed to off-hook queuing.

Tie Trunk Queuing

The following considerations apply specifically to queuing parameters and types assigned to Main / Satellite and tandem tie trunk arrangements.

- Off-Hook Queuing

  Off-hook queuing should only be assigned at the tandem end of an access tie trunk. This reduces the blocking rate for calls from other tandem switches, but increases blocking for calls from subtending Main / Satellite locations. With off-hook queuing calls can queue at more than one switch before completion.

- Ringback / Priority Off-Hook Queuing

  When a call from a subtending (satellite / tributary) location enters a ringback queue at the main, the main has to seize a tie trunk to ring back the subtending location. If all the tie trunks are busy, the call moves to the bottom of the queue. Because this may cause blocking, ringback / priority off-hook queuing is only recommended for 1-way outgoing trunks at the Main. Furthermore, a Main / Tributary complex cannot use this type of queuing unless the complex has a coordinated numbering plan. With ringback / priority off-hook queuing calls that tandem through the network can only enter a queue once.

- Off-Hook / Priority Off-Hook Queuing

  Off-hook / priority off-hook queuing can be assigned to both ends of a 2-way intertandem tie trunk group. However, as with off-hook queuing, blocking increases for calls from subtending locations.

Hard and Soft Processor Swaps

Outgoing trunk queues are stored in a status portion of switch memory. Therefore, if a call is queued to a trunk group when a hard processor swap occurs, the call is never routed outside the System 85 or DEFINITY Generic 2. The queue is cleared.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

The callback sequence associated with ringback queuing is delayed until there are no waiting calls.

AAR (Automatic Alternate Routing)

An incoming tie trunk that infers AAR routing cannot have ringback queuing.

Automatic Callback

The Automatic Callback feature and ringback queuing, at the local switch, cannot be activated at the same time by a terminal.

Activation of the Automatic Callback feature towards a terminal line that has Queuing activated is allowed, but deferred, until either the Queuing process is resolved or the administered Queuing interval times out.

ARS (Automatic Route Selection)

An incoming tie trunk that infers ARS routing cannot have ringback queuing.

Call Coverage

A ***Queuing Callback Call*** is treated as a priority call and does not route to coverage. These callbacks ring at the terminal that originally placed the call. Callback calls do not cover, even if coverage is in effect.

Call Pickup

Call Pickup cannot be used to answer a queuing callback from the local switch, directed at a different member of the Call Pickup group. When this is attempted, the switch returnes busy tone.

A callback from a remote (tandem) switch looks like any other incoming tie trunk call and can be picked up. Also, a callback call between a main and subtending switch appears as a normal tie trunk call and can be picked up.

Call Waiting

For local calls, the Call Waiting feature and the Queueing feature are mutually exclusive on the same station at the same time. That is, a call cannot wait on a station that has a call in queue. The caller receives busy tone. This is true for any situation (including ringback queuing when active on another call) where a call might otherwize be expected to wait. Also, a station that has a call waiting cannot place a call in queue. If a callback attempt is made from a tandem switch to a subtending switch, the call appears as an ordinary incoming tie trunk call to the subtending switch. Consequently, if the calling party has call waiting assigned and is busy on another call, the callback attempt notifies the called party. If the callback attempt is answered, the call can be completed while the third party waits on hold. If the called party ignores the tone, the attempt is treated as "don't answer attempt."

Conference—Three Party

A station that is in off-hook queueing cannot use the Conference feature. A button press on the conference button is ignored.

DCA (Data Communications Access)

The Queuing feature is compatible with the DCA feature. Queuing can provide a waiting list for DCA trunk groups when these data resources are busy. Then, as resources become available, the first user in queue is served.

A DCA trunk group can be directly accessed using a data terminal. For data-terminal access, off-hook queuing is the only type of queuing that can be used. With off-hook queuing the data terminal user receives a RINGING prompt until a circuit becomes available.

A DCA trunk group can also be indirectly accessed using a voice terminal and then transferring the call to an associated data terminal. For voice-terminal access followed by transfer to a data terminal, either off-hook or on-hook queuing can be used. For off-hook queuing the switch returns the music, announcement, or silence. But, for on-hook queuing the calling party receives confirmation tone and then goes on-hook to wait for the queuing callback.

To allow data terminals to directly access and voice terminals to indirectly access the same DCA trunk group, off-hook queuing must be assigned the DCA trunk group.

If the calling party hears reorder tone, the queue might be full. The user should go on-hook and try again.

Hold

A voice terminal user can be placed in a ringback queue while holding another call. Queuing allows a voice terminal user to dial a busy outgoing trunk group and to be placed in a queue and called back when a trunk is available. If queuing is activated and the calling voice terminal has a call on hold, going on-hook returns the held call first (Hold takes precedence over queuing callback). After the calling voice terminal user goes on-hook and the dialed trunk group is idle, the calling voice terminal is rung back.

Host Computer Access

The Queuing feature provides a waiting list when all resources are busy. As resources become available, the first user in queue is served. Queuing can be used with Host Computer Access. Voice terminal users can be provided with on-hook or off-hook queuing. Data terminal users can be provided off-hook queuing. If every Host Computer Access port is busy, users dialing the Host Computer Access access code receive confirmation tone (on-hook queuing), music-on-hold, a recorded announcement, or silence (off-hook queuing). If reorder tone is received, it maybe an indication that the queue is full. It might also mean the switch is in heavy use. The user should go on-hook and try again.

Hunting

When the local switch attempts a queuing callback, the callback call does not hunt.

ISN (Information Systems Network) Interface

System 85 or DEFINITY Generic 2 endpoints can use queuing if all ISN Interface trunk circuits are busy and if the queue is not full. Voice terminal users can be provided with on-hook (if administered) or off-hook queuing. Data terminal users can use off-hook queuing only. Queuing is not provided by ISN. That is, when a call from the System 85 or DEFINITY Generic 2 side has passed the ISN Interface point and is under ISN control, Queuing is no longer available. Also, queuing is not available to ISN station users until after the call has passed the interface point.

ISDN—PRI (Primary Rate Interface)

The Queuing feature works a little differently for certain ISDN calls. Ordinarily, a call will queue (if necessary) on the first-choice trunk group in the appropriate AAR, ARS, or WCR pattern. For ISDN calls, if the COS (Class of Service) indicates ISDN facilities only, the call will queue on the first accessible ***ISDN trunk group*** with the needed bearer capability in the pattern (see discussion under AAR, ARS, and WCR interaction). This may not be in the first-choice trunk group for the routing pattern.

Last Number Dialed

The Queuing feature handles external calls placed with the Last Number Dialed feature in the same way that calls placed with manually dialed digits are handled. That is, when a trunk becomes available for the call, the destination number does not need to be redialed (assuming it was dialed originally).

Main/Satellite

When a Main / Satellite complex serves as an access arrangement for an ETN, the tandem switch and the main switch can both have ringback queuing. However, a single queue (located at the tandem) serves both switches. If the tandem switch attempts to ringback a local terminal user (no tie trunk involved), the switch ignores certain features. These features are:

- Call Forwarding—Follow Me
- Call Forwarding—Busy and Don't Answer
- Call Pickup
- Call Waiting
- Call Coverage
- Hunting.

When a tandem or main routes a ringback call to a subtending location by way of a tie trunk, the preceding features work normally. The reason is that the subtending switch sees the call as an ordinary incoming tie trunk call.

At a main or satellite location, terminals (stations) can have either ringback or off-hook queuing. Terminals at a tributary location can only have off-hook queuing.

Modem Pooling

When a data call is queued, the modem pool conversion resource is held waiting along with the queued call.

Override

Override is denied when the called extension is in an off-hook queue waiting for an idle trunk.

Precedence Calling

Queuing (except attendant queue) is denied on a Precedence Calling call. Precedence calls that cannot preempt the needed facility are directed to the attendant rather than being placed in a facility queue.

Priority Calling

The callback sequence associated with queuing is delayed until there are no waiting calls.

Remote Access

Remote access calls cannot use ringback (on-hook) queuing. The serving CO drops the connection when the Remote Access caller goes on-hook. The System 85 or DEFINITY Generic 2 switch has no way to call back to the off-net number (within the public network).

Restriction—Attendant Control of Voice Terminals

A call cannot be placed in a ringback queue at the local switch if a terminating restriction is in effect for the calling terminal. However, the call can be placed in a ringback queue at a tandem switch. When the callback attempt is made from a tandem switch, the call appears as an incoming tie trunk call. The original calling party will not receive the callback attempt.

Restriction—Voice Terminal Restrictions

A call from a terminal with Termination or Manual Terminating Line restriction in effect cannot be placed in a ringback queue. This is true for both local (same switch) calls and for calls placed between switches (for example, in a Main/Satellite or Tandem Network arrangement).

Ringing—Distinctive Ringing

Three-burst distinctive ringing only applies to voice terminals assigned to a local switch. When the switch routes a ringback call to a subtending location over a tie trunk, 1-burst ringing is provided by the subtending switch.

Route Advance

Route Advance applies only to calls dialed using a trunk group dial access code. When the Queuing feature is administered with Route Advance active (Procedure 330, Word 2, field 9 set to "1") and every trunk group in the Route Advance pattern is busy, the call queues on the originally accessed trunk group. The switch then periodically scans the trunk group for an available trunk. If a trunk is not available in the first trunk group, the switch checks the subsequent trunk groups for an available trunk.

If the first trunk group in the route advance pattern is not assigned queuing, the call does not queue. Rather, the switch returns reorder tone (fast busy tone) to the calling party.

Tenant Services

The Queuing feature is not affected by petitioning. If queuing is administered for an outgoing trunk group and every trunk in the group is busy, then a call (that is otherwise allowed to use the trunk group) can queue on the trunk group. If queuing is not administered for the trunk group and every trunk in the group is busy, the switch will return reorder tone to the calling party.

Transfer

A station that is in an off-hook queue, cannot use the Transfer feature. A button press on the transfer button is ignored.

WCR (World Class Routing)

The Queuing feature works with the WCR feature in the same way it did with the earlier networking features (AAR and ARS). If no accessible trunks are immediately available in the selected routing pattern, and Queuing is available, a WCR call will queue on the best choice trunk group for the call within the routing pattern.

Hardware Requirements

For Traditional Modules:

Off-hook queuing with music requires one auxiliary trunk circuit (SN231) per module, even if the customer does not provide an audio source.

For Universal Modules:

Off-hook queuing with music requires one auxiliary trunk circuit (TN763C) per module, even if the customer does not provide an audio source.

Regardless of the Module Type:

If the customer does provide an audio source, a voice coupler circuit (36A or 89A) and transformer (2012D) are required.

Feature Administration

Assignment of the Queuing feature is on a per system basis for general queuing characteristics and on a per-trunk group basis for specific queue characteristics and trunk-to-trunk queuing.

On System 85 switches, this feature is administered using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or FM (Facilities Management) feature.

For DEFINITY Generic 2, the Queuing feature is administered using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURE — QUEUING			
PROCEDURE	WORD	PURPOSE	SMT
100	1	Assigns a trunk-group dial access code and trunk type to a trunk group. The applicable trunk types include: 67 Audio (for music in off-hook queue)	No
101	1	Administers the characteristics of trunks assigned to a trunk group.	No
150	1	Associates a trunk group with an equipment location for the audio circuit for off-hook queuing with music.	No
275	1	Assigns features to the system class of service including music-on-hold for off-hook queuing.	Yes
330	1	Activates the Queuing feature and administers queuing parameters to the switch, including type of queuing and priority or nonpriority queues for terminal originated calls. Also administers pattern queuing parameters and threshold and FRL raising values for AAR, ARS, and WCR calls.	Yes
330	2	Administers queuing and trunk-group queuing parameters including priority and nonpriority queue lengths, time-in-queue-limits, and trunk-to-trunk queuing characteristics.	Yes

(Continued)

ADMINISTRATION PROCEDURES — QUEUING *(Continued)*			
	WORD	**PURP0SE**	**SMT**
350	1	Assigns the first digit of dial access codes, including codes used with the Queuing feature.	No
350	2	Assign the feature dial access codes. The applicable encode is: 9 Cancel Ringback Queuing.	No

The following are the applicable FM procedures used with the AP 16.

FM SCREENS — QUEUING	
PATH NAME	**PURPOSE**
facilities-mgmt system-parameters queuing	Turns on and off queuing for a switch. This screens also used to display the threshold FRL, raised FRL, the queuing type for local terminals, and the preference depth (beginning with R2 V3).
facilities-mgmt trunk-groups queuing	Turns queuing on and off for a trunk group and changes queuing attributes associated with a trunk group including time-in-queue limits and queue length (priority and nonpriority). A printed report can also be generated.

Notes:

Radio Paging Access

Description

This feature allows users to page a person over a radio receiver. The paged party must be carrying a radio receiver that is set to the radio paging system. The paged party can answer the page by using a voice terminal and dialing an answer-back channel. Remote Access trunk users can also use this feature.

Radio Paging Access is useful for persons who do not normally remain at one location or who cannot remain out of reach for even short periods of time. Possible users of this feature include medical, managerial, or emergency personnel, or anyone requiring a personal paging service.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Page Multiple Parties

With an alerting tone from a voice terminal:

1. Go off-hook or press an idle appearance button. [Dial tone]
2. Dial the radio paging access code. [Ringback tone is heard, and the paging equipment automatically becomes available. The paging tone is generated over the radio receiver when ringback tone ends.]
3. If answer-back is desired, stay off-hook.
4. If answer-back is not desired, go on-hook. [Paging equipment is released.]

With an alerting tone from an attendant without a DTGS button:

1. Press an idle loop button. [ATND lamp lights.]
2. Press **[START]** (If there is a source party, the SPLIT lamp lights, and the source party is split from the connection.) [Dial tone]
3. Dial the radio paging access code. [Ringback tone is heard, the ANS lamp lights, and the paging equipment automatically becomes available. The paging tone is generated over the radio receiver when ringback tone ends.]
4. If answer-back is desired, stay off-hook.

5. If answer-back is not desired, press **[RELEASE]** . [Paging equipment is released.]

With an alerting tone from an attendant with a DTGS button:

1. Press an idle loop button. [ATND lamp lights.]
2. Press the Direct Trunk Group Selection (DTGS) button. (If there is a source party, the SPLIT lamp lights, and the source party is split from the connection.) [Ringback tone is heard, and the paging equipment automatically becomes available. The paging tone is generated over the radio receiver when ringback tone ends.]
3. If answer-back is desired, stay off-hook.
4. If answer-back is not desired, press **[RELEASE]** . [Paging equipment is released.]

With voice announcements from a voice terminal:

1. Go off-hook or press an idle appearance button. [Dial tone]
2. Dial the radio paging access code. [Ringback tone is heard. Paging equipment automatically becomes available when ringback tone ends.]
3. When ringback tone ends, make the announcement. (Announcement is heard over the radio receiver.)
4. If answer-back is desired, stay off-hook.
5. If answer-back is not desired, go on-hook. [Paging equipment is released.]

With voice announcements from an attendant without a DTGS button:

1. Press an idle loop button. [ATND lamp lights.]
2. Press **[START]** . (If there is a source party, the SPLIT lamp lights, and the source party is split from the connection.) [Dial tone]
3. Dial the radio paging access code. [Ringback tone is heard, and the ANS lamp lights. Paging equipment automatically becomes available when ringback tone ends.]
4. When ringback tone ends, make the announcement. (Announcement is heard over the radio receiver.)
5. If answer-back is desired, stay off-hook.
6. If answer-back is not desired, press **[RELEASE]** . [Paging equipment is released.]

With voice announcements from an attendant with a DTGS button:

1. Press an idle loop button. [ATND lamp lights.]
2. Press the DTGS button. (If there is a source party, the SPLIT lamp lights, and the source party is split from the connection.) [Ringback tone is returned. Paging equipment automatically becomes available when ringback tone ends.]
3. When ringback tone ends, make the announcement. (Announcement is heard over the radio receiver.)

4. If answer-back is desired, stay off-hook.
5. If answer-back is not desired, press **[RELEASE]** . [Paging equipment is released.]

To Page an Individual Party

With an alerting tone from a voice terminal:

1. Go off-hook or press an idle appearance button. [Dial tone]
2. Dial the radio paging access code. [Second dial tone is heard]
3. Dial the individual page number. [Ringback tone is heard, and the paging equipment automatically becomes available. The paging tone is generated over the radio receiver when ringback tone ends.]
4. If answer-back is wanted, wait off-hook.
5. If answer-back is not wanted, go on-hook. [Paging equipment is released.]

With an alerting tone from an attendant without a DTGS button:

1. Press an idle loop button. [ATND lamp lights.]
2. Press **[START]** . (If there is a source party, the SPLIT lamp lights, and the source party is split from the connection.) [Dial tone]
3. Dial the radio paging access code. [Second dial tone, ANS lamp lights.]
4. Dial the individual page number. [Ringback tone is head, and the paging equipment automatically becomes available. The paging tone is generated over the radio receiver when ringback tone ends.]
5. If answer-back is wanted, wait off-hook.
6. If answer-back is not wanted, press **[RELEASE]** . [Paging equipment is released.]

With an alerting tone from an attendant with a DTGS button:

1. Press an idle loop button. [ATND lamp lights.]
2. Press the DTGS button. (If there is a source party, SPLIT lamp lights, and the source party is split from the connection.) [Dial tone is returned, ANS lamp lights.]
3. Dial the individual page number. [Ringback tone is heard, and the paging equipment automatically becomes available. The paging tone is generated over the radio receiver when ringback tone ends.]
4. If answer-back is wanted, wait off-hook.
5. If answer-back is not wanted, press **[RELEASE]** . [Paging equipment is released.]

With a voice announcement from a voice terminal:

1. Go off-hook or press an idle appearance button. [Dial tone]
2. Dial the radio paging access code. [Second dial tone]

3. Dial the individual page number. [Ringback tone is heard. Paging equipment automatically becomes available when ringback tone ends.]
4. When ringback tone ends, make the announcement. (Announcement is heard over the radio receiver.)
5. If answer-back is wanted, wait off-hook.
6. If answer-back is not wanted, go on-hook. [Paging equipment is released.]

With a voice announcement from an attendant without a DTGS button:

1. Press an idle loop button. [ATND lamp lights.]
2. Press **[START]** . (If there is a source party, the SPLIT lamp lights, and the source party is split from the connection.) [Dial tone]
3. Dial the radio paging access code. [Second dial tone is heard, and ANS lamp lights.]
4. Dial the individual page number. [Ringback tone is heard. Paging equipment automatically becomes available when ringback tone ends.]
5. When ringback tone ends, make the announcement. (Announcement is heard over the radio receiver.)
6. If answer-back is wanted, wait off-hook.
7. If answer-back is not wanted, press **[RELEASE]** . [Paging equipment is released.]

With a voice announcement from an attendant with a DTGS button:

1. Press an idle loop button. [ATND lamp lights.]
2. Press the DTGS button. (If there is a source party, the SPLIT lamp lights, and the source party is split from the connection.) [Second dial tone is heard, and ANS lamp lights.]
3. Dial the individual page number. [Ringback tone is heard. Paging equipment automatically becomes available when ringback tone ends.]
4. When ringback tone ends, make the announcement (Announcement is heard over the radio receiver.)
5. If answer-back is wanted, wait off-hook.
6. If answer-back is not wanted, press **[RELEASE]** . [Paging equipment is released.]

To Answer a Page:

1. Go off-hook. [Dial tone]
2. Dial the answer-back access code. [Ringback tone]
3. Paged party is connected to the paging party. [Paging equipment is released.]

Considerations

Busy Tone

Busy tone is returned if the paged party calls the paging party and the paging party has gone on-hook and then off-hook. Busy tone is also returned if both paging trunks are busy or if the primary trunk is in the process of paging.

Reorder Tone

Reorder tone is returned if all paging circuits are busy.

Outside (Off-Premises) Calls

Callers on outside trunks (except Remote Access) cannot page over the radio paging system. However, an attendant or a voice terminal user can page the party for them.

Touch-Tone Dialing Required

Radio Paging Access calls must be made from terminals with a touch-tone telephone dial to access the paging equipment.

Single Circuit Access Limit

When one paging circuit is busy, either on a page or waiting for answer-back, the other paging circuit cannot be accessed. The switch returns busy tone in this situation. The second paging circuit can be accessed as soon as the first paging circuit connects to an answer-back channel.

Interactions With Other Features

None.

Restricting Feature Use

The following features can restrict access to the Radio Paging Access feature.

- Attendant Control of Trunk Group Access
- Attendant Control of Terminal Access
- Voice Terminal Restriction.

Hardware Requirements

The Radio Paging Access feature requires the following additional or special hardware.

For Traditional Modules:

- An SN230 circuit pack for two complete paging channels
- An SN253 auxiliary tone plant to generate tones for the customer-provided equipment

For Universal Modules:

- A TN747B circuit pack for four complete paging channels
- A TN768 tone plant to generate tones for the customer-provided equipment.

Regardless of the Module Type:

- A J58824CD interface circuit
- A J59204CA G1-type touch-tone receiver
- A 36A voice coupler
- A 2012D power transformer to supply power to the voice coupler
- Customer-provided radio transmitting and receiving equipment.

Feature Administration

Assignment of the Radio Paging Access feature is on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — RADIO PAGING ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to an extension number.	Yes
010	3	Assigns a Miscellaneous Trunk Restrictions group to an extension class of service.	Yes
100	1	Assigns the trunk type and dial access code for a Radio Paging Access trunk group. The applicable trunk-type encodes include: 17 1-way outgoing DOD.	No
101	1	Administers the characteristics for the trunk groups administered in Procedure 100, Word 1.	No
102	1	Assigns the radio paging and answer-back trunk groups (using dial access codes) to Miscellaneous Trunk Restriction groups.	Yes
150	1	Assigns the SN230 or TN747B equipment location for each radio paging and answer-back trunk to the trunk groups assigned in Procedure 100, Word 1.	No
202	1	Administers the desired trunk group selection buttons on the attendant console(s).	No
350	1	Assigns the first digit of the trunk group dial access code (if required).	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — RADIO PAGING ACCESS	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns a Miscellaneous Trunk Restrictions group to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Notes:

Recall Signaling

Description

This feature allows a single-appearance voice terminal user who is busy on a 2-party call to place the second party on hold and obtain recall dial tone. The voice terminal user can then call another party or activate a feature.

NOTE: This feature does not apply to multiappearance voice terminals. Multiappearance voice terminal users can use the Conference, Hold, or Transfer features to achieve the same results. See the Glossary for a description of the RECALL button functions on different types of voice terminals.

Feature History and Development

This feature was first available for System 85 in Release 1. It has not changed since its introduction.

User Operations

The following are the user operating procedures for this feature.

To Place the Second Party on Hold and Obtain Recall Dial Tone

For terminals without a RECALL button:

Momentarily press the switchhook. [Recall dial tone]

(Connected party placed on soft hold.)

For terminals with a RECALL button:

Press **[RECALL]** . [Recall dial tone]

(Connected party placed on soft hold.)

Considerations

This feature cannot be used when the second party to the connection is an attendant console position.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 feature affects or is affected by the operation of this feature.

Remote Access

This feature can be used by a local station when a Remote Access user is the second party on the connection. However, the Remote Access user cannot use the Recall Signaling feature.

Hardware Requirements

None.

Feature Administration

The Recall Signaling feature is always provided on a per-system basis. Assignment is not required.

Recorded Telephone Dictation Access

Description

This feature permits voice terminal users within the switch and at remote locations to access customer-provided dictation equipment. The following functions are provided:

- Start function (voice or dial activated)
- Stop function (voice or dial activated)
- Playback (dial activated)
- Corrections (dial activated).

Easy access to dictation equipment enhances and simplifies the customer's dictation services. Thus, the feature reduces secretarial requirements and facilitates word processing.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Access a Recorded Telephone Dictation Trunk:

1. Go off-hook. [Dial tone]
2. Dial the recorded telephone dictation trunk-group access code. [Second dial tone or 1400-hertz ready tone]

To Start Operation of the Dictation Equipment:

1. Speak into the voice terminal handset (for voice-activated systems).
2. Dial the digit **[1]** (if the dictation equipment is dial controlled).

To Stop the Recording Process:

1. Stop speaking (for a voice-activated system). The absence of speech stops the recording process.
2. Dial the digit **[1]** again (for a dial-activated system).

Options are available to allow the user to control other aspects of the dictation process. These options include:

To Initiate Playback:

Dial the digit **[3]** .

To Cancel Playback:

Dial the digit **[1]** .

To indicate a Correction:

Dial the digit **[2]** .

To Access the Dictation Attendant:

Dial the digit **[0]** .

Whenever the digits 5 through 9 or the characters * and # are dialed, the user is routed to the dictation attendant to indicate a dialing error.

To End the Dictation Session:

Dial the digit **[4]** .

Considerations

Denied Access

Attendants and tie trunk users cannot access a telephone dictation trunk.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Control of Trunk Group Access

If Attendant Control of Trunk Group Access is active on a recorded telephone dictation trunk group when the switch is in the preselected call routing mode of operation, any attempt to access a recoded telephone dictation trunk in the restricted trunk group results in intercept tone.

Attendant Call Waiting

The Attendant Call Waiting feature is denied toward an extension with the Recorded Telephone Dictation feature activated.

AAR (Automatic Alternate Routing)

Autpomatic Alternate Routing cannot be used in conjunction with the Recorded Telephone Dictation Access feature.

ARS (Automatic Route Selection)

Automatic Route Selection cannot be used in conjunction with the Recorded Telephone Dictation Access feature.

Call Waiting

A call is not allowed to wait on a terminal that has activated Recorded Telephone Dictation Access.

Conference—Three Party

The Conference—Three Party feature cannot be used in conjunction with the Recorded Telephone Dictation feature Any attempt to access a recorded telephone dictation trunk from a 2-party connection is denied.

Off-Premises Terminal

Recorded telephone dictation access trunks are not available to trunk port off-premises

Priority Calling

Priority Calling is denied when Recorded Telephone Dictation Access is active at the called terminal.

Tenant Services

The trunk group for the Recorded Telephone Dictation Access feature is not partitioned. By default, the provided trunks are equally accessible to voice terminal users in any extension partition.

Voice terminal access to the Recorded Telephone Dictation Access feature can be limited in the extension class of service. This is done by assigning a Miscellaneous Trunk Restrictions group containing the dictation trunk group to a extension class of service in Procedure 010, Word 3.

WCR (World Class Routing)

World Class Routing cannot be used in conjunction with the Recorded Telephone Dictation Access feature.

Restricting Feature Use

The following features can restrict the Recorded Telephone Dictation Access feature.

- Attendant Control of Trunk Group Access
- Attendant Control of Voice Terminals
- Miscellaneous Trunk Restrictions.

Hardware Requirements

The Recorded Telephone Dictation Access feature requires the following additional or special hardware.

For Traditional Modules:

- SN231 Auxiliary Trunk Circuit Pack
- Each dictation trunk uses one circuit of an SN231 circuit pack (four circuits per SN231).
- SN253 Auxiliary Tone Plant
- An SN253 auxiliary tone plant to provide tones to the voice terminals.

For Universal Modules:

- TN763C Auxiliary Trunk Circuit Pack
- Each dictation trunk requires one circuit of a TN763C circuit pack (four circuits per TN763C).
- TN768 Tone/Clock
- A TN768 tone/clock to provide tones to the voice terminals.

Regardless of the Module Type (PEC-65241):

- A frequency generator unit (J58889N, L1).
- A frequency interrupter unit (J58889N, L2).
- A recorded telephone dictation unit (J58827E-1, L1 and L7).
- A 36A voice coupler.
- A 2012D power transformer to supply power to the voice coupler.

If the switch uses touch-tone dialing and external touch-tone dialing conversion, the following hardware is also required.

- A recorded telephone dictation touch-tone dialing translation unit (J58827E, L1, L2, and L7)
- Touch-tone calling receivers (J99289C, L1, C, SA, or J59204CA, L1).

NOTE: The specific interface equipment required for the Recorded Telephone Dictation Access feature depends on the capabilities of the dictation recording unit and whether dictation is being originated from a telephone that uses rotary dialing.

If the dictation recording unit ***can*** interpret touch-tone Commands, then the only equipment required is an SN231 or TN763C auxiliary trunk at the switch and a 36A voice coupler.

If the dictation recording unit ***cannot*** interpret touch-tone commands, then an assortment of conversion equipment and additional interface equipment is required. This equipment is necessary because the dictation recording unit only recognizes dial-pulse signals. However, if dictation is given from rotary phones only, the touch-tone equipment is not needed. [Refer to ***System Description*** (555-105-201) for more details.]

Feature Administration

Assignment of the Recorded Telephone Dictation Access feature is on a per-trunk group basis. Terminal access to a Recorded Telephone Dictation trunk is assigned on a per-extension class of service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The Customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES RECORDED TELEPONE DICTATION ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to an extension number.	Yes
010	3	Assigns a Miscellaneous Trunk Restrictions group to an extension class of service	Yes
100	1	Administers the trunk type and dial access code for Recoded Telephone Dictation. The applicable trunk-type encode is: 51 Telephone Dictation Interface.	No
100	2	For DEFINITY Generic 2, administer the Bearer Capability Class of Service for trunk groups. For Recorded Telephone Dictation BCCOS 0 as predefined is applicable.	No
102	1	Assigns the trunk group (using dial access code) to Miscellaneous Trunk Restriction groups.	Yes
150	1	Assigns the SN231 or TN763C equipment location to the trunk-group number administered m Procedure 100, word 1.	No
350	1	Assigns the first digit of the trunk-group dial access codes (if required).	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — RECORDED TELEPHONE DICTATION ACCESS	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns a Miscellaneous Trunk Restrictions group to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Remote Access

Description

The Remote Access feature allows external touch-tone dialing users to access the System 85 or DEFINITY Generic 2 switch through the public network, and then use many (but not all) features and services of the switch as though they were local station users.

Sources of Access

Remote Access service can be provided using local CO (Central Office) trunks, FX (Foreign Exchange) trunks, or WATS (800 Service) facilities. For Release 2, Version 3 and 4 switches and DEFINITY Generic 2 switches, direct access facilities such as MEGACOM WATS service can also be used. In System 85, Version 3, DS1 facilities are required for Remote Access MEGACOM WATS service.

Feature History and Development

The Remote Access feature was first available for System 85 with Release 1.

For Release 2, Version 4 of System 85, the capacity for Remote Access trunks was increased from 45 to 6000. This increased capacity was provided to meet the needs of partitioned switches (Tenant Services feature). However, the full capacity of 6000 is also available to switches that are not partitioned.

Also for System 85, Release 2, Version 4 and subsequently for DEFINITY Generic 2, the Remote Access feature was enhanced to allow the use of special access facilities such as MEGACOM WATS service.

Adjunct Enhanced Security software was added late in the System 85, Release 2, Version 4 time frame. This enhancement supports a future adjunct based capability and applies to System 85, Release 2, Version 3 (Issue 1.2 and later), Release 2, Version 4 (Issue 1.1 and later), and DEFINITY Generic 2 switches.

Basic Access Options

The Remote Access feature can be administered to provide three different modes of access or use.

Direct Dialed Access

With direct dialed access, the Remote Access user dials the assigned remote access number, followed by any required access control procedures. Direct dialed access requires the use of a touch-tone dialing terminal. If direct dialed access is provided, an incoming remote access call receives System 85 or DEFINITY Generic 2 dial tone, and the user can proceed to dial the rest of the call without attendant assistance.

Shared Access

If access is "shared" with LDN service, a Remote Access call is routed to the attendant under normal (business hours) conditions, and the attendant extends the call like any other LDN call. When Unattended Console Service is active, "shared" non-DID LDN service becomes inactive, and Remote Access calls are handled as direct dialed access calls.

In effect, with "shared" non-DID LDN service, the Remote Access feature is turned off while the attendant is on duty, and remote access trunks function as non-DID LDN trunks.

Time-Out To Attendant

A timeout to attendant option is also available for Remote Access calls. With this option active, the caller can maintain control by dialing (dialing the barrier or authorization code and then dialing an extension number, trunk-group access code, or feature access code) within 10 seconds after receiving switch dial tone. Otherwise, the call is automatically routed to an attendant or to the Unattended Console Service terminal.

The time-out to attendant option allows the Remote Access user with a rotary dial terminal to receive attendant assistance. Rotary dial terminals cannot be used with remote access in the direct dialed access mode.

Security Measures Available

Shared LDN Service

The non-DID LDN "shared" service option provides a degree of security for Remote Access during normal operations (attendant on duty) by allowing the attendant to screen remote access calls before extending them.

Additional security measures available include the use of a Barrier Code or the Authorization Code feature, and will include the use of a call screening adjunct device for enhanced security.

Barrier Codes

Barrier Codes are 4-digit codes assigned system wide (one code for all users). A Barrier Code can be required for direct dialed remote access calls. A remote access caller who does not know the current barrier code would be denied service (intercept tone) or routed to attendant intercept.

When "Shared" non-DID LDN service is in effect, the Barrier Code is not required before the call is routed to the attendant. These codes are assigned by the attendant and should be changed frequently to maintain their security value.

Authorization Codes

Authorization Codes can be from 4 to 7 digits and are assigned to individual users. The Authorization Code feature can be used instead of a Barrier Code, or it can be used to supplement the protection provided by a Barrier Code.

With System 85 R2 V3 and later switches, and with DEFINITY Generic 2 switches, there can be up to 90,000 separate authorization codes assigned to the switch. Earlier System 85 switches are limited to 9,000 authorization codes.

Network Access Flag

The network access flag is administered to each authorization code using Procedure 282, Word 1, Field 3. This flag adds another dimension to the security provided by using the Authorization Code feature The network access flag determines whether or not the associated authorization code can be used with the Remote Access feature. If the network access flag is set to "0," that authorization code cannot be used with Remote Access. That is, an authorization code with network access flag set to "0" cannot be used to gain access to the Remote Access feature, or if access is gained either by using a barrier code or through attendant assistance, that authorization code cannot be used to raise the FRL of a call placed using the Remote Access feature and routed by one of the network routing features (AAR, ARS, or WCR).

Adjunct Enhanced Security

The adjunct enhanced security (AES) software for the Remote Access feature became available late in the System 85, Release 2, Version 4 time frame. Adjunct enhance security is designed as an adjunct-operated security measure that is applicable to R2 V3 (1.2 and later) and R2 V4 (1.1 and later) System 85s that use the Remote Access feature, and to DEFINITY Generic 2 switches. The adjunct hardware needed for enhanced security operations is under development and will be available at some time in the future.

With Adjunct Enhanced Security, calls arriving on remote access trunk groups are intercepted by either ACD or Call Vectoring and routed to the adjunct. The caller is then prompted by the security adjunct to enter an authorization code. The authorization code is verified (or not verified) by the adjunct. If the authorization code is verified, the caller is then prompted for a password phrase. This password phrase is compared to a recorded sample that is used to verify the identity of the caller.

On a System 85, R2 V3 switch, if the call fails to pass the Adjunct Enhanced Security tests, the call will be dropped by the adjunct. On System 85, R2 V4 and Generic 2 switches, the AES adjunct will pass the call back to the switch with a report of its results in the form of an access code, indicating one of the following conclusions:

- Valid Authorization Code entered and password phrase verified
- Valid Authorization Code entered but password phrase not verified

Interactions Between Security Measures

Barrier Code and Authorization Code

When a Barrier Code is used to control use of the Remote Access feature, the Authorization Code feature can be used separately for control of other switch features and facilities. For example, the Authorization Code feature is used with the FRL (Facilities Restriction Level) feature for access control to selected trunk facilities.

While the Authorization Code feature functions in the same way with FRL when a barrier code is not used, using both a Barrier Code and the Authorization Code feature in combination provides the flexibility of immediate attendant action (changing the Barrier Code) when a breach of security is detected. When both a Barrier Code and Authorization Code are used, a low FRL is assigned to remote access trunks. This becomes the default FRL for all Remote Access calls using these trunks. Further access to facilities you want to control, such as MEGACOM WATS service, WATS, FX or private network trunks, can be controlled by assigning higher FRLs to these facilities and requiring Remote Access users to enter an Authorization Code that will raise their FRL before they are granted access to these controlled facilities.

When the Authorization Code is used instead of a Barrier Code for control of Remote Access, the same authorization code is "remembered" by the switch and used if needed later in the call.

Adjunct Enhanced Security and Authorization Code

Adjunct Enhanced Security (AES) adds an additional layer of security screening to Remote Access calls. With adjunct enhanced security, Authorization Code must be used The Barrier Code cannot be used with the AES option. That is, the Authorization Code must always be dialed, with AES adding another test before remote access callers are granted access to the switch. The Authorization Code is used by the adjunct as an individual identity code, to identify the caller and select the appropriate password phrase. After verifying (or not verifying) the identity of the caller, the adjunct passes control back to the switch (along with the results of the AES checks in the form of access codes) for appropriate handling (connect the call, Intercept Treatment, etc.).

If the adjunct becomes inoperable because of a major alarm condition, the call is not passed to AES. In this case, standard Authorization Code access screening is used by the switch before access is granted to the Remote Access feature.

Bearer Capability

For DEFINITY Generic 2 switches, the Bearer Capability feature can have a significant impact on the Remote Access feature. Bearer Capability is used to identify the type of call being processed and the call support facilities required for a specific call. For general purpose type Remote Access trunk groups, a BCCOS (Bearer Capability Class of Service) of "3" (unknown digital) or "4" (unknown analog) would normally be used. However, with the Bearer Capability feature, it is possible to set up "special purpose" Remote Access trunk groups. That is, a digital Remote Access trunk group could be assigned a BCCOS of 1. This would result in a Remote Access trunk group that could be used only for mode 2 data calls. Another Remote Access trunk group could be assigned a BCCOS of 0. This would be a Remote Access trunk group for voice calls only. A BCCOS of 5 would result in a Remote Access trunk group for data calls that would always receive Modem Pooling support.

User Operations

The following are the user operating procedures for this feature.

To Use Remote Access From an External Location

Without "Shared" Non-DID LDN Service:

1. Dial the 7- or 10-digit Remote Access number. [Ringback tone followed by second dial tone]
2. If an Authorization or Barrier Code is required, dial the appropriate code followed by the desired extension number or access code.

   NOTE: If the switch returns recall dial tone, an Authorization Code is needed to complete the call.

3. If an Authorization or Barrier Code is not required, dial the desired extension number or access code.

With "Shared" (Non-DID) LDN Service:

1. Dial the 7- or 10-digit Remote Access number. [Ringback tone] (Call routes directly to attendant.)
2. Ask the attendant to extend the call as desired.

With Adjunct Enhanced Security:

1. Dial the 7- or l0-digit Remote Access number. [Ringback tone] (Call is routed to the adjunct.)

   The security adjunct prompts for an Authorization Code. [Dial Tone.]

2. Dial the assigned Authorization Code. [The adjunct prompts for the password phrase.]
3. Give the assigned password phrase. [Dial Tone]
4. Dial the desired extension number or access code.

To Change the Barrier Code From the Attendant Console:

1. Press an idle loop button.
2. Press **[START]** . [Dial tone]
3. Dial the change barrier code access code. [Second dial tone]
4. Enter the new 4-digit barrier code. [Confirmation tone]
5. Press **[RELEASE]** . (New barrier code is in effect.)

Considerations

Security Considerations

The Remote Access feature is a powerful tool that could easily be abused. For that mason, it is important that it be protected from unauthorized use. Specific measures can be taken to limit feature access to authorized users. These include:

- Shared Non-DID LDN Service
- Barrier Code
- Authorization Codes
- Adjunct Enhanced security.

Each of these security measure options was discussed earlier in the Description section.

The customer is responsible for the security of remote access to the AT&T (or other common carrier) network and is responsible for all charges incurred by such access, whether authorized or unauthorized. Please consult with your local account team for more details.

Echo-Suppression Requirements

On certain long haul network trunking facilities (such as, MEGACOM WATS or WATS trunks), echo-suppressors are sometimes used to prevent propagation delays from interfering with call setup signals. Unfortunately, the use of echo-suppressors can prevent the completion of a Remote Access call over these trunks. It may be necessary to employ special echo-suppression signaling techniques on some long distance trunks to be able to use them for the Remote Access feature.

Automatic Callback and Ringback Queuing

Automatic Callback and Ringback Queuing require a call back to the caller. Both of these capabilities are denied to the Remote Access caller because the switch has no way of identifying the calling (outside) number. Off-hook queuing, if administered, can be used by Remote Access callers.

Switchhook Flashing

Any feature that requires signaling the switch with a switchhook flash (for example, Conference—Three Party, Hold, and Transfer) or equivalent (button) cannot be used via Remote Access. The user is not able to send a switchhook flash signal over the trunk connection.

Loudspeaker Paging Access

The Remote Access and the Loudspeaker Paging Access features cannot be assigned to the same trunk group.

Touch-Tone Dialing Requirement

A Remote Access user must use a voice terminal equipped for touch-tone dialing if a barrier code or authorization code is required by the switch. A rotary dial user cannot enter a barrier code or authorization code. A call placed from a rotary dialing terminal, in a switch requiring a barrier or authorization code, can only be completed with attendant assistance.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

When the Abbreviated Dialing feature is administered to allow "Global" access to the System List, this includes Remote Access users on Release 2, Version 3 and later switches.

AUDIX

The AUDIX feature is available to Remote Access callers on the same basis as for any outside call. That is, Remote Access callers can use AUDIX in the same way as outside callers. Local call functionality such as Leave Word Calling and return call is not available on Remote Access AUDIX calls and AUDIX service (voice mail) cannot be provided for the "simulated call appearance" used by Remote Access trunks.

Authorization Code

The Authorization Code feature can be used for access control (instead of a barrier code) by the Remote Access feature. This provides an individual identity check rather than the group identity check provided by a barrier code. When adjunct enhanced security is used, the authorization code is required; a barrier code cannot be used. However, if the network access flag for a specific authorization code is set to "0," that authorization code cannot be used with the Remote Access feature for any purpose, regardless of how access to the Remote Access feature is controlled.

Automatic Callback

A Remote Access user cannot use the Automatic Callback feature. The switch has no way of placing a callback call to an off-net number.

Bearer Capability

The Bearer Capability feature applies to Remote Access trunk groups on DEFINITY Generic 2 switches in the same way that it applies to other types of trunk groups. Care must be taken in administering BCCOS to Remote Access trunk groups as this has a significant effect on call processing for remote access calls. The Bearer Capability feature does not apply to System 85 switches.

CDR (Call Detail Recording)

If a Remote Access call terminates in the switch or tandems through the switch, the call record stores the attendant ID number (DAC), if assigned, for the remote access group. The user can dial a CDR account code after the remote access code.

Remote access to the network may require a dialed barrier code or an authorization code. However, the call record only stems a dialed authorization code.

If a station receives a call on an incoming remote access trunk and then transfers the call, the station user who received the transferred call will be recorded as the dialed number.

Conference—Attendant Five Party

A Remote Access user can be a member of an attendant established conference only as the conference requester (first party added to the conference).

Conference—Attendant Six Party

A Remote Access user cannot be a member of an attendant established 6-party conference except as the requesting conferee (first party connected to the conference circuit). This is because the attendant has no way of dialing the simulated line appearance of the remote access trunk. Even if this were possible, once seized by a Remote Access user, the appearance would be busy to any incoming call.

Conference—Three Party

A Remote Access user can be a member of a 3-party conference but cannot establish the conference. To be a member of a 3-party conference, the Remote Access user first calls a local station user, and then that local station user must establish the conference by adding the third party. Otherwise, there is no way for a Remote Access user to send the necessary signals to the switch.

Dial Access to Attendant

A Remote Access user can call the attendant group by dialing the attendant access code.

Beginning with R2 V3 System 85, a Remote Access user can call the attendant group by dialing a DID LDN.

DS1 Interface

The DS1 Interface feature is compatible with the Remote Access feature. That is, DS1 trunks, when available, can be used for Remote Access. In System 85, Release 2, Version 3, DS1 trunks must be used if Remote Access is to be via MEGACOM WATS service.

Display—Voice Terminal

The voice terminal display for a Remote Access call is **[OUTSIDE CALL]** . This display appears even if a different name is assigned to the trunk group in Procedure 012.

Foreign Exchange Access

The FX Access feature is fully compatible with the Remote Access feature. That is, based on restrictions applied, incoming FX Access calls can use the Remote Access feature and Remote Access calls to the switch can use outgoing FX Access trunks.

Hold

An incoming remote access call can be placed on hold; however, the Remote Access user cannot use the Hold feature.

ISDN—PRI (Primary Rate Interface)

The Remote Access feature works normally with ISDN through the interworking function.

ISDN—PRI trunks that are assigned trunk type 120 (Dynamic Trunk Type) cannot be used for the Remote Access feature.

Last Number Dialed

The Last Number Dialed feature does not store and redial the digits dialed during a Remote Access call.

Leave Word Calling

The Leave Word 3 Calling feature is not available to Remote Access users. The switch has no way of knowing what actual public network number is being used and return call functionality is not available.

Look-Ahead Interflow

At a sending switch, a Remote Access user is allowed to dial a VDN after the sending switch returns second dial tone. When this is done, the Remote Access call will complete to the VDN with a vector assigned that may contain command(s) for Look-Ahead Interflow. In turn, these incoming Remote Access calls will interflow normally (that is, according to the commands in the sending and receiving vectors).

Multiple Listed Directory Numbers

When the switch is in the day mode and Remote Access has shared service with non-DID LDNs, users dialing the Remote Access number are routed to the attendant queue. The calls are then extended by an attendant rather than using the Remote Access feature.

When the switch is in the unattended console mode or the Remote Access is not shared with non-DID LDN service, a Remote Access user can dial an assigned LDN to reach the attendant group or the night terminal.

Precedence Calling

Remote Access can be used in conjunction with the Precedence Calling feature if precedence calling is administered to class of service 31.

Queuing

Remote access calls cannot use ringback (on-hook) queuing. If ringback queuing is attempted, the serving CO will drop the connection when the Remote Access caller goes on-hook. Once this connection is lost, the System 85 or DEFINITY Generic 2 switch has no way to call back to this off-net number (within the public network).

Recall Signaling

This feature can be used by a local station when a Remote Access user is the second party on the connection. However, the Remote Access user cannot use the Recall Signaling feature.

Restriction—Voice Terminal Restrictions

The Remote Access user cannot complete a call to a switch terminal if any of the following features are active on the called extension

- Manual Terminating Line
- Termination Restriction

Transfer

A remote access call can be transferred by a regular System 85 or DEFINITY Generic 2 station user. However, the Remote Access caller cannot use the Transfer feature.

Trunk Verification—Voice Terminal

Remote Access trunks (trunk type 50) can be "maintenance busied" or "unbusied" via the Trunk Verification—Voice Terminal feature. However, a Remote Access user cannot use the Trunk Verification—Voice Terminal feature.

WATS Access

The Remote Access feature can be used to provide access to the WATS feature, and Remote Access services can be provided to incoming calls on 800 Service trunks.

WCR (World Class Routing)

The WCR feature is fully compatible with the Remote Access feature for outgoing calls. That is, subject to class of service limitations, a Remote Access caller can place outgoing calls over WCR networks in the same way as a similarly restricted local caller.

Restricting Feature Use

Security Measure Restrictions

Barrier codes or authorization codes are used primarily for switch security. These provide a form of restricting use of the Remote Access feature and must be dialed immediately after obtaining switch dial tone. An additional control measure is available with the use of authorization codes. Each authorization code has a network access flag assigned. The network access flag is used to specify whether or not that specific authorization code can be used with the Remote Access feature.

Restriction Features

The Restriction—Voice Terminal Restrictions feature can be used to apply class of service restrictions to the "simulated line appearance" of the Remote Access trunk. This identifies the calling privileges and restrictions applicable to Remote Access users.

The Restiction—Miscellaneous Trunk Restriction feature can be used to restrict Remote Access users. This is used in conjunction with the Restriction—Voice Terminal Restrictions feature to control access through dial access codes to selected trunk groups.

Hardware Requirements

The Remote Access feature requires the following additional or special hardware.

For Traditional Modules:

- SN230 Central Office Trunk Circuit Pack four circuits each

  Provides analog trunk interface for the Remote Access feature.

- ANN11 (D or E) DS1 Interface Trunk Circuit 24 channels each.

  Provides optional digital trunk interface for the Remote Access feature.

For Universal Modules:

- TN747B Central Office Trunk Circuit Pack, eight circuits each

  Provides analog trunk interface for the Remote Access feature.

- TN767 DS1 Interface Trunk Circuit, 24 channels each.

  Provides optional digital trunk interface for the Remote Access feature.

Regardless of the Module Type:

- LORAIN variable voice switched gain amplifier (optional)

  This amplifier compensates for undesirable transmission conditions (such as too much gain or not enough gain) that may interfere with Remote Access calls.

- Polarity guard

  The polarity guard is necessary for any originating remote telephone on which the polarity is reversed when the call is answered (for example, telephones served by step-by-step Central Offices and dial-tone-first coin telephones). The polarity guard allows the calling party to use touch-tone dialing which is necessary for the Remote Access feature.

Feature Administration

Assignment of the Remote Access feature is on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal), the TCM (Terminal Change Management).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — REMOTE ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
010	1 to 4	Assigns the characteristics of class of service 31 for Remote Access.	Yes
031	2	Assigns termination of Remote Access calls to a Vector Directory number when AES is used (R2V4 and DEFINlTY Generic2 only).	No
100	1	Assigns the trunk-group dial access code and trunk type for a Remote Access trunk group. The applicable trunk type is: 50 Remote access; 2-way dial tone in/ground start and dial tone out.	No
100	2	Assigns a BCCOS to a Remote Access trunk group.	N/A
101	1	Administers the characteristics for the trunk groups administered in Procedure 100, Word 1.	No
102	1	Assigns the trunk groups (using dial access code) to Miscellaneous Trunk Restriction groups.	Yes
103	1	Assigns Authorization Code requirements for incoming trunks (Field 6). Also assigns echo-suppression trunk signaling in Field 11 as follows: 0 = Normal Dial Tone 1 = Precursor Tone / Dial Tone 2 = Abbreviated Dial Tone / Dial Tone.	Yes
115	1	Assigns termination of Remote Access trunks to an ACD Split when the AES option is used (R2 V3 and later).	No
150	1	Assigns the equipment location to the trunk-group number. The trunk is also dessignated as Remote Access in this procedure.	No

(Continued)

ADMINISTRATION PROCEDURES — REMOTE ACCESS *(Continued)*			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
263	1	Assigns alarming for the Adjunct Enhanced Security unit when used.	No
275	2	Administers shared LDN Access for Remote Access trunks. Specifically, in Field 10: 0 = Direct Dial (no shared LDN access) 1 = Shared LDN Access.	Yes
282	1	Assigns an authorization code and its network access flag.	Yes
285	1	Field 1 specifies the type of Remote Access security for the switch. 0 = barrier code not required 1 = barrier code required 2 = authorization code required 3 = enhanced security required. **NOTE:** When enhanced security is required, authorization code is also required.	Yes
286	1	Assigns Remote Access Time-out to the attendant.	Yes
289	1	Assigns Intercept Treatment for invalid attempts.	Yes
350	1	Assigns the first digit of the feature dial access code and the Remote Access trunk-group dial access codes (if required).	No
350	2	Assigns dial access codes including those used with the Remote Access feature. The encode for attendant use is: 26 Remote Access - change barrier code. The applicable encodes for adjunct enhanced security codes (R2 V4 and Generic 2) are as follows: 101 Enhanced security accept 102 Enhanced security fail 103 Unadministered authorization code entered 104 No authorization code entered.	No

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — REMOTE ACCESS	
PATH NAME	**PURPOSE**
terminal-change system parameters (select the Access-Codes option)	Assigns the Remote Access barrier code to the system class of service

Special Administration Requirements for Adjunct Enhanced Security

When the Adjunct Enhanced Security option is assigned, the switch requires administration that is over and above the administration for the Remote Access feature itself. The applicable administration will depend on the switch version that is being used The following additional administration must be performed to accommodate the enhanced security adjunct.

- On a System 85, Release 2, Version 3 switch:

  Lines serving the adjunct are assigned as an ACD Split. Agent positions should be administered as Auto-Available.

- On a System 85, Release 2, Version 4 or a DEFINITY Generic 2 switch:

  The lines serving the adjunct can be administered as an ACD Split in the same way as for R2V3 or calls can be routed to the adjunct by the Call Vectoring feature.

  Special dial access codes will be used by the adjunct to communicate with the switch. These must be administered using Procedure 350, Word 2, as follows:

Encode 101	Adjunct — Accept
Encode 102	Adjunct — Fail
Encode 103	Unadministered Authorization Code Entered
Encode 104	No Authorization Code Entered.

See either the Automatic Call Distribution feature or the Call Vectoring feature for details on the applicable administration procedures.

Restriction—Attendant Control of Voice Terminals

Description

This feature allows an attendant to activate or cancel restrictions for specific extension numbers or groups of extension numbers. The switch returns Intercept Treatment when a call restricted by this feature is attempted.

Applications

This feature is useful as a mains of placing temporary restrictions on voice terminals. Some possible applications of this feature include

Unauthorized call control

It can be used to control terminals that might be used by unauthorized personnel.

Hotel—Motel Services

In the hospital environment, a guest, who wishes not to be disturbed, can request use of the Controlled Termination Restriction. Also, should a hotel guest decline the use of toll access or direct outward dialing, the attendant can restrict the voice terminal from placing outgoing calls.

Hospital Services Management

In the hospital environment, a doctor may request limited telephone access for a patient so that the patient can remain quiet.

Six Available Restrictions

An attendant can assign any of the following six restrictions to one extension number or to a predetermined group of extension numbers:

1. Controlled Outward Restriction

   Voice terminal users cannot place outgoing calls but can still call a local attendant (using a dial access code), a local attendant (or an attendant in the DCS network) using an LDN, or local voice terminal user (or voice terminal users within the DCS network). The user can receive calls normally.

2. Controlled Terminal-to-Terminal Restriction

   Voice terminal users cannot receive any calls from other local voice terminal users (or users within the DCS network). However, the voice terminal user can receive attendant calls or incoming calls. Also, this restriction does not prevent users from originating calls.

3. Outward and Terminal-to-Terminal Restriction

   Voice terminal users cannot place outgoing calls or receive calls from other local voice terminal users (or users within the DCS network). The voice terminal user can call an attendant and other local voice terminal users (or users within the DCS network). Also, the user can receive incoming outside calls.

4. Controlled Total Restriction

   Voice terminal users cannot receive or place any calls on that extension number.

5. Controlled Termination Restriction

   Voice terminal users cannot receive any calls. Call origination is not affected.

6. Outward and Termination Restriction

   Voice terminal users cannot place outgoing calls or receive any calls. However, the user can call a local attendant (or an attendant in the DCS network) or other local voice terminal users (or users within the DCS network).

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to the feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Activate an Attendant Controlled Restriction

For a single extension number:

1. Press an idle loop button. [PA lamp goes out.]
2. Press **[START]** . [Dial tone]
3. Dial the access code for a single extension number. [Silence]
4. Dial the 1-digit code for the desired restriction. [Second dial tone]
5. Dial the extension number to be restricted. [Confirmation tone]
6. Press **[RELEASE]** . [PA lamp lights.]

For a predesignated group of extension numbers:

1. Press an idle loop button. [PA lamp goes out.]
2. Press **[START]** . [Dial tone]
3. Dial the access code for a group of extensions. [Silence]
4. Dial the 1-digit code for the restriction to be applied. [Second dial tone]
5. Dial the 2-digit group number (e.g., "03"). [Confirmation tone]
6. Press **[RELEASE].** [PA lamp lights.]

To Deactivate an Attendant Controlled Restriction

For a single extension number:

1. Press an idle loop button. [PA lamp goes out.]
2. Press **[START]** . [Dial tone]
3. Dial the access code for a single extension. [Silence]
4. Dial **[0]** . [Second dial tone]
5. Dial the extension number that is to be unrestricted. [Confirmation tone]
6. Press **[RELEASE]** . [PA lamp lights.]

For a group of extension numbers:

1. Press an idle loop button. [PA lamp goes out.]
2. Press **[START]** . [Dial tone]
3. Dial the access code for a group of extensions. [Silence]
4. Dial **[0]** . [Second dial tone]
5. Dial the 2-digit group number (e.g., "04"). [Confirmation tone]
6. Press **[RELEASE]** . [PA lamp lights.]

Considerations

Restriction Groups

An attendant can control a total of 63 restriction groups.

Any number of extension numbers can be assigned to a restriction group. However, an individual extension number can be assigned to only one group.

Multiappearance Extensions

If an attendant activates a restriction toward an extension number with multiple appearances, the restriction applies to every appearance of the extension number.

Hard Processor Swaps

Attendant controlled restrictions are stored in a translation portion of switch memory. Therefore, if an attendant activates a controlled restriction and then a hard processor swap occurs, the restriction will endure the hard swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

ACD (Automatic Call Distribution)

When Controlled Termination Restriction is applied to the individual extension of an ACD (or EUCD) agent, direct calls to the agent do not terminate at the agent's terminal. However, calls to the ACD (or EUCD) split terminate normally at the agent's position.

Automatic Callback

If any of the Attendant Control of Voice Terminals restrictions are activated on either the calling or called extension after the Automatic Callback call origination (but prior to the start of the call completion sequence), the switch ignores the restrictions.

Automatic Route Selection

Extensions with Controlled Outward Restriction or Controlled Total Restriction applied cannot access Automatic Route Selection.

Bridged Call

The Attendant Control of Voice Terminals feature allows an attendant to apply a restriction to an extension or a predefined group of extensions. When one of these restrictions is applied to a ***shared extension,*** the restriction applies to every image of the extension.

Call Coverage

If Controlled Termination Restriction is activated toward an extension with coverage active, the restriction overrides, and calls to that extension do not route to coverage.

Call Forwarding—Busy and Don't Answer

Call Forwarding—Busy and Don't Answer cannot be activated to forward calls to a terminal that is ***already restricted*** by Controlled Terminal-to-Terminal, Controlled Termination, or Controlled Total restriction. When this is attempted, the switch returns intercept tone.

If calls are ***already being forwarded*** to a voice terminal before an attendant activates a termination restriction against the forwarded-to terminal, these forwarded calls are allowed to terminate to the restricted terminal.

If a voice terminal has forwarding activated, and then an attendant activates a termination restriction against the forwarding terminal, calls to the forwarding terminal do not terminate or forward. The switch returns intercept tone to the calling party.

Call Forwarding—Don't Answer

Call Forwarding—Don't Answer cannot be activated to forward calls to a terminal that ***is already restricted*** by Controlled Terminal-to-Terminal, Controlled Termination, or Controlled Total restriction. When this is attempted the switch returns intercept tone.

If calls are ***already being forwarded*** to a voice terminal before an attendant activates a termination restriction against the forwarded-to terminal, these forwarded calls are allowed to terminate to the restricted terminal.

If a voice terminal has forwarding activated, and then an attendant activates a termination restriction against the forwarding terminal, calls to the forwarding terminal do not terminate or forward. The switch returns intercept tone to the calling party.

Call Forwarding—Follow Me

Call Forwarding—Follow Me cannot be activated to forward calls to a terminal that ***is already restricted*** by Controlled Terminal-to-Terminal, Controlled Termination, or Controlled Total restriction. When this is attempted the switch returns intercept tone.

If calls are ***already being forwarded*** to a voice terminal before an attendant activates a termination restriction against the forwarded-to terminal, these forwarded calls are allowed to terminate to the restricted terminal.

If a voice terminal has forwarding activated, and then an attendant activates a termination restriction against the forwarding terminal, calls to the forwarding terminal do not terminate or forward. The switch returns intercept tone to the calling party.

Call Pickup

A voice terminal that is othewise restricted from receiving calls (by the Controlled Terminal-to-Terminal, Outward and Terminal-to-Terminal, Controlled Termination, or Outward and Termination Restriction) is allowed to pickup group members' calls using Call Pickup.

Call Vectoring

An attendant is not allowed to activate an Attendant Control of Voice Terminals restriction against a VDN. When this is attempted, the switch returns intercept tone.

Call Waiting

If an Attendant Control of Voice Terminals restriction prevents a call from terminating at a terminal, Call Waiting is also prevented to that terminal.

Data Call Setup

Restrictions applied to a data module (i.e., for the keyboard dialing subfeature) can be bypassed through the use of the one button data call transfer procedures unless the same restrictions are applied to the activating voice terminal.

Data Communications Access

An attendant can restrict selected terminals from access to the DCA feature with the Attendant Control of Voice Terminals feature. An attempt to access a DCA port by one of these terminals is redirected to the attendant for screening. These attendant-extended calls are denied Data Protection.

Data Protection

Data Protection—Temporary is not available on attendant-extended calls. If the voice terminal extension is denied direct access to the trunk by the activation of this feature, Data Protection—Temporary cannot be used. Data Protection—Permanent can be provided, however.

DDC (Direct Department Calling)

If any Attendant Control of Voice Terminals restriction (Outward, Terminal-to-Terminal Only Calling, Termination, or Total) is assigned to a DDC controlling terminal or a DDC group extension, then the restriction assigned is applied to the entire group.

Direct Inward Dialing

Through the use of the Attendant Control of Voice Terminals feature, Direct Inward Dialing can be denied as a means of access to terminals that ordinarily can be reached by this method.

Direct Outward Dialing

Through the use of the Attendant Control of Voice Terminals feature, Direct Outward Dialing can be denied to terminals which ordinarily have this capability.

DCS (Distributed Communications System)

From a voice terminal user's perspective, the Attendant Control of Voice Terminals feature is transparent in the DCS environment. For example, whenever a restricted voice terminal is allowed to place calls to or receive calls from a local voice terminal, calls can also be placed to or received from voice terminals residing in other DCS nodes.

From an attendant's perspective, the Attendant Control of Voice Terminals feature is not transparent in the DCS environment. An attendant residing in the same node as the voice terminal must activate and deactivate these restrictions. If an attendant tries to activate a restriction against a voice terminal in a different node, the switch returns intercept treatment.

EUCD (Enhanced Uniform Call Distribution)

Same as ACD (Automatic Call Distribution) interactions.

Host Computer Access

An attendant can restrict selected terminals from access to the Host Computer Access trunk groups with the Attendant Control of Voice Terminals feature. An attempt to access a Host Computer Access port from a restricted terminal is redirected to an attendant. If the attendant is to screen these calls, a voice terminal equipped with transfer capabilities must be provided near the attendant console to perform the transfer. The transfer cannot be performed from the attendant console.

Hunting

When a call is placed directly to a restricted voice terminal in a hunt group, the switch returns Intercept Treatment to the calling party. The call does not hunt.

However, when a call is placed to an unrestricted terminal in a hunt group, the call can hunt to and/or through the restricted terminal.

Look-Ahead Interflow

An attendant is not allowed to activate an Attendant Control of Voice Terminals restriction against a Look-Ahead Interflow VDN. When this is attempted, the switch returns intercept tone.

Override

The use of Override toward a terminal with Controlled Termination restriction active is denied.

Personal Central Office Line

Personal Central Office Line calls are not affected when an attendant activates a restriction toward an extension on a voice terminal.

Precedence Calling

Attendant Control of Voice Teminals functions normally for Precedence calling calls. Calls to or from a restricted voice terminal route to the attendant priority queue for processing.

Priority Calling

If an attendant restricts call termination to a voice terminal, Priority Calling is also restricted to that terminal.

Queuing

A call cannot be placed in a ringback queue at the local switch if a terminating restriction is in effect for the calling terminal. However, the call can be placed in a ringback queue at a tandem switch. When the callback attempt is made from a tandem switch to a subtending switch, the call appears as an ordinary incoming tie trunk call to the subtending switch. Consequently, the calling party will not receive the callback attempt.

Restriction—Voice Terminal Restrictions

Administered restrictions applied to voice terminals using the Voice Terminal Restrictions feature are checked by System 85 or DEFINITY Generic 2 software before attendant-activated restrictions using the Attendant Control of Voice Terminals feature. However, the administered restrictions are not applied in preference to the attendant-activated restrictions. Instead, the restraints applied to a uers's calling privileges ***can accumulate*** as additional restrictions are either activated by an attendant or assigned by the switch administrator. As an example, if Origination Restriction were assigned to a voice terminal by the switch administrator and an attendant activated Controlled Termination Restriction toward the same terminal, calls could not be placed or received at the terminal.

Straightforward Outward Completion

Controlled Outward Restriction can be bypassed using the Straightforward Outward completion feature.

Tenant Services

In general practice, the Attendant Control of Voice Terminals feature allows an attendant to activate and cancel temporary restrictions for a specific voice terminal or a predefined group of voice terminals. The Tenant Services feature limits the operation of this feature.

An attendant (in a partition other than Attendant Partition 0) is allowed to activate or cancel restrictions for specific extensions. However, this operation is only allowed when the extension resides in an extension partition that is assigned to the attendant's partition. If the attendant ties to activate or cancel a restriction for an extension in any other partition, the switch returns intercept treatment to the attendant.

An attendant (in a partition other than Attendant Partition 0) is also allowed to activate or cancel restrictions for controlled restriction groups. However, this operation is only allowed when the controlled restriction group has been assigned to the attendant's partition in Procedure 270, Word 2. If the attendant tries to activate or cancel a restriction for any other controlled restriction group, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to activate and cancel restrictions for any extension or any controlled restriction group in the switch.

It is the responsibility of the system manager to assign extensions to controlled restriction groups. However, there are no tests in Procedure 000, Word 2 to ensure partitioning of the 63 groups. It is strongly recommended that every member of each group belong to the same extension partition.

Through Dialing

Controlled Outward Restriction can be bypassed using the Through Dialing feature.

UCD (Uniform Call Distribution)

If any Attendant Control of Voice Terminals restriction (Outward, Terminal-to-Terminal Only Calling, Termination, or Total) is assigned to a UCD controlling terminal or a UCD group extension, then the restriction assigned is applied to the entire group.

Hardware Requirements

None.

Feature Administration

Assignment of the Attendant Control of Voice Terminals feature is on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES
RESTRICTION—ATTENDANT CONTROL OF VOICE TERMINALS

PROCEDURE	WORD	PURPOSE	SMT
000	2	Assigns a controlled restriction group number (from 1 to 63) to an extension number.	Yes
075	1	Displays the voice terminals assigned to a controlled restriction group.	Yes
204	1	Administers the desired alphanumeric display for restricted calls. The applicable encode is as follows: R2 V1 to R2 V3: 293 Controlled restriction R2 V4 and later: 2293 Controlled restriction.	No
350	1	Assigns the first digit of the feature dial access codes.	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 27 Single extension control 28 Group of extension control.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS RESTRICTION—ATTENDANT CONTROL OF VOICE TERMINALS	
PATH NAME	**PURPOSE**
terminal-change group restriction	Views or prints a report of the controlled restriction groups.
terminal-change extensions attributes	Assigns a controlled restriction group number to an extension number.

Restriction—Code Restriction

Description

This feature allows restriction of calls by selected extension numbers to areas defined by specific area codes and/or office codes. The feature designates the 3-digit area codes, 3-digit office codes, and 6-digit combinations of area codes and office codes that a restricted voice terminal user is allowed to access. The switch returns intercept tone whenever the caller dials a code that is not allowed to the caller.

Each of the 63 terminal classes of service is assigned a code restriction level of 1, 2, 3, or 0. Likewise, each selected code is assigned a level of 1, 2, 3, or 0. The level "1" is the most restricted, and the level "0" is the least restricted. For example, a voice terminal user with an extension number assigned as level 2 can access any code with a restriction level of 1 or 2. An extension number assigned as level 0 can access any of the specified codes.

The following table shows the code restriction levels that are allowed.

TABLE 104-A. Allowable Code Restriction Levels

Voice Terminal	Area Code or Area Code and Office Code 1	2	3	0
1	x			
2	x	x		
3	x	x	x	
0	x	x	x	x

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

None

Considerations

Number of Allowable Codes for Trunk Groups

Calls to the public network are made using CO (Central Office) or FX (Foreign Exchange) trunk groups. Code Restriction may limit access to area and/or office codes on as many as five CO trunk groups and on as many as four FX trunk groups. There is one list of allowed codes that serves the CO trunk groups; whereas, each FX trunk group is served by a separate list of allowed codes.

Code Restriction Parameters

For CO trunk groups, calls can be allowed to a maximum of 1000 codes. This 1000-code (or less) list can have as many as:

- 160 area codes
- 800 office codes for the home area
- 40 six-digit combinations of area codes and office codes.

FX Trunk Groups

For FX trunk groups, calls can be independently allowed to a maximum of 40 codes. This 40-code (or less) list can be any assortment of area codes, office codes, and combinations of area codes and office codes.

Code and Toll Restriction Together

The same trunk group cannot use Code Restriction and Toll Restriction together.

Digit Absorption (Procedure 301, Word 2)

The Code Restriction feature denies calls to certain area and / or office codes over a trunk group that an otherwise unrestricted party has accessed with the dial access code. Since calling parties use the dial access code to access such a trunk group, Touch-Tone Calling Senderized Operation does not apply to these calls. Instead, the System 85 or DEFINITY Generic 2 seizes an outgoing (or 2-way) trunk from within the CO or FX trunk group, and the serving CO returns public-network dial tone.

As the calling party dials the destination digits that are immediately sent over the trunk, the Code Restriction feature monitors and verifies the dialed digits. Whenever the user dials a restricted area code or office code, the System 85 or DEFINITY Generic 2 tears down the established connection to the serving CO and denies the call.

However, some step-by-step (electromechanical) COs, to minimize the needed selector banks, do not use the full seven digits (NXX - XXXX) to route calls to certain office codes. And, if these COs receive the full seven digits, they "absorb" the leading digit(s) before routing the call. So, for example, the calling party could either dial 484 + 1234 or 4 + 1234 to place a call to the same destination.

Therefore, the Code Restriction feature needs a way to screen both of the possible digit strings that a step-by-step CO would route to the same destination. This capability is provided with Procedure 301, Word 2. This procedure emulates the digit-absorption plan used by the serving CO. In this way, as the Code Restriction feature monitors the dialed digits, the System 85 or DEFINITY Generic 2 can ignore digits that the CO will absorb and either deny or allow calls (according to the assignment in Procedure 301, Word 3) based on the first three digits that the CO actually uses to route the call.

Tie Trunks

The Code Restriction feature cannot be assigned to tie trunks.

Attendant Position

Attendant positions are not code restricted. Therefore, voice terminal users can access restricted codes with attendant assistance.

Hard Processor Swaps

Code restriction assignments are stored in a translation portion of switch memory. Therefore, these restrictions will endure a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

AAR (Automatic Alternate Routing)

On System 85 and DEFINITY Generic 2.1 switches, the Code Restriction feature has no effect on calls placed using the AAR feature. When a voice terminal user dials the AAR access code (followed by RNX + XXX) to access the private network, this access is usually limited by the user's FRL (not by the Code Restriction feature).

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, Code Restriction does not apply to calls placed through ARS. If Code Restriction and ARS are provided on the same switch, Miscellaneous Trunk Restriction is recommended for the ARS trunk groups to prevent dial access to these trunk groups by those terminals assigned Code Restriction.

Bridged Call

A Code Restriction level is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When a Code Restriction level is assigned to a ***shared extension,*** the level applies to every image of the extension.

Call Vectoring

The Code Restriction feature does not limit the routing of "route to" steps to destinations outside the switch. "Route to" steps utilize networking feature software to route these calls, and Code Restriction checks are not made by the networking features.

Look-Ahead Interflow

The Code Restriction feature does not limit the routing of Look-Ahead Interflow "route to" steps outside a sending switch. "Route to" steps utilize networking feature software to route these calls, and Code Restriction checks are not made by the networking features.

Restriction—Toll Restriction

Code Restriction and Toll Restriction cannot be used together in the same extension class of service.

Route Advance

Route Advance routes outgoing calls over alternate facilities when the first trunk group (in the route advance pattern) is busy. When a CO trunk group is assigned to the CO (primary) code restriction list, any associated route advance CO trunk groups with the same restrictions need not be assigned to individual code restriction lists. They default to the same restrictions as the CO trunk group assigned to the CO code restriction list. However, if a CO trunk group is assigned to an FX (secondary) code restriction list and is in a route advance sequence, every CO trunk group in the sequence must be assigned to a separate code restriction list and given the same allowed codes. If any of the CO trunk groups in the sequence are not assigned to an FX code restriction list, they default to the CO (primary) list.

Straightforward Outward Completion

Straightforward Outward Completion bypasses Code Restriction.

Through Dialing

Through Dialing bypasses Code Restriction.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, the Code Restriction feature has no effect on calls placed using the WCR feature. When a caller dials a WCR network access code access is usually controlled by the FRL, BCCOS, and other administrable parameters of the WCR feature, not by the Code Restriction feature.

Hardware Requirements

None.

Feature Administration

Assignment of the Code Restriction feature is on a per-extension class of service and per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — RESTRICTION—CODE RESTRICTION			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the extension class of service to an extension number.	Yes
010	3	Assigns a Code Restriction level (1, 2, 3, or 0) to a terminal class of service.	Yes
301	1	Administers a Code Restriction trunk group and trunk type.	Yes
301	2	Emulates the digit-absorption plan used by a trunk group's serving step-by-step CO.	Yes
301	3	Assigns a Code Restriction level (1, 2, 3, or 0) to a 3-digit office or area code.	Yes
301	4	Specifies which office codes are toll.	Yes
302	1	Assigns allowed 6-digit strings of area codes and office codes to a trunk group.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — RESTRICTION—CODE RESTRICTION	
PATH NAME	PURPOSE
terminal-change class-of-service attributes	Assigns a Code Restriction level (1, 2, 3, or 0) to a voice terminal class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Notes:

Restriction—Miscellaneous Trunk Restrictions

Description

This feature restricts preselected (by class of service) voice terminals from the use of certain trunk groups [such as, WATS (Wide Area Telecommunications Service), Loudspeaker Paging, or Call Park]. A voice terminal user who is restricted from and tries to access these trunk groups receives intercept tone. The switch administrator determines the trunks to be restricted. Each switch is individually tailored to meet the customer's need.

Restriction Groups

A maximum of eight restriction groups can be provided, with each group containing one to four trunk groups. Any number of the eight restriction groups can be assigned to each of the 63 extension classes of service.

Feature History and Development

This feature was first available on System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

None.

Considerations

Hard Processor Swaps

Miscellaneous trunk restrictions are stored in a translation portion of switch memory. Therefore, these restrictions will endure a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

AAR (Automatic Alternate Routing)

Miscellaneous Trunk Restriction does not apply to a call using Automatic Alternate Routing for routing.

ARS (Automatic Route Selection)

Miscellaneous Trunk Restriction does not apply to a call using Automatic Route Selection for routing.

Bridged Call

Miscellaneous Trunk Restrictions are assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When these restrictions are assigned to a shared extension, the restrictions apply to every image of the extension.

Conference—Three Party

A terminal restricted from accessing a trunk group due to the Miscellaneous Trunk Restrictions feature can be added to a conference via Conference—Three Party involving the restricted trunk group. However, the restricted terminal cannot originate a conference via Conference—Three Party involving the restricted trunk group.

Data Communications Access

Local voice terminal extensions may be restricted from accessing a DCA trunk group by the Miscellaneous Trunk Restriction feature.

Host Computer Access

Local voice terminal extensions may be restricted from accessing a Host Computer Access trunk by the Miscellaneous Trunk Restriction feature.

Look-Ahead Interflow

The Miscellaneous Trunk Restrictions feature denies selected extensions dial access to preselected trunk groups. Miscellaneous Trunk Restrictions do not limit preference selection using AAR/ARS patterns. Therefore, when Miscellaneous Trunk Restrictions and Look-Ahead Interflow are both assigned, the Miscellaneous Trunk Restrictions feature does not apply to Look-Ahead Interflow calls being routed with a "route to" step.

Precedence Calling

The Miscellaneous Trunk Restrictions feature does not restrict access to Precedence Capable trunk groups in an AUTOVON Access configuration. However, when trunk-group dial access codes are used to access ROUTINE Only trunk groups, this access can be limited by Miscellaneous Trunk Restrictions.

Straightforward Outward Completion

If an attendant determines that the terminal should be able to complete the call, the Miscellaneous Trunk Restriction checks are bypassed when the attendant places the call for the voice terminal user and then releases when ringback is returned.

Tenant Services

Miscellaneous Trunk Restrictions can be assigned to an extension class of service in the normal manner. However, the capacity of this feature has not been increased for use with the Tenant Services feature. There remains a capacity of four trunk groups in each of eight restriction groups. With this limited capacity, the Miscellaneous Trunk Restrictions feature is only useful for limited applications (e.g., to restrict Loudspeaker Paging or Call Park).

Through Dialing

The Through Dialing feature operates normally from the restricted terminal at an attendant's discretion. If the attendant determines that the terminal should be allowed access to the trunk, the Miscellaneous Trunk Restriction checks are bypassed when the attendant seizes the trunk and then realeases to allow the terminal user to complete the dialing.

Transfer

The Transfer feature can transfer a restricted terminal (due to the Miscellaneous Trunk Restrictions feature) to a restricted trunk group. However, the restricted terminal cannot originate a transfer via the Transfer feature to a restricted trunk group.

WCR (World Class Routing)

Miscellaneous Trunk Restriction does not apply to a call using World Class Routing for trunk selection.

Hardware Requirements

None.

Feature Administration

Assignment of the Miscellaneous Trunk Restrictions feature is on a per-extension class of service and per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES RESTRICTION—MISCELLANEOUS TRUNK RESTRICTIONS			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the voice terminal class of service to an extension number.	Yes
010	3	Assigns Miscellaneous Trunk Restriction groups to an extension class of service.	Yes
102	1	Assigns trunk groups (using dial access codes) to Miscellaneous Trunk Restriction groups.	Yes
175	1	Displays the trunk group(s) (by dial access code) in a Miscellaneous Trunk Restriction group.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS RESTRICTION—MISCELLANEOUS TRUNK RESTRICTIONS	
PATH NAME	PURPOSE
terminal-change class-of-service attributes	Assigns Miscellaneous Trunk Restriction groups to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.
terminal-change group trunk-restrictions	Assigns trunk groups (using dial access codes) to Miscellaneous Trunk Restriction groups.

Restriction—Toll Restriction

Description

This feature restricts callers at specified extension numbers from placing toll calls by using trunk group DACs (Dial Access Codes). Users at restricted extensions can place toll calls with attendant assistance. The Toll Restriction feature includes two types of restrictions:

- *Battery Reversal*

  The central office sends a battery reversal signal to the System 85 or DEFINITY Generic 2 when the caller dials a toll call. The switch then checks the calling extension's class of service. If the extension is toll restriction the switch releases the connection to the central office, and returns intercept tone.

- *0/1 Toll Restriction*

  This form of toll restriction is controlled within the local switch. For System 85 or DEFINITY Generic 2.1 switches, if the first or second digit dialed following the trunk-group DAC is a 0 or 1, and the first three digits of the called address are not in the free (allowed) call list, toll restricted extensions are given intercept tone. For Generic 2.2 switches, if the first digit dialed after the trunk group DAC is 0 or 1, and the first three digits of the called address are not in the free (allowed) call list, the call is assumed to be toll.

Free Call List

The free call list is administered in Procedure 300, Word 1. It provides up to ten customer selected 3-digit codes such as area codes [NPA (Numbering Plan Area)] or special service codes that are considered to be toll free. This allows access by toll restricted extensions to selected area codes and service codes such as 411, 800, or 911. If the code is on the free call list, the call is allowed. If not, the call is denied (even if it would not incur toll charges).

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

None.

Considerations

Attendant Position

Attendant positions are not toll restricted. The attendant can extend a call from a restricted extension (and tie-trunk user) to the toll network.

Credit Card Calls

With Toll Restriction, it will not be possible to dial credit card calls using a trunk group dial access code from toll restricted extensions, even though the toll charges are to be applied elsewhere. (Credit card calls can be placed using a networking feature such as ARS or WCR.)

Multiple Types of Toll Restrictions

Both types of Toll Restriction (battery reversal and 0/1 restriction) can be used in the same switch but are mutually exclusive on a per-trunk group basis. The networking features (AAR, ARS, and WCR) each have their own form of toll call control which can also be used independently from the Restrictions—Toll Restriction feature.

Hard Processor Swaps

Toll Restriction assignments are stored in a translation portion of switch memory. Therefore, these restrictions will endure a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

AAR (Automatic Alternate Routing)

The Toll Restriction feature has no limiting effect on AAR calls placed using the AAR access code. Toll Restriction denies toll calls placed over specific trunk groups using the ***trunk-group*** access code. To limit an AAR users ability to place toll calls, a low FRL should be assigned to the user's class of service in Field 23 of Procedure 010, Word 3.

ARS (Automatic Route Selection)

The Toll Restriction feature has no limiting effect on ARS calls placed using the ARS access code. Toll Restriction denies toll calls placed over specific trunk groups using the ***trunk-group*** access code.

To restrict an ARS user from placing toll calls, ARS Toll Restriction should be assigned to the user's class of service in Procedure 010, Word 3, field 22. Toll tables are then used to identify toll vrs non-toll routes.

Bridged Call

Toll Restriction is assigned to a class of service in Procedure 010, Word 3 (field 21). The class of service is then assigned to an extension in Procedure 000, Word 1. When Toll Restriction is assigned to a ***shared extension,*** the restriction applies to every image of the extension.

Call Vectoring

The Toll Restriction feature does not limit the routing of "route to" steps to destinations outside the switch. "Route to" steps utilize AAR, ARS, or WCR software to route these calls. Each of these features has its own way of providing toll call limitation and Toll Restriction feature checks are not made by these networking features.

Code Calling Access—Traditional and Universal

Extensions with Toll Restriction in the extension class of service may not use the number "1" as a first or second digit of the Code Calling Access or answer-back code.

Look-Ahead Interflow

The Toll Restriction feature does not limit the routing of Look-Ahead Interflow "route to" steps outside a sending switch. "Route to" steps utilize AAR, ARS, or WCR software to route these calls. Each of these features has its own way of providing toll call limitation and Toll Restriction feature checks are not made by these networking features.

Restriction—Code Restriction

Code Restriction and Toll Restriction cannot be used together in the same extension class of service.

Straightforward Outward Completion

The Straightforward Outward Completion feature bypasses Toll Restriction.

Through Dialing

Through Dialing bypasses Toll Restriction.

WCR (World Class Routing)

The Toll Restriction feature has no limiting effect on WCR calls placed using a network access code. Toll Restriction denies toll calls placed over specific trunk groups using the ***trunk-group*** access code.

To restrict a WCR user from placing toll calls, WCR Toll Restriction must be assigned to the user's class of service in Procedure 010, Word 3, field 22. Toll tables are then used to identify toll vrs non-toll routes.

Restricting Feature Use

The Toll Restriction feature can be overridden for a terminal by an attendant.

Hardware Requirements

None.

Feature Administration

Assignment of the Toll Restriction feature is on a per-extension class of service and/or per incoming dial repeating tie trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRTION PROCEDURES
RESTRICTION—TOLL RESTRICTION

PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the extension number and extension class of service to an equipment location.	Yes
010	3	Assigns Toll Restriction to an extension class of service (field 21).	Yes
101	1	Assigns battery reversal to a trunk group.	No
300	1	Administers the free call list (a list of unrestricted codes) for the system.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS
RESTRICTION—TOLL RESTRICTION

PATH NAME	PURPOSE
terminal-change class-of-service attributes	Administers Toll Restriction to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Restriction—Voice Terminal Restrictions

Description

The Voice Terminal Restrictions feature provides six fixed restrictions to control voice terminals. Voice Terminal Restrictions control call receiving and originating permissions at the affected voice terminals. These restrictions are applied through COS (Class of Service) assignment.

Six Types of Restriction

Inward Restriction

Inward restrictions prevents voice terminal users at specified extension numbers from receiving public network calls (DID and CO trunk calls). These calls will be intercepted. These voice terminals can receive local calls and private network calls.

Manual Terminating Line Restriction

Manual Terminating Line Restriction prevents voice terminal users at specified extension numbers from receiving calls other than direct or extended calls from a local attendant (or an attendant within the DCS network). All other calls route to the appropriate intercept. These voice terminal users can originate calls and activate features.

Origination Restriction

Origination Restriction prevents voice terminal users on specified extension numbers from originating calls. If voice terminal users attempt to place calls, their extension number is taken out of service after 10 seconds and returned to service when the user hangs up.

Outward Restriction

Outward Restriction prevents callers on specified extension numbers from directly accessing outgoing trunks to the public network. Calls can be made to other local voice terminal users (or voice terminal users within the DCS network), to a local attendant [or an attendant in the DCS network (using an LDN)], to a local attendant (using a dial access code) and to tie trunks. An attendant can assist in completing calls to outside numbers using the Straightforwad Outward Completion or the Through Dialing feature.

Terminal-to-Terminal Only Calling Restriction

Terminal-to-Terminal Only Calling Restriction restricts all calls from specified extension numbers except those made to local extension numbers (or extension numbers within the DCS network). All other calls are intercepted.

Termination Restriction

Termination Restriction prevents voice terminal users on specified extension numbers from receiving calls but not from originating calls.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

None.

Considerations

Multiappearance Extensions

If a Voice Terminal Restriction is assigned to an extension number with multiple appearances, the restriction applies to every appearance of the extension.

Dial Access Capabilities of Origination Restricted Voice Terminals

Origination restricted single-appearance voice terminal users, after answering a call, can place calls on hold and access the following features:

- Automatic Callback
- Call Pickup
- Call Waiting
- Conference—Three Party
- Hold
- Leave Word Calling
- Transfer.

Hard Processor Swaps

Voice terminal restrictions are stored in a translation portion of switch memory. Therefore, these restrictions will endure a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINTIY Generic 2 features affect or are affected by the operation of this feature.

APLT (Advanced Private Line Termination)

An attendant cannot extend incoming automatic tie trunk calls to an inward restricted voice terminal (intercept tone is returned to the attendant). Incoming dial repeating tie trunk calls can be completed to an Inward restricted voice terminal if the calling party dials the voice terminal number (the call is treated like a voice terminal-to-voice terminal call).

ACD (Automatic Call Distribution)

When Termination Restriction is applied to the individual extension number of an ACD agent, direct calls to the agent do not terminate at the agent's terminal. However, calls to the ACD split terminate normally at the agent's position.

AAR/ARS (Automatic Alternate Routing/Automatic Route Selection)

On System 85 or DEFINITY Generic 2.1 switches, extensions with any of the following Voice Terminal Restrictions cannot access either AAR or ARS:

- Origination Restriction
- Outward Restriction
- Terminal-to-Terminal Only Calling.

Bridged Call

Voice Terminal Restrictions are assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When one (or more) of these restrictions is assigned to a ***shared extension,*** the restriction applies to every image of the extension.

Call Coverage

If Termination Restriction is assigned to an extension with coverage active, the restriction overrides, and calls to that extension do not redirect to coverage.

Call Forwarding—Busy and Don't Answer

Incoming calls on public network trunks may not forward to an Inward Restricted voice terminal.

Call Forwarding functions normally for an Origination restricted voice terminal when an attendant activates forwarding for the voice terminal. However, the Origination restricted voice terminal is only allowed to activate Call Forwarding from a hold or recall dial tone state. The Origination restricted voice terminal is not allowed to activate Call Forwarding from an idle state.

Calls may not be forwarded to a voice terminal with Voice Terminal Restrictions (Termination, Manual Terminating Line, or Terminal-to-Terminal Calling) assigned.

Call Forwarding—Don't Answer

Calls may not forward to a voice terminal with Restriction—Voice Terminal Restrictions (Termination, Manual Terminating Line, or Terminal-to-Terminal Calling) activated.

Incoming calls on public network trunks may not forward to an Inward Restricted terminal.

Call Forwarding—Don't Answer functions normally for an Origination restricted terminal when the attendant or a single-line voice terminal user activates Call Forwarding—Don't Answer from a hold or recall dial tone state. An Origination restricted terminal cannot activate Call Forwarding—Don't Answer from an idle state.

If an unrestricted voice terminal is assigned as the forwarded-to voice terminal and then Restriction—Voice Terminal Restrictions (Inward, Termination, Terminal-to-Terminal Only, or Manual Terminating Line) restriction is activated in its extension class of service, calls will still forward to the voice terminal.

Call Forwarding—Follow Me

Incoming calls on public network trunks may not forward to an Inward Restricted voice terminal.

Call Forwarding functions normally for an Origination restricted voice terminal when an attendant activates forwarding for the voice terminal. However, the Origination restricted voice terminal is only allowed to activate Call Forwarding from a hold or recall dial tone state. The Origination restricted voice terminal is not allowed to activate Call Forwarding from an idle state. Therefore, if Call Forwarding—Follow Me is activated, only an attendant can deactivate it.

Calls may not be forwarded to a voice terminal with Termination, Manual Terminating Line, Inward, or Terminal-to-Terminal Only Calling Restriction assigned.

If an unrestricted voice terminal is designated as the forwarded-to voice terminal and then Voice Terminal Restrictions (Inward, Termination, Terminal-to-Terminal Only Calling, or Manual Terminating Line) restriction is assigned to its extension class of service, calls still forward to the voice terminal.

Call Pickup

A voice terminal with Voice Terminal Restrictions (Inward, Termination, Terminal-to-Terminal Only Calling, or Manual Terminating Line) assigned may pickup a call directed to another voice terminal in the restricted voice terminal's call pickup group.

Call Vectoring

Voice terminal restrictions that are assigned to a VDN's class of service ***are not applied*** to the VDN's vector processing. As an example, if Termination Restriction is assigned to Class of Service 1, and Class of Service 1 is assigned to VDN 7300, this restriction is ignored by the Call Vectoring feature. Calls are allowed to terminate to the vector.

Voice terminal restrictions do not limit the routing of "route to" steps to an answering destination. For example, if Origination or Outward Restriction is assigned to a VDN's class of service, this assignment is ignored.

Call Waiting

If a Voice Terminal Restriction prevents a call from terminating at a terminal, Call Waiting is also prevented to that terminal.

Conference—Attendant Five Party

An Inward restricted terminal may be added to an established conference involving an incoming trunk.

Conference—Attendant Six Party

An attendant can add Inward restricted voice terminal to an attendant conference involving an incoming trunk.

Conference—Three Party

An Inward restricted voice terminal may be added to an established connection involving an incoming trunk via the Conference—Three Party feature.

An Outward restricted voice terminal can access a public network trunk if the restricted voice terminal calls an unrestricted voice terminal. From here, the unrestricted voice terminal could use the Conference—Three Party feature to connect the restricted voice terminal to the public network.

An origination-restricted single-appearance voice terminal may originate a call from a 2-party connection by first accessing the Conference—Three Party feature.

Data Call Setup

Restrictions applied to a data module (for example, for the keyboard dialing subfeature) can be bypassed through the use of the 1-button data call transfer procedures unless the same restrictions are applied to the activating voice terminal.

DCS (Distributed Communications System)

From a voice terminal user's perspective, the Voice Terminal Restrictions feature is transparent in the DCS environment. For example, whenever a restricted voice terminal is allowed to place calls to or receive calls from a local voice terminal, calls can also be placed to or received from voice terminals residing in other DCS nodes.

EUCD (Enhanced Uniform Call Distribution)

When Termination Restriction is applied to the individual extension number of an EUCD agent, direct calls to the agent do not terminate at the agent's terminal. However, calls to the EUCD split terminate normally at the agent's position.

Hold

The user of an origination-restricted single-appearance voice terminal can place a calling party on hold. From this state, the user of restricted voice terminal can originate a call, activate a feature, or return to the held call.

When the origination-restricted user places a call on hold, the voice terminal is treated as a ***fully unrestricted terminal*** unless another restriction (such as Outward Restriction or Terminal-to-Terminal Only Calling) has also been applied to the terminal. If another restriction does apply, the origination-restricted user's dialing capabilities are limited to the capabilities allowed by the second restriction.

Hunting

When a call is placed directly to a restricted voice terminal in a hunt group, the switch returns Intercept Treatment to the calling party. The call does not hunt.

However, when a call is placed to an unrestricted voice terminal in a hunt group, the call can hunt to and/or through the restricted terminal.

Look-Ahead Interflow

At a sending or receiving switch, voice terminal restrictions that are assigned to a VDN's class of service ***are not applied*** to the VDN's vector processing. As an example, if Termination Restriction is assigned to Class of Service 1, and Class of Service 1 is assigned to VDN 7300, this restriction is ignored by the Call Vectoring feature. Calls are allowed to terminate to the vector.

At a sending (or tandeming) switch, voice terminal restrictions do not limit the routing of "route to" steps to an answering destination. For example, if Origination or Outward Restriction is assigned to a VDN's class of service, this assignment is ignored.

Override

The use of Override toward a terminal with Termination or Manual Terminating Line Restriction active is denied.

Personal Central Office Line

Incoming calls on Personal Central Office Lines are not affected when the Voice Terminal Restrictions (Termination) feature is assigned to a extension class of service.

Priority Calling

When Termination Restriction is assigned to a voice terminal, Priority Calling is denied toward that terminal.

Queuing

A call from a local terminal cannot be placed in a ringback queue if Termination or Manual Terminating Line restriction is in effect at the calling terminal.

A callback attempt made between switches (for example in a Main / Satellite or Tandem Network arrangement) toward a voice terminal with either Termination or Manual Terminating Line restriction in effect is also denied.

Remote Access

The Remote Access user cannot complete a call to a system terminal if any of the following features are active on the called extension:

- Manual Terminating Line
- Termination Restriction.

Restriction—Attendant Control of Voice Terminals

Administered restrictions applied to voice terminals using the Voice Terminal Restrictions feature are checked by System 85 or DEFINITY Generic 2 software before attendant-activated restrictions using the Attendant Control of Voice Terminals feature. However, the administered restrictions are not applied in preference to the attendant-activated restrictions. Instead, the restraints applied to a user's calling privileges ***can accumulate*** as additional restrictions are either activated by the attendant or assigned by the switch administrator. As an example, if Origination Restriction were assigned to a voice terminal by the switch administrator and an attendant activated Controlled Termination Restriction toward the same voice terminal, calls could not be placed or received at the terminal.

Straightforward Outward Completion

Straightforward Outward Completion bypasses Outward Restriction.

Through Dialing

Through Dialing bypasses Outward Restriction.

Transfer

The Transfer feature may be used to transfer an incoming trunk call to an Inward restricted voice terminal.

An Outward restricted voice terminal can access a public network trunk if the restricted voice terminal calls an unrestricted voice terminal. The unrestricted voice terminal uses the Transfer feature to connect the restricted voice terminal to the public network trunk.

An origination-restricted single-appearance voice terminal may originate a call from a 2-party connection by first accessing the Transfer feature.

A voice terminal with the Terminal-to-Terminal Only Calling restriction assigned can be transferred to the attendant by another voice terminal user via the Transfer feature. A Terminal-to-Terminal Only Calling restricted voice terminal may transfer a call to another voice terminal but not to the attendant.

Unattended Console Service—Call Answer From Any Voice Terminal

Dialing the Call Answer From Any Voice Terminal access code is denied from an Inward Restricted terminal, Terminal-to-Terminal Only Calling terminal, or Origination Restricted terminal.

Unattended Console Service—Preselected Call Routing

The Preselected Call Routing feature takes precedence over the Voice Terminal Restrictions (Inward, Termination, Terminal-to-Terminal Only Calling, and Manual Terminating Line) feature. Therefore, incoming trunk calls are allowed to terminate on the preselected voice terminal, and the preselected terminal is allowed to initiate outgoing trunk calls with these restrictions active. This is the only facet of the restrictions that is overridden. For example, all calls, besides incoming trunk calls, are denied unless the calls are attendant completed for manual terminating line restricted voice terminals.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, extensions with any of the following Voice Terminal Restrictions cannot access WCR:

- Origination Restriction
- Outward Restriction
- Terminal-to-Terminal Only Calling.

Hardware Requirements

None.

Feature Administration

Assignment of the Voice Terminal Restriction feature is on a per-extension class of service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, the feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES RESTRICTION—VOICE TERMINAL RESTRICTIONS			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the class of service to an extension number.	Yes
010	3	Assigns the type of Voice Terminal Restriction to an extension class of service (Fields 15 through 20).	Yes

The following are the the applicable TCM path names used with the AP 16.

TCM SCREENS RESTRICTION—VOICE TERMINAL RESTRICTIONS	
PATH NAME	PURPOSE
terminal-change class-of-service attributes	Assigns the type of Voice Terminal Restriction to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Notes:

Ringing — Abbreviated and Delayed

Description

The Ringing — Abbreviated and Delayed feature provides voice terminal users with manual transfer or delayed automatic transfer of ringing. The ringing can be directed to any subset of the voice terminals sharing an appearance with the primary terminal(s).

Automatic Transfer

With automatic transfer of ringing the primary terminal(s) rings for a predetermined number of cycles (2, 4, 8, or 16 rings). After the ringing cycle, ringing transfers to the preassigned voice terminal(s) sharing the appearance with the primary voice terminal.

Manual Transfer

With manual transfer of ringing the currently ringing call can be transferred to the secondary voice terminal(s). Manual transfer is done by pressing the ABRV RING button.

Prior to the transfer, the other voice terminal(s) does not ring.

TABLE 108-A. Ringing Characteristics by Type

Ringing Type (Encode)	Before Transfer	After Transfer
No Ringing (0)	Voice terminal doesn't ring.	Voice terminal doesn't ring.
Ringing (1)	Voice terminal rings.	Voice terminal rings.
Delayed Ringing (2)	Voice terminal doesn't ring.	Voice terminal rings.
Abbreviated Ringing (3)	Voice terminal rings.	Voice terminal doesn't ring.

Feature History and Development

This feature was first available for System 85 with Release 1. With previous releases, single-appearance voice terminals were not allowed to participate in Abbreviated and Delayed Ringing relationships. However, beginning with the R2 V2, Issue 1.2 software package, one single-appearance terminal (administered as a straight line set) can participate in each of these relationships.

User Operations

The following is the user operating procedure for this feature.

To Manually Transfer Ringing at a Primary Terminal to a Secondary Terminal:

1. Be sure the primary terminal is ringing.
2. Press **[ABRV RING]** .

Considerations

Automatic and Manual Ringing Transfer

An appearance of the primary extension may have both automatic and manual transfer of ringing. This combination is obtained by administering automatic and manual transfer of Procedure 052, Word 2, Field 3 and administering the ABRV RING button in Procedure 054, Word 1.

Straight Line Sets

Manual transfer of ringing, requiring an ABRV RING button, cannot be assigned to a Straight line set.

Abbreviated Ringing Without Delayed Ringing

Abbreviated Ringing can also be assigned to ***unshared*** appearances. When this is done, the unshared appearance rings for the specified number of ringing cycles (2, 4, 8, or 16) and then the ringing stops while the green status lamp continues flashing.

As an example, a multiappearance voice terminal has three unshared appearances. Ringing is assigned to the first appearance, while Abbreviated Ringing (set for two ringing cycles) is assigned to the second and third appearances. In this way, when the user of this voice terminal is active on the first appearance, a subsequent call that terminates to the second or third appearance will only ring twice. Silencing this ringing after two ringing cycles limits the audible disturbance to the active call until the second call is either redirected to coverage, answered by the called party, or abandoned by the calling party.

> **NOTE:** The called party can also press the Send All Calls button to hasten the redirection of the new call. Pressing this button immediately redirects the ringing call and silences the ringing on the called terminal.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 feature affect or are affected by the operation of this feature.

Call Coverage

The Send All Calls and Cover All functions of the Call Coverage feature take precedence over Abbreviated and Delayed Ringing. These functions control the redirection of ringing.

The Cover Don't Answer function of the Call Coverage feature takes precedence over Abbreviated and Delayed Ringing when the amount of ringing cycles used to time both features are equal.

The details of the don't answer condition are as follows: If the timing interval for the coverage group ***is less than or equal to*** the timing interval for Abbreviated and Delayed Ringing, terminating calls redirect to coverage without ringing the images(s) assigned delayed ringing. However, if the timing interval for the coverage group ***is greater than*** the timing interval for Abbreviated and Delayed Ringing, terminating calls first ring the abbreviated ringing image(s). Then, ringing transfers to the delayed ringing image(s), and these images ring for the rest of the Cover Don't Answer timing interval. After the Cover Don't Answer interval elapses, the call redirects to coverage.

Since Call Coverage takes precedence over Abbreviated and Delayed Ringing, ring-ping, if assigned in Release 2, Version 4, is heard for immediately redirected calls at the called party's terminal, except if delayed ringing (encode 2) is active.

Call Forwarding—Busy and Don't Answer

Call Forwarding—Busy and Don't Answer takes precedence over Abbreviated and Delayed Ringing when the amount of ringing cycles used to time both features are equal.

The details of the don't answer condition are as follows: If the timing interval for call forwarding ***is less than or equal to*** the timing interval for Abbreviated and Delayed Ringing, terminating calls forward to the destination extension without ringing the image(s) assigned delayed ringing. However, if the timing interval for call forwarding ***is greater than*** the timing interval for Abbreviated and Delayed Ringing, terminating calls first ring the abbreviated ringing image(s). Then, ringing transfers to the delayed ringing image(s), and these images ring for the rest of the call forwarding timing interval. After the call forwarding timing interval elapses, the call forwards to the destination extension.

Call Forwarding—Don't Answer

Call Forwarding—Don't Answer takes precedence over Abbreviated and Delayed Ringing when the amount of ringing cycles used to time both features are equal.

The details are as follows: If the timing interval for call forwarding ***is less than or equal to*** the timing interval for Abbreviated and Delayed Ringing, terminating calls forward to the destination extension without ringing the image(s) assigned delayed ringing. However, if the timing interval for call forwarding ***is greater than*** the timing interval for Abbreviated and Delayed Ringing, terminating calls first ring the abbreviated ringing image(s). Then ringing transfers to the delayed ringing image(s), and these images ring for the rest of the call forwarding timing interval. After the call forwarding timing interval elapses, the call forwards to the destination extension.

Call Forwarding—Follow Me

Call Forwarding—Follow Me takes precedence over Abbreviated and Delayed Ringing. Call Forwarding—Follow Me controls the redirection of ringing.

Ringing Transfer

Ringing Transfer takes precedence over Abbreviated and Delayed Ringing. If an extension on a voice terminal has both an Abbreviated and Delayed Ringing button and a Ringing Transfer button, each button provides independent ringing characteristics to the images of an appearance. The ringing characteristics for Abbreviated and Delayed Ringing are specified using the Ring (Alert) Type (Field 11 of Procedure 052, Word 1), while the characteristics for Ringing Transfer are specified using the Ringing Transfer Encode (Field 4 of Procedure 052, Word 2).

Hardware Requirements

Multiappearance voice terminals are primarily used for this feature. However, one single-appearance voice terminal (administered as a straight line set) can share an appearance with as many as 15 other multiappearance voice terminals. This single-appearance voice terminal is allowed to participate in an Abbreviated and Delayed Ringing relationship.

Feature Administration

Assignment of the Abbreviated and Delayed Ringing feature is on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Adminstration Panel). The customer can fully administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES RINGING — ABBREVIATED AND DELAYED			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
052	1	Administers the type of ringing to an image (bridged appearance) on a multiappearance voice terminal or straight line set. Assign ***abbreviated ringing*** to the image on the primary terminal and ***delayed ringing*** to the image on the secondary voice terminal.	Yes
052	2	Assigns either automatic or manual transfer of ringing to both terminals.	Yes
054	1	Assigns the ABRV RING button to a primary voice terminal. The applicable encode is as follows: 13 Abbreviated and Delayed Ringing.	Yes
061	1	Specifies the number of ringing cycles (2, 4, 8, or 16) used with automatic transfer of ringing.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS RINGING — ABBREVIATED AND DELAYED	
PATH NAME	**PURPOSE**
terminal-change system parameters (select the Signaling option)	Specifies the number of ringing cycles used with automatic transfer of ringing.
terminal-change terminal buttons	Assigns the ABRV RING button to a primary voice terminal.
terminal-change terminal equipment	Assigns the type of ringing to an image (bridged appearance) on a multiappearance voice terminal or straight line set. (Assign ***abbreviated ringing*** to the image on the primary terminal and ***delayed ringing*** to the image on the secondary terminal.) Also, use this screen to assign either automatic or manual transfer of ringing to ***both*** terminals.

Notes:

Ringing Cutoff

Description

The Ringing Cutoff feature turns off ringing at a particular terminal without affecting the status lamp functions.

This feature is useful during a conference, meeting, recording session, or any time that audible ringing might result in an unacceptable interruption.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Activate Ringing Cutoff:

1. Be sure the feature isn't activated. [The RINGING CUTOFF green status lamp is dark.]
2. Press **[RINGING CUTOFF]** . [The RINGING CUTOFF green status lamp lights.]

To Deactivate Ringing Cutoff:

1. Be sure the feature is activated. [The RINGING CUTOFF green status lamp is lit.]
2. Press **[RINGING CUTOFF]** . [The RINGING CUTOFF green status lamp goes out.]

Considerations

RINGING CUTOFF Buttons

A terminal should have only one RINGING CUTOFF button. Additional RINGING CUTOFF buttons serve no useful purpose.

Hard and Soft Processor Swaps

Ringing Cutoff activations are stored in a status portion of switch memory. Therefore, if a voice terminal user activates Ringing Cutoff and then a hard processor swap occurs, Ringing Cutoff will not be activated after the hard swap finishes. At this time, the user can reactivate Ringing Cutoff.

The Ringing Cutoff feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Call Coverage

When Send All Calls and Ringing Cutoff are both active at a called voice terminal, ring-ping is not provided as the call directs to coverage.

Call Forwarding—Follow Me

When Call Forwarding—Follow Me and Ringing Cutoff are both active at a called voice terminal, ring-ping is not provided as the call forwards.

Intercom—Automatic, Dial, and Manual

Ringing Cutoff overrides all intercom ringing for the given terminal.

Manual Signaling

Ringing Cutoff denies Manual Signaling for the given terminal.

Hardware Requirements

Multiappearance voice terminals are required for this feature.

Feature Administration

Assignment of the Ringing Cutoff feature is on a per-terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can fully administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following is the applicable administration procedure.

ADMINISTRATION PROCEDURE — RINGING CUTOFF			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
054	1	Assigns the RINGING CUTOFF button to a multiappearance voice terminal. The applicable encode is as follows: 11 Ringing Cutoff.	Yes

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — RINGING CUTOFF	
PATH NAME	**PURPOSE**
terminal-change terminal buttons	Assigns the RINGING CUTOFF button to a multi-appearance voice terminal.

Notes:

Ringing — Distinctive

Description

This feature lets voice terminal users distinguish between various types of incoming calls. There are three distinctive patterns of ringing that indicate the types of incoming call. The patterns of ringing are:

- One-burst ringing — Terminal-to-terminal calls and remote-access calls
- Two-burst ringing — Attendant calls or incoming trunk calls (calls from outside the switch)
- Three-burst ringing — Priority calls, including Automatic Callback, Priority Calling, Callback, or calls to attendants that are answered by designated voice terminals assigned Preselected Call Routing.

For each analog set, ringing patterns are generated by the module processor that serves the set's port circuit. Table 110-A shows ringing-pattern timing for analog sets served by traditional module port circuits and universal module port circuits. Single carrier cabinet modules (also known as XE modules) use the same timing as universal modules. Regardless of the module type, all three patterns repeat every 5.2 seconds.

For ISDN—BRI and DCP digital sets, ringing patterns are generated by the sets in response to messages sent from the switch. Default ringing-pattern timing for digital sets may be slightly different from the timing provided for analog sets. Some digital sets allow custom ringing patterns to be specified by the sets' users. Customized ringing patterns can vary radically from default ringing patterns.

TABLE 110-A. Distinctive Ringing Cycles for Analog Terminals

Ringing Pattern		Ringing Cycle (Seconds)
1-Burst Ringing	Traditional Module	ring 1.1, quiet 4.1
	Universal Module	ring 0.9, quiet 4.3
2-Burst Ringing	Traditional Module	ring 0.2, quiet 0.4, ring 0.5 quiet 4.1
	Universal Module	ring 0.4, quiet 0.2, ring 0.3, quiet 4.3
3-Burst Ringing	Traditional Module	ring 0.2, quiet 0.1, ring 0.2, quiet 0.1, ring 0.5, quiet 4.1
	Universal Module	ring 0.2, quiet 0.1, ring 0.2, quiet 0.1, ring 0.3, quiet 4.3

Feature History and Development

This feature was first available for System 85 in Release 1.

The only change to this feature has been the difference in intracycle timing between traditional and universal modules, introduced with the universal module on the DEFINITY Generic 2.1 switch.

User Operations

None.

Considerations

Other Sources of Distinctive Ringing

The Distinctive Ringing feature applies only to terminals whose ringing pattern is provided and controlled by the switch. Specific terminals such as DCP (Digital Communications Protocol) and ISDN—BRI (Basic Rate Interface) voice terminals ring in response to messages from the switch. While the general pattern (1-burst, 2-burst, and 3-burst ringing) may conform to the distinctive ringing patterns of this feature, the timing and specific form of ringing provided is controlled by the terminal rather than the switch. In this way, some of these terminals can provide customized ringing patterns which can be programmed locally. These characteristics are not part of the Distinctive Ringing feature.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Data Protection

When permanent Data Protection is assigned to a class of service, all ringing for that class of service is 1-burst ringing.

Manual Signaling

When Distinctive Ringing is in progress, Manual Signaling is denied. Broken fluttering on the Manual Signaling status lamp is used to show denial.

Precedence Calling

The Precedence Calling feature uses 3-burst Distinctive Ringing to alert an idle appearance when a precedence call (PRIORITY or higher) is ringing at the station.

Queuing

Distinctive ringing for a callback call applies to local switch terminals only. Whenever a callback call is directed over a tie trunk, 1-burst ringing is provided by the local switch.

Ringing Cutoff

Ringing Cutoff silences distinctive ringing for the terminal.

Tenant Services

The switch provides 1-burst ringing for terminal-to-terminal calls inside an extension partition. The switch also provides 1-burst ringing for terminal-to-terminal calls between an extension partition and Extension Partition 0.

Otherwise, terminal-to-terminal calls are not allowed to cross partition boundaries. In order to call a voice terminal in any other partition, a voice terminal user must dial the appropriate 7-digit number that routes the call over a CO (Central Office) trunk. When this is done, the switch provides 2-burst ringing at the called voice terminal.

The switch provides 2-burst ringing for incoming calls from the public network in the usual manner.

In a partitioned System 85 or DEFINITY Generic 2, the switch provides the usual 2-burst ringing for direct attendant calls or attendant-extended calls to voice terminals (whenever these calls are allowed by the Tenant services feature).

When Priority Calling, Override, Automatic Callback, or On-Hook Queuing are provided on the switch and allowed in the partitioned environment, the switch provides 3-burst ringing in the usual manner.

Transfer

When a local voice terminal user transfers an incoming trunk call to another local extension, the switch provides 1-burst ringing for the transferred-to voice terminal.

Hardware Requirements

None.

Feature Administration

Distinctive Ringing is provided with all systems and requires no assignment.

Notes:

Ringing Transfer

Description

The Ringing Transfer feature allows a multiappearance voice terminal user to transfer all ringing for a given extension number to other voice terminal(s). When Ringing Transfer is active for an extension, a call terminating to an appearance of that extension rings a predefined subset of the other terminals sharing the same appearance.

For example, an executive who normally receives all calls can transfer ringing to the secretary's voice terminal. This is useful when the executive is out of the office or otherwise occupied.

Feature History and Development

This feature was first available for System 85 in Release 1. In early releases, single-appearance voice terminals could not be used with this feature. However, beginning with R2 V2, Issue 1.2, a single-appearance terminal (administered as a straight line set) can function as the secondary (that is, transferred-to) terminal.

User Operations

The following are the user operating procedures for this feature.

To Activate Ringing Transfer:

1. Be sure the feature isn't activated. [The RINGING TRANSFER green status lamp is dark.]
2. Press **[RINGING TRANSFER]** . [The RINGING TRANSFER green status lamp lights.]

To Deactivate Ringing Transfer:

1. Be sure the feature is activated. [The RINGING TRANSFER green status lamp is lit.]
2. Press **[RINGING TRANSFER]** . [The RINGING TRANSFER green status lamp goes out.]

Considerations

RINGING TRANSFER Buttons

Only one RINGING TRANSFER button can be assigned for any one extension number. Ringing Transfer applies only to the extension number associated with the button.

Preselected Subsets

Ringing can be transferred to any preselected subset of the images of an appearance. (An executive with two secretaries, for example, could allow both secretaries to share the first terminating appearance, while allowing only one of the secretaries to share the second terminating appearance. In this way, when Ringing Transfer is active, a call terminating to the first appearance would ring both secretaries, while a call to the second appearance would ring only one secretary.)

Straight Line Sets

Straight line sets (to which RINGING TRANSFER buttons cannot be assigned) cannot function as the primary (transferring) voice terminal in a Ringing Transfer relationship.

Hard and Soft Processor Swaps

Ringing Transfer activations are stored in a status portion of switch memory. Therefore, if a voice terminal user activates Ringing Transfer and then a hard processor swap occurs, Ringing Transfer will not be activated after the hard swap finishes. At this time, the user can reactivate Ringing Transfer.

The Ringing Transfer feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Call Coverage

Call Coverage takes precedence over Ringing Transfer. That is, a call to a terminal with both Ringing Transfer and Call Coverage active will always redirect to coverage when the assigned coverage criteria is met. This is true whether Ringing Transfer has taken place or not Ringing Transfer will occur only when the Ringing Transfer interval is shorter than the Call Coverage redirection interval. In the case when Send All Calls is active, Ringing Transfer will not occur.

Call Forwarding—Follow Me

Call Forwarding—Follow Me takes precedence over Ringing Transfer. Call Forwarding—Follow Me controls the redirection of ringing.

Ringing—Abbreviated and Delayed

Ringing Transfer takes precedence over Abbreviated and Delayed Ringing. If an extension has both an Abbreviated and Delayed Ringing button and a Ringing Transfer button, each button provides ***independent*** ringing characteristics. The ringing characteristics for Abbreviated and Delayed Ringing are specified using the Ring (Alert) Type (Field 11 of Procedure 052, Word 1), while the characteristics for Ringing Transfer are specified using the Ringing Transfer Encode (Field 4 of Procedure 052, Word 2).

Tenant Services

There are no tests in Procedures 052, Word 2 and 054, Word 1 to ensure that the members of a Ringing Transfer relationship belong to the same extension partition. If the Bridged Call feature is correctly partitioned, the Ringing Transfer feature will also be fully partitioned. It is the responsibility of the system manager to ensure that these relationships do not cross partition boundaries.

Hardware Requirements

Multiappearance voice terminals are generally used for this feature. However, a single-appearance terminal (administered as a straight line set) can be used for the secondary (transferred to) terminal.

Feature Administration

Assignment of the Ringing Transfer feature is on a per-extension basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can fully administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES RINGING TRANSFER			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
052	2	Administers Ringing Transfer characteristics (including the extension and appearance number) to an image on a multiappearance voice terminal or straight line set.	Yes
054	1	Assigns a RINGING TRANSFER button to a multiappearance voice terminal. 12 Ringing Transfer.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS RINGING TRANSFER	
PATH NAME	**PURPOSE**
terminal-change terminal buttons	Assigns a RINGING TRANSFER button to a multi-appearance voice terminal.
terminal-change terminal equipment	Administers Ringing Transfer characteristics (including extension and appearance number) to an image on a multiappearance voice terminal or straight line set.

Route Advance

Description

This feature automatically reroutes outgoing calls over alternate trunk groups when the initially-accessed trunk group is busy.

The Route Advance feature offers efficient use of available trunk groups. To provide minimum traffic interference, the first-choice trunk group could be 1-way outgoing trunks. Subsequent trunk groups might be 2-way. The alternate trunk groups are used primarily for incoming traffic. This allows spill over from the first-choice trunk group. The last (fifth-choice) trunk group could be assigned for Remote Access. These remote access trunks, being the last-choice trunk, should then remain virtually unblocked for Remote Access. The trunk-group access code determines the first-choice trunk group.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The Route Advance feature can be accessed in two ways: either through the use of the extension number steering capability administered on the switch or directly by using the trunk group dial access code for the first trunk group in a route advance pattern.

Extension number steering is typically used with the Main/Satellite/Tributary feature to access route advance patterns between the main and subtending switches. extension number steering is used, feature operations are automatic and there special user operations.

Direct access, via a trunk group dial access code, is used with the Prcedence Calling feature for routine only calling. The following are the user operating procedures Route Advance feature when a trunk group dial access code is used.

To Access the Route Advance Feature via a Trunk Group Dial Access Code:

1. Go off-hook. [Dial tone]
2. Dial the trunk group dial access code for the first trunk group in the desired route advance pattern. [Dial tone is silenced when the first digit of the dial access code is dialed. When a trunk has been seized, dial tone is returned.]
3. Dial the destination address code (4- or 5-digit extension number, or 7- or 10-digit office code address) for the station you wish to call. [Call progress tones.]

Considerations

AUTOVON Access

AUTOVON access is normally provided through the Precedence Calling feature. The Route Advance feature can be used, in conjunction with the Precedence Calling feature to provide an overflow route for Routine calls and for calls placed over Routine Only trunk groups. See the Precedence Calling feature description for details on this use of the Route Advance feature for AUTOVON access.

Extension Number Steering

Extension Number (or Multi-Digit) Steering can be used to invoke a dial access code. When the first trunk group in a route advance pattern is accessed in this way, the Route Advance feature functions to extend the number of trunks available through the use of a single steeing code.

Lists

Each Route Advance list can have five trunk groups assigned (one first-choice group and four alternate trunk groups).

Trunk Compatibility

Compatible trunk groups should be assigned to each Route Advance list. That is, all alternate trunk groups should be capable of passing the same rolls to the same locations as the first-choice trunk group. An FX (Foreign Exhange) trunk group should not be used in the same Route Advance list with local CO (Central Office) or WATS (Wide Area Telecommunications Service) trunk groups.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Control of Trunk Group Access

The Route Advance feature overrides Attendant Control of Trunk Group Access unless attendant control is active on the first trunk group in the Route Advance sequence. Attendant Control of Trunk Group Access works for the first trunk group of a Route Advance sequence. However, once the switch has activated a Route Advance sequence, the Attendant Control of Trunk Group Access feature provides no control over trunk groups that appear later in the sequence.

AAR (Automatic Alternate Routing)

The Route Advance feature has no effect on the way the AAR feature selects a preference (trunk group) with a routing pattern. If a trunk group in a routing pattern is also the first trunk group in a Route Advance sequence, the AAR software ignores the Route

Advance pattern and selects an available (and accessible) trunk group from the AAR routing pattern.

ARS (Automatic Route Selection)

The Route Advance feature has no effect on the way the ARS feature selects a preference (trunk group) within an ARS pattern. If a trunk group in an ARS pattern is also the first trunk group in a Route Advance sequence, the ARS software ignores the alternate Route Advance trunk groups while selecting an available (and accessible) trunk group in the ARS pattern.

Call Vectoring

Using extension number steering to steer to a trunk-group dial access code, the first trunk group in a Route Advance sequence can be programmed as the destination of a "route to" vector step. When this is done, the call can route over an idle trunk in an alternate trunk group in the sequence.

Data Call Setup

The Route Advance feature provides access to as many as five trunk groups with a single access code. When every circuit in the first trunk group is busy, the switch checks the other trunk groups for an idle circuit. Since each data trunk group can contain as many as 99 trunks, this arrangement can provide access to as many as 495 DCA ports using a single dial access code.

Data Communications Access

The Route Advance feature is compatible with the DCA feature and can be used to provide access to as many as five DCA trunk groups using a single access code. The switch checks each trunk group, in sequence until an available data port is found. Since each data trunk group can contain up to 99 trunks, the Route Advance can provide access to as many as 495 computer ports with a single dial access code.

Host Computer Access

The Route Advance feature is compatible with the HCA feature and can provided access to as many as five trunk groups (up to 495 HCA ports) using a single access code.

ISN (Information Systems Network) Interface

The Route Advance feature provides access to as many as five trunk groups with a single access code. When every circuit in the first trunk group is busy, the switch checks the next trunk group and soon. With up to 256 trunks in a trunk group, this provides up to 1280 port appearances with a single access code.

Look-Ahead Interflow

Since the Route Advance feature has no effect on the way the AAR, ARS or WCR features select preferences within patterns, Route Advance has no effect on the routing of Look-

Ahead Interflow calls. If a trunk group in a routing pattern is also the first trunk group in a Route Advance sequence, the AAR, ARS, or WCR software ***ignores*** the alternate Route Advance trunk groups while selecting an available (and accessible) trunk group in the pattern.

Main/Satellite/Tributary

To route calls through a Main / Satellite / Tributary configuration, extension number steering can point to the dial access code corresponding to the first trunk group in a Route Advance sequence. When this is done, an alternate route can be selected so that the call can either route over a parallel direct trunk group to the same location or route to another location for subsequent steering to the desired location.

Precedence Calling

The Precedence Calling feature is compatible with the Route Advance feature only for *routine only* calls. An outgoing ROUTINE precedence call can be routed over non-precedence trunks (also known as routine only trunks) when prececdence capable trunks are not available and a non-precedence trunk group is included in the AUTOVON routing pattern. However, even if the non-precedence trunk group is the first trunk group in a route advance pattern, the Route Advance feature is not invoked.

The Route Advance feature is used with the Precedence Calling feature when a route advance pattern is set up with the AUTOVON (or precedence capable) switch as its destination. Calls over these trunk groups are placed using the dial access code for the first trunk group in the route advance pattern. These trunk groups are referred to as *ROUTINE ONLY*. Routine Only route advance patterns can include precedence capable trunk groups at the end of the route advance pattern but should not begin with a precedence capable trunk group.

Queuing

When the Queuing feature is assigned with Route Advance and every trunk group in the Route Advance list is busy, the call queues on the designated trunk group. The switch then periodically scans the trunk group for an available trunk.

If a trunk is not available in the first-choice trunk group, the switch checks the alternate trunk groups for an available trunk. Should the first-choice trunk group not be assigned to the queuing list, the all is not placed in queue. Rather, the switch returns reorder tone (fast busy tone) to the calling party.

Restriction—Code Restriction

When a CO (Central Office) trunk group is assigned to the CO (primary) code restriction list, any associated route advance CO trunk groups with the same restrictions need not be assigned to individual code restriction lists. They default to the same restrictions as the. CO trunk group assigned to the CO code restriction list. However, if a CO trunk group is assigned to an FX (secondary) code restriction list and is in a route advance sequence, every CO trunk group in the sequence must be assigned to a separate code restriction list and given the same allowed codes. If any of the CO trunk groups in the sequence are not assigned to an FX code restriction list, they default to the CO (primary) list.

Tenant Services

On a partitioned switch, when a user attempts to place a call over an outgoing trunk group, the switch checks to determine whether the user is allowed to use the trunk group and whether there is an available trunk in the group. If one of these conditions is not met, the switch then checks to determine whether a Route Advance list is assigned to the originally dialed trunk group.

- If a Route Advance list (pattern) ***is not assigned:***

  The switch returns intercept treatment to the calling party (for trunk-group denial)

  or

  The switch returns reorder tone or queues the call (for a busy trunk group).

- If a Route Advance list (pattern) ***is assigned:***

  The switch checks the next trunk group in the list to determine whether the user is allowed to use the trunk group and whether there is an available trunk in the group. If one of these conditions is not met, the switch performs the same check on successive trunk groups in the list until:

  The call can be compeletd over an allowable and available trunk,

  or

  If an available trunk cannot be found:

  The call is queued,

  or

  Reorder tone is returned

  or

  If an allowable trunk group cannot be found intercept treatment is returned.

  The recommended approach for assigning Route Advance lists is to assign trunk groups that are dedicated to a partition as the initial trunk groups in the list. These trunk groups can be followed by shared trunk groups that are also accessible to the same partition. However, there are no tests in Procedure 100, Word 4 to prevent other ways of assigning these lists. It is the responsibility of the system manager to ensure that the Route Advance lists are designed in a practical manner.

WCR (World Class Routing)

The Route Advance feature is not used by the WCR feature or by calls routed via the WCR feature. If trunks in a WCR routing pattern are also members of a route advance group, the route advance pattern is not followed when these trunks are encountered.

Hardware Requirements

None.

Feature Administration

Assignment of the Route Advance feature is assigned on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — ROUTE ADVANCE			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
100	1	For System 85, Release 2, Version 3 and earlier switches. Assigns trunk groups to a Route Advance sequence.	No
100	4	For System 85, Release 2, Version 4 and DEFINITY Generic 2. Assigns trunk groups to a Route Advance sequence.	No
330	2	Assigns Route Advance to Queuing.	Yes

Serial Calls

Description

This feature allows an attendant to connect a caller from the public network to two or more voice terminals in succession.

The attendant extends the call and when a talking path is made, places the connected parties on a held loop. After the conversation is finished and while still in a talking condition, the internal voice terminal user momentarily presses the switchhook or the RECALL button to reconnect the attendant. With the attendant back in a 3-way talking connection, the called pary can go on-hook and the calling party can request additional assistance. When the called party goes on-hook without recalling the attendant, the calling party reconnects to the attendant after 30 seconds.

This feature provides a time and cost savings for a long-distance caller. The separate, shorter calls are attached into a single, longer call. The cost savings occurs since the highest cost per time segment occurs at the beginning of a call. Also, the feature furnishes a professional business image to the customer.

Feature History and Development

This feature was first available for System 85 in Release 1. An administrable recall button was provided for R2 V4 and was also retrofitted to the R2 V2 and R2 V3 software packages.

User Operations

The following are the user operating procedures for this feature.

Voice Terminal User Requesting a Serial Call:

1. Call the attendant using the Dial Access to Attendant feature, or any other appropriate means available.
2. When each step of the Serial Call is completed, flash the switchhook (if you are using a 2500 Series terminal), or press **[RECALL]** .

Attendant Support of a Serial Call:

1. Obtain calling sequence instructions from calling party.
2. Press **[START]** . [Dial tone]
3. Dial the next desired extension,

   or

   Use the DXS button for the next desired extension. [Call-progress tone]

4. Press **[HOLD]** when the called party answers. [Hold lamp lights.]
5. After RECALL is received and the called party goes on-hook, repeat the preceding sequence until the Serial Call has been completed.

Considerations

Two-Party Hold

Two-Party Hold on the attendant console must be assigned when Serial Calls is provided.

The attendant must hold the incoming connection until every call except the last is finished. On the final call, the attendant can release the call from the console after extending the call.

Trunk-to-Trunk Connections

The Trunk-to-Trunk Connection features must be assigned if a serial call is to be extended from an incoming trunk to an outgoing trunk.

Calling Party Recall

The called party must call the attendant at the end of the call before hanging up to eliminate the 30-second wait between calls.

Administrable Recall Buttons

Some multiappearance voice terminals do not have a fixed RECALL button. If RECALL buttons are needed for these voice terminals, RECALL buttons can be assigned to the terminals using Procedure 054, Word 1.

Hard and Soft Processor Swaps

If a hard processor swap occurs while a Serial Call is held on an attendant console, the Serial Call is dropped.

If a hard processor swap occurs while an attendant is connected to the Serial Call, the 2- or 3-party connection endures the hard swap. However, the attendant cannot extend the calling party to another voice terminal during the hard swap.

The Serial Calls feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Auto—Manual Splitting

If the Serial Calls feature is provided, the call is not split when an attendant presses the HOLD button to hold the call on the console.

Bridged Call

Bridging is allowed for a shared appearance that is involved in a Serial Call.

Busy Verification of Lines

Busy Verification of Lines is denied toward a line involved in a serial call.

CDR (Call Detail Recording)

If the calling party in a serial call is an incoming trunk and is marked for CDR, the duration of each called party is recorded separately. If the calling party is a local terminal user, only the outgoing trunks marked for CDR are recorded.

Call Pickup

When Serial Calls is in effect, pressing the RECALL button at a local terminal recalls the attendant. Therefore, a terminal user cannot access the Call Pickup feature during a serial call.

Conference—Attendant Five Party

The Serial Calls feature is denied when the called party is involved in an attendant established conference.

Conference—Attendant Six Party

Serial Calls is denied when the called party is involved in an attendant established conference.

Conference—Three Party

For terminals without CONFERENCE or TRANSFER buttons, when Serial Calls is in effect, pressing the RECALL button at a local terminal or momentarily pressing the switchhook, recalls the attendant. Therefore, these terminal users cannot use the Conference—Three Party feature during a serial call.

Override

The Override feature is denied toward a line or trunk involved in a serial call.

Privacy—Attendant Lockout

When the Privacy—Attendant Lockout feature is assigned, the attendant cannot reenter a trunk-to-terminal/terminal-to-trunk serial call without being recalled. Privacy—Attendant Lockout is automatically disabled for trunk-to-trunk serial calls.

Tenant Services

The Serial Calls feature is naturally partitioned on System 85 and DEFINITY Generic 2. In a properly partitioned switch, an attendant cannot receive a call that breaks the rules of partitioning. The attendant must extend each call according to partitioning rules and, the Serial Calls feature always returns the call to the same attendant console that extended the initial call of the series.

Transfer

When Serial Calls is in effect, pressing the RECALL button at a local terminal recalls the attendant. Therefore, a terminal user cannot access the Transfer feature during a serial call.

Trunk Verification—Attendant and Voice Terminal

The Trunk Verification feature is denied toward a trunk involved in a serial call.

Hardware Requirements

None.

Feature Administration

Assignment of the Serial Calls feature is on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). This feature can be partially administered by the customer using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURE — SERIAL CALLS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
054	1	Assigns the RECALL button to a multiappearance voice terminal without a fixed RECALL button. The applicable encode is: 27 Recall.	Yes
200	1	Assigns the 2-Party Hold capability to the attendant console(s).	No

Straightforward Outward Completion

Description

This feature allows an attendant to complete an outgoing trunk call for a voice terminal user. This feature is used to extend calls under the following circumstances:

- For voice terminals that are outward restricted
- For voice terminals with insufficient FRLs (Facilities Restriction Levels) for the trunks required
- For trunks under attendant control.

By completing outgoing calls, the attendant can screen calls and control their destinations.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Extend a Call

With a DTGS (Direct Trunk Group Selection) button:

1. Be sure a call is at the attendant console. [A 2-way connection is established, the ATND lamp is lit, and the ICI (Incoming Call Identification) display shows the extension number.]
2. Press the appropriate DTGS button. [The call is split away from the attendant, the ANS and SPLIT lamps light and dial tone is heard.]
3. Dial the destination number. [Ringback tone]
4. Press **[RELEASE]** . [ATND, ANS, and SPLIT lamps go dark. The calling voice terminal user hears ringback tone.]

Without a DTGS button:

1. Be sure a call is at the attendant console. [A 2-way connection is established, the ATND lamp is lit, and the ICI display shows the extension number.]
2. Press **[START]** . [The calling voice terminal is split away from the attendant, the SPLIT lamp lights, and dial tone is heard.]
3. Dial the desired trunk-group access code. [The ANS lamp lights, and second dial tone is heard.]

4. Dial the destination extension number. [Ringback tone]
5. Press **[RELEASE]** . [ATND, ANS, and SPLIT lamps go out The caller hears ringback tone.]

To Release From a Call When Ringback Tone Is Not Heard:

1. Press **[CANC]** . [Attendant is reconnected to calling voice terminal. The SPLIT lamp goes out.]
2. Inform the calling voice terminal user that the call was not completed.
3. Press **[RELEASE]** . [Call is disconnected. The ANS and ATND lamps go out.]

Considerations

Through Dialing and Straightforward Outward Completion

The Through Dialing feature is similar to the Straightforward Outward Completion feature. Using Through Dialing, the attendant presses the RELEASE button after hearing second dial tone. At this time, the calling party (instead of the attendant) dials the desired number.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Restriction—Attendant Control of Voice Terminals

Straightforward Outward Completion bypasses Controlled Outward Restriction.

Restriction—Code Restriction

Straightforward Outward Completion bypasses Code Restriction.

Restriction—Miscellaneous Trunk Restrictions

Straightforward Outward Completion bypasses Miscellaneous Trunk Restrictions.

Restriction—Toll Restriction

Straightforward Outward Completion bypasses Toll Restriction.

Restriction—Voice Terminal Restrictions

Straightforward Outward Completion bypasses Outward Restriction.

Tenant Services

The Straightforward Outward Completion feature generally allows an attendant to complete outgoing calls for a voice terminal user in the switch. However, in a partitioned System 85 or DEFINITY Generic 2, this operation is controlled. If an attendant (in a partition other than Partition 0) whose partition is assigned to the voice terminal user's partition* tries to place a call over a trunk that partitioning would not allow the voice terminal user to access, the switch returns intercept treatment to the attendant.

An attendant in Partition 0 can place outgoing calls over any trunk for voice terminal users in any partition. When these attendants press the RELEASE button after making an outgoing connection, the outgoing call is left intact.

Hardware Requirements

None.

Feature Administration

Straightforward Outward Completion is automatically provided on every switch feature assignment or administration is not required.

* An attendant whose partition is not assigned to the voice terminal user's partition cannot access that extension partition's trunk group.

Notes:

Tenant Services

Description

The Tenant Services feature allows a large System 85 or DEFINITY Generic 2 to appear to users of the switch as many small, independent switches. This capability allows a single large switch to be shared among a wide assortment of user groups (referred to as "tenants"). A System 85 or DEFINITY Generic 2 configured as a "partitioned switch" can appropriately serve the telecommunications needs of several situations:

- Major airports
- Industrial parks
- Large medical centers
- Large office complexes.

Tenant Perspective

From a tenant's perspective, the cost advantages of the Tenant Services feature can be significant. Small to medium sized tenants may choose to locate their businesses in a building with Tenant Services to benefit from the enhanced features of a large switching system (such as, WCR [World Class Routing] Message Center, and AUDIX) at a relatively modest cost.

Additional cost advantages can be realized by the tenant. In a Tenant Services arrangement, tenants have access to a single centralized source (the shared System Manager) for assistance with their telecommunications needs. With Tenant Services, many tenants would no longer need their own telecommunications experts or the services of outside consultants.

Owner Perspective

From the perspective of a building owner or developer, the Tenant Services feature can serve as an attraction to prospective tenants. When shared telecommunications services are provided effectively, the owner can benefit through higher occupancy rates. Shared telecommunications services become a natural extension to utilities (such as, water, electricity, and gas) that are already provided to tenants from a common point of access.

In some locations, where permitted by the Public Utilities Commission, managing a shared switch can also provide a source of direct revenue for the owner.

Feature History and Development

This feature was first generally available for System 85 in Release 2, Version 4.

Extension Partitions in the Tenant Services Environment

An extension partition may contain one or more extensions. However, each extension number can only be assigned to one extension partition. The partitioning itself is based on extension numbers, not on appearances, images, or equipment locations. A partitioned System 85 or DEFINITY Generic 2 can contain as many as 1000 extension partitions, numbered 0 to 999.

Voice Terminal Call Origination

A voice terminal user (in a partition other than Extension Partition 0) is allowed to place calls (using an extension number) to any voice terminal residing in the same extension partition or in Extension Partition 0. If the user tries to place a call (using an extension number) to a voice terminal residing in any other extension partition, the switch returns intercpt treatment to the user.

> **NOTE:** These voice terminal users can always dial the appropriate dial access code and 7-digit number to place calls to voice terminals in other extension partitions.

A voice terminal user in Extension Partition 0 is allowed to place calls (using an extension number) to any extension in the switch.

> **NOTE:** The IPA (Interpartition Access) feature can be used in conjunction with the Tenant Services feature to allow calling between extension partitions. For more information, refer to the Interpartition Access chapter of this manual.

Attendant Partitions in the Tenant Services Environment

An attendant partition may contain one or more attendant consoles. Each console (except consoles in Attendant Partition 0) can be a member of more than one attendant partition. A partitioned System 85 or DEFINITY Generic 2 can contain as many as 40 attendant consoles residing in as many as 41 attendant partitions, numbered 0 to 40.

Attendant Call Origination

An attendant (in a partition other than Attendant Partition 0) is allowed to place direct calls (using an extension number) to any voice terminal residing in an extension partition that is assigned to the attendant's partition. If the attendant tries to place a direct call (using an extension number) to a voice terminal residing in any other extension partition, the switch returns intercept treatment to the attendant.

> **NOTE:** These attendants can always access the appropriate trunk group and dial the appropriate 7-digit number to place direct calls or extend calls to voice terminals in these other partitions.

An attendant in Attendant Partition 0 is allowed to place direct calls (using an extension number) to any extension in the switch.

An attendant (in a partition other than Attendant Partition 0) is allowed to place direct outgoing calls over any trunk that is assigned to the attendant's partition. If the attendant tries to place a direct outgoing call over any other trunk group, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to place direct outgoing calls over any trunk in the switch.

If a call (from an extension partition other than Extension Partition 0) is active on a console loop, the attendant is allowed to extend the call to another extension in the same extension partition or to an extension in Extension Partition 0. The attendant can also extend the call over a trunk group to which the extension's partition would otherwise have access.

If an outside call is active on a console loop, the attendant (in a partition associated with the incoming trunk group) is allowed to extend the call to an extension in an extension partition assigned to the attendant's partition. Also, the attendant is allowed to extend the call over any outgoing trunk group that is assigned to the attendant's partition.

Operation of the Attendant Queue

One ***common*** attendant queue serves every attendant partition However, the number of calls waiting to be answered by an attendant partition is ***separately determined*** for each attendant partition.

As is done for an unpartitioned switch, the Calls Waiting Level is assigned for the entire partitioned switch in Procedure 200, Word 1. Using a partitioned switch, this level sets the threshold of calls waiting in the shared queue for each attendant partition before the Call Waiting lamp will flash on that partition's console(s). However, if a console is a member of more than one attendant partition, that console's lamp would not flash until all of the console's partitions had exceeded the threshold. As an example, a partitioned switch has three attendant partitions 1, 2, and 3. The Calls Waiting Level is set to "4" for every partition in Procedure 200, Word 1. Partition 1 has two consoles: A and B. Console B is also the only member of Partition 2. Console C is the only member of Partition 3. Currently, there are 11 calls in the shared queue: 5 calls for Partition 1, 2 calls for Partition 2, and 4 calls for Partition 3. At this time, only Console A's Call Waiting lamp is flashing. Console B's lamp would flash if 3 more calls entered the shared queue for Partition 2, and Console C's lamp would flash if one more call entered the queue for Partition 3.

Figure 115-1 shows the basic partitioning concepts for System 85 and DEFINITY Generic 2. The double arrows in the figure represent the lines of access between partitions. In fact, ***every arrow*** in the figure is intended to convey a double arrow. Those arrows that appear as "single arrows" should be imagined to ***pass behind*** the attendant partition and connect with the "single arrow" on the opposite side of the partition (forming a double arrow between Partition 0 and each extension partition).

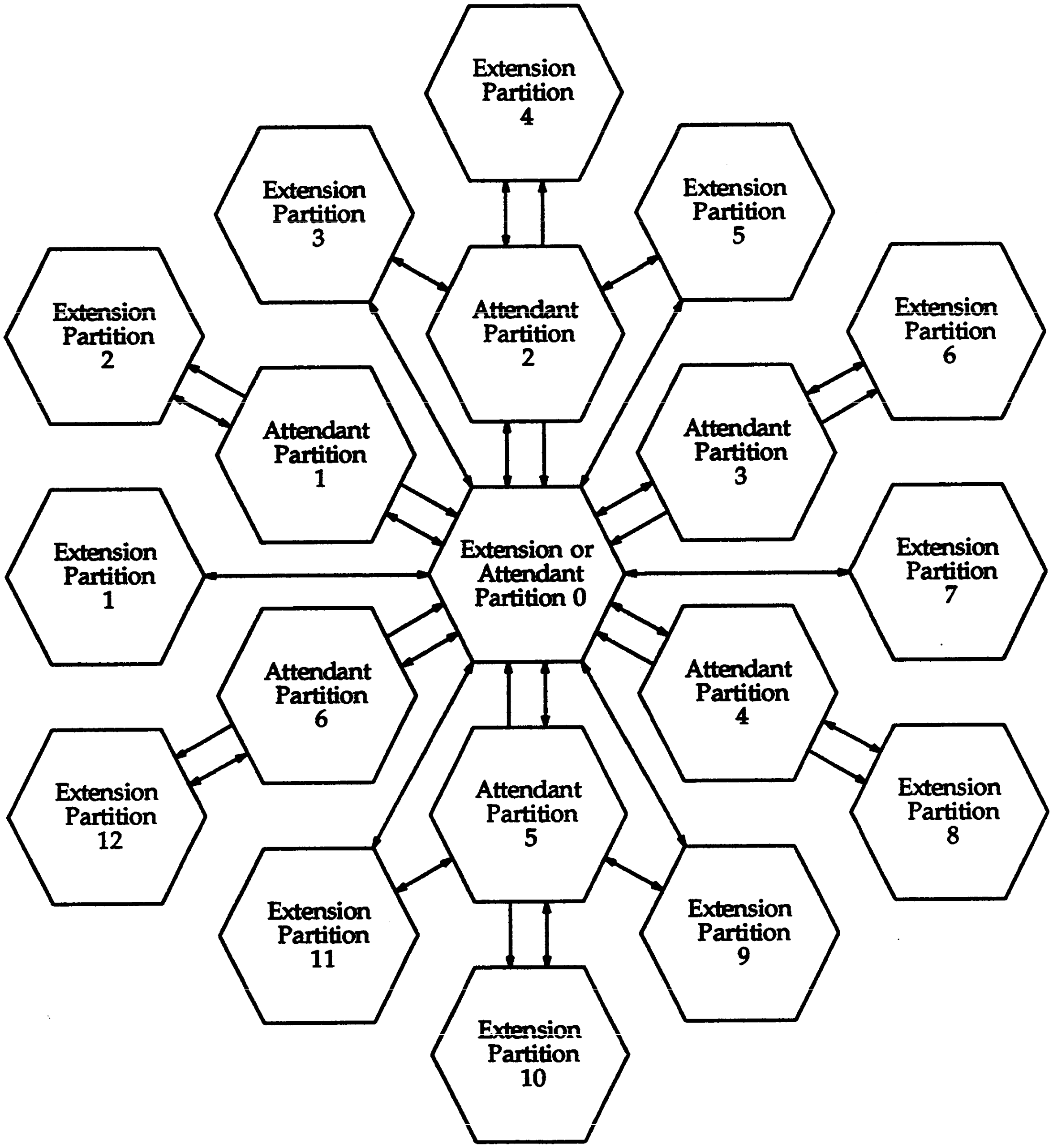

Figure 115-1. Basic Partitioning Concepts

Unattended Console Service in a Partitioned Switch

The Unattended Console Service features have been partitioned as part of the Tenant Services feature. Under this partitioning arrangement, attendant partitions are marked for treatment by the Unattended Console Service features.

General Operation

When a call enters the attendant queue, the switch determines which attendant partition the call should be routed to. If that attendant partition has activated Unattended Console Service, the call is routed to the appropriate night terminal. The algorithm that the switch follows to distribute these calls is similar (but not identical) to the algorithm used for regular Unattended Console Service. The ordered sequence of checks follows.

1. If an incoming call is from over a trunk and that trunk is assigned to a night terminal, the call is handled by trunk-to-terminal processing for that terminal. (Partitioning checks are not performed by the call processing software. These trunk-to-terminal associations should be carefully administered in Procedure 116, word 1 and Procedure 150.)

2. If a trunk-to-terminal association ***has not been assigned,*** the call is handled by processing for the called attendant partition's common night terminal. (The common night terminal is entered by the partition's controlling attendant.)

3. If the common night terminal has not been entered, the call is handled by processing for the called attendant partition's default night terminal. (The default night terminal is assigned in Procedure 270, Word 3.)

4. If an ***available*** night terminal is not found (in Steps 1, 2, or 3), the call routes to the system-wide CAAVT queue (if assigned).

5. If CAAVT is assigned, the call enters the system-wide CAAVT queue. If not, the call remains in the attendant queue, and the switch periodically repeats the sequence of checks.

Controlling Attendant Console

One specific attendant console, the "controlling attendant console," in each attendant partition (except Attendant Partition 0) has special abilities in the attendant partition. The controlling attendant is allowed to activate and deactivate Unattended Console Service for the partition. The controlling attendant is also allowed to assign the common night terminal, or perform the "clear all terminals” function for the partition. If an attendant tries to perform these functions from any other console in the partition, the switch returns intercept treatment to the attendant.

The controlling attendant console for each partition is assigned by the system manager in Procedure 210, Word 2. (One attendant console can serve as the controlling console for more than one attendant partition, but each attendant partition can only have one controlling console.)

UNA (Position Unattended) Button and Lamp

The operation of the UNA button is limited for consoles in a partitioned switch. This button can only be used to activate or deactivate Unattended Console Service when the meaning of a button press is certain. If the console involved is allowed to activate

Unattended Console Service for more than one attendant partition, pressing the UNA button is ignored by the switch. (In this case, the access code must be used for activation or deactivation of Unattended Console Service.)

The UNA lamp lights steadily on an attendant console whenever Unattended Console Service is active for all of the console's attendant partitions. As an example, a particular console belongs to Attendant Partitions 1, 2, and 3. Unattended Console Service is presently active for Partitions 1 and 2. However, this console can still answer calls directed to Partition 3. So, the UNA lamp on this console would remain unlit until Partition 3 also activates Unattended Console Service. As a result, when Unattended Console Service is activated for an attendant partition, the UNA lamp may not light on every console belonging to that partition.

Attendants in Attendant Partition 0

Every attendant in Attendant Partition 0 has full Unattended Console Service capabilities. These attendants are allowed to activate, deactivate, and control Unattended Console Service for any attendant partition in the switch.

In addition, an attendant in Partition 0 can activate or deactivate Unattended Console Service for the entire switch by dialing the fictitious partition number "99" during the activation / deactivation process.

These consoles have similar UNA button and lamp operations as previously described for consoles in other partitions. For example, since an attendant assigned to Partition 0 cannot also belong to another attendant partition, pressing the UNA button on that console activates or deactivates Unattended Console Service for Partition 0.

Attendant Overflow

An attendant partition contains one or more attendant consoles that have been assigned to the partition by the system manager. For general call handling, the call waiting lamp lights on the partition's console(s) when every attendant in the partition is busy, and when a new call for the partition is received.

The attendant overflow function of the Tenant Services feature can provide an alternate operation. In cases where every attendant in a partition is busy, subsequent calls can overflow to another attendant partition (called an "overflow partition”) instead of waiting on the originally called partition.

Attendant overflow can also operate when every attendant console in a partition is in the POSITION BUSY condition.

In Procedure 270, Word 2, the system manager can select one of the following conditions for triggering overflow. These conditions are assigned on a per-attendant partition basis.

- No overflow (default).
- Overflow when every attendant in the partition has the handset (or headset) removed or is in the POSITION BUSY condition.

- Overflow when every attendant in the partition has the handset (or headset) removed, is in the POSITION BUSY condition, ***or*** is using all six of the console's switched loops.

  NOTE: Attendant overflow and overflow destinations cannot be assigned to Attendant Partition 0. However, Attendant Partition 0 can be assigned as the overflow destination for another attendant partition.

Overflow Chains

When attendant overflow occurs, System 85 or DEFINITY Generic 2 call-processing software allows as many as two "links" in an overflow "chain." As an example, Attendant Partition 2 is assigned to overflow to Attendant Partition 5. Meanwhile, Attendant Partition 5 is assigned to overflow to Attendant Partition 9. During periods when Partitions 2 and 5 are quite busy, an incoming call to Partition 2 can overflow to Partition 5, and then, in turn, overflow to Partition 9. If, Attendant Partition 9 is also busy, the call will wait on Attendant Partition 9 (the most recently checked attendant partition).

Network Route Selection in a Partitioned Switch

Figure 115-2 is a simplified picture of partitioned ARS routing. On DEFINITY Generic 2.2 switches, the AAR and ARS features are replaced by the WCR (World Class Routing) feature. The scheme reflected in Figure 115-2 applies to the WCR feature on Generic 2.2 switches, except that with WCR, partitioning can be used for private network routing patterns as well as public network routing patterns, and more routing patterns and time-of-day plans are available. Partitioned ARS provides more flexible access to the 64 ARS patterns (1023 patterns are available with WCR). Several attributes of this figure are worth some discussion.

- ***More than one*** extension partition can route calls via the ***same*** call category.
- With ARS routing (System 85 and Generic 2.1), each call category points to 64 "routing designators" that correspond to the dialed digits (the way ***ARS patterns*** do in an unpartitioned switch).
- In turn, each routing designator points to a "pattern" that has not been previously assigned to that call category's routing designator.
- With WCR routing (Generic 2.2), call category is determined by partition, time-of-day plan, and conditional routing count. Call category is then used with VNI (Virtual Nodepoint Identifier) to select a routing pattern.

  NOTE: The routing designators for Call Category 0 are all pointing to patterns with the ***same*** numbers. This relationship is the ***default*** for unpartitioned switches. If desired, this relationship can be assigned differently.

- Each pattern can have up to 16 "preferences" (as shown for Pattern 2). These preferences are really trunk groups arranged in descending order of desirability.

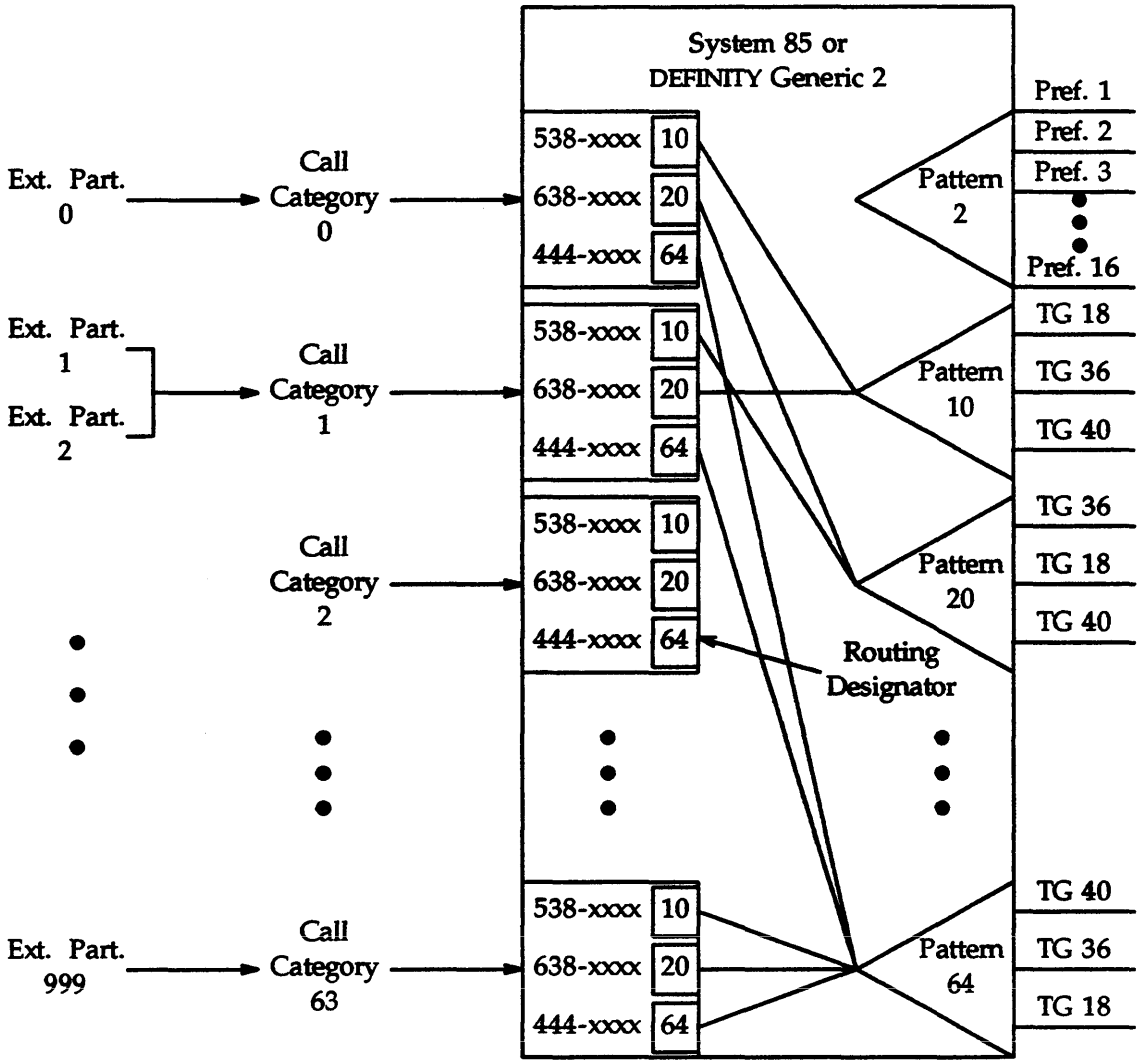

Figure 115-2. Network Routing in the Tenant Services Environment

- Patterns 10, 20, and 64 each contain three preferences Trunk Groups 18, 36, and 40. But, these three preferences are arranged in a different order for each pattern. This type of arrangement is where some of the increased flexibility resides. In this way, when these trunk groups are shared, ***different*** extension partitions can access the ***same*** trunk groups in different preferential orders.

  However, a more basic flexibility is derived from another attribute of this routing scheme. As shown in the drawing, ***different*** extension partitions can access ***different*** patterns even though the same public-network digits are dialed.

  NOTE: Patterns can also have trunk groups that are not assigned to other patterns. But, this arrangement is not shown in this figure.

- An extension partition does not need to be allowed use of every trunk group in every pattern that the partition can use. A partitioned System 85 or DEFINITY

Generic 2 checks the trunk group / partition assignments individually within each pattern.

Whenever a partition's terminals cannot use a trunk group in these patterns, the trunk group is skipped by the ARS / WCR / Tenant Services software. If the voice terminal cannot access any of the trunk groups, the switch denies the call with Intercept Treatment.

The routing shown in Figure 115-2 is based on 3-digit NXXs. This is not a limitation of partitioned routing. Rather, the drawing shows 3-digit NXXs for simplicity. The partitioned software can also screen 3-digit NPAs and 6-digit NPA-NXXs with ARS, or up to 18 digits with WCR.

User Operations

The following are the user operating procedures for this feature.

To Activate Unattended Console Service for Every Partition

An attendant in Partition 0 should:

1. Press an idle loop button. [PA lamp goes out.]
2. Press the **[START]** button. [Dial tone]
3. Dial the Activate Unattended Console Service access code. [Second dial tone]
4. Dial the fictitious partition number **[9] [9]** . [The switch returns confirmation tone, and the UNA lamp lights on every console.]
5. Press the **[RELEASE]** button.

To Deactivate Unattended Console Service for Every Partition

An attendant in Partition 0 should:

1. Press an idle loop button.
2. Press the **[START]** button. [Dial tone]
3. Dial the Deactivate Unattended Console Service access code. [Second dial tone]
4. Dial the fictitious partition number **[9] [9]** . [The switch returns confirmation tone, and the UNA lamp goes out on every console.]
5. Press the **[RELEASE]** button. [PA lamp lights.]

To Activate Unattended Console Service for a Partition

The partition's controlling attendant should:

1. Press an idle loop button. [PA lamp goes out]
2. Press the **[START]** button. [Dial tone]
3. Dial the Activate Unattended Console Service access code. [Second dial tone]
4. Dial the appropriate partition number (e.g, "09"). [Confirmation tone]
5. Press the **[RELEASE]** button. [If the attendant is associated with another "attended" attendant partition, PA lamp lights.]

NOTE: Any attendant in Partition 0 can perform this operation for any attendant partition in the switch.

To Deactivate Unattended Console Service for a Partition

The partition's controlling attendant should:

1. Press an idle loop button. [PA lamp goes out.]
2. Press the **[START]** button. [Dial tone]
3. Dial the Deactivate Unattended Console Service access code. [Second dial tone]
4. Dial the appropriate partition number (e.g., "40"). [Confirmation tone]
5. Press the **[RELEASE]** button. [PA lamp lights.]

NOTE: Any attendant in Partition 0 can perform this operation for any attendant partition in the switch.

To Activate Unattended Console Service for a Partition Using the UNA Button

The partition's controlling attendant (who is a member of only one attendant partition) should:

Press the **[UNA]** button. [UNA lamp lights, and PA lamp goes out.]

NOTE: An attendant in Partition 0 can perform this operation for Partition 0.

To Deactivate Unattended Console Service for a Partition Using the UNA Button

The partition's controlling attendant (who is a member of only one attendant partition) should:

Press the **[UNA]** button. [UNA lamp goes out, and PA lamp lights.]

NOTE: An attendant in Partition 0 can perform this operation for Partition 0.

Considerations

Switch Capacities

A single System 85 or DEFINITY Generic 2 can contain as many as 1000 Extension Partitions, 41 Attendant Partitions, and 999 trunk groups. From these 999 trunk groups, as many as 982 trunk groups can be dedicated to or shared by tenants.

Legal Considerations

Laws governing the use of shared telecommunications services vary in different locations, and are subject to change. It is the responsibility of the System Manager to understand and comply with the applicable regulations.

Partitioned Trunk Types

Partitioned trunk groups can be dedicated for use by a single partition or shared by several partitions. Most trunk groups with commonly used trunk types can be partitioned. The trunk-type encodes that allow for partitioning include

- 12 to 50 (APLT, CO, FX, WATS, DID, Tie, and Remote Access trunks)
- 70 to 78 (Special tie trunks)
- 103 to 109 (PDM, TDM, AP 32 DCPI, EIA, ISN, and DMI trunks)
- 120 (ISDN Dynamic trunks).

Trunk groups that are not included here cannot be partitioned. These unpartitionable trunk groups are ***equally accessible*** to any user in the switch.

Trunk-to-Trunk Partitioning

When an incoming trunk call is connected to an outgoing trunk, a partitioned System 85 or DEFINITY Generic 2 does not check the Tenant Services permissions. Primarily for this reason, a partitioned switch should not:

- Serve as a tandem node in an ETN or DCS network.
- Serve as the main a Main / Satellite / Tributary arrangement.

- Serve as a branch location in a CAS (Centralized Attendant Service) arrangement.
- Route data calls from incoming trunk groups to host computers over outgoing trunk groups.

However, System 85 and DEFINITY Generic 2 can check Tenant Services permissions for trunk-to-trunk connections involving Remote Access trunks. These checks are applied when an authorization code is required to access the switch (see the Remote Access interaction).

Extension Number Steering

Extension number steering is partially compatible with the Tenant Services feature. When a local voice terminal dials an extension number that steers to a trunk-group dial access code, the partitioning checks are made after the software converts the extension number to the dial access code. (This operation allows voice terminal users at a satellite location to dial extension numbers that steer to the main location. Or, internal users can dial extension numbers that steer to data trunk groups.)

However, calls arriving over incoming trunk groups are treated differently. When an outside caller dials a System 85 or DEFINITY Generic 2 extension number that steers to an outgoing trunk-group dial access code, the partitioning checks are not performed Therefore, Extension Number Steering does not provide a limited form of trunk-to-trunk partitioning.

Originating Registers

Originating registers for outgoing calls are not partitioned. These registers are a system resource, and are provided to users on an "as needed" basis. As a result, members of a partition can experience dial tone delays during periods when another partition is heavily engaged in placing outgoing calls.

Dial Access Codes

The feature dial access codes for a partitioned System 85 or DEFINITY Generic 2 are assigned on a system-wide basis. So, for example, the Automatic Callback or the Leave Word Calling dial access code is dialed using the same digits from every partition in the switch. By default, any voice terminal user can access any administered switch feature (either intentionally or inadvertently) by dialing the correct digits. Therefore, when the system manager desires to limit access to features, appropriate restrictions should be ***positively administered*** whenever possible. If certain features cannot be specifically restricted, the system manager should be careful not to publicize the access codes to those users for whom the features are not intended.

Classes of Service

An R2 V4 System 85 or DEFINITY Generic 2 provides 63 classes of service that are shared by the tenants in a partitioned switch.

Class of Service 31 is applied to every Remote Access user.

Numbering Plan

The numbering plan for System 85 and DEFINITY Generic 2 is assigned on a switch basis. That is, 3-, 4-, and 5-digit extensions cannot coexist on the same switch. The extension numbers for every extension partition must contain the same number of digits. (The Flexible Numbering feature is not available on System 85 or DEFINITY Generic 2.)

Multiple Extensions on Multiappearance Voice Terminals

Typically, every extension number on a multiappearance voice terminal should belong to the same extension partition. However, there are no tests in the administration software to prevent assigning extensions from different partitions to the same multiappearance terminal. So, if this capability proves useful, the administration can be performed. However, it is the responsibility of the system manager to ensure that extensions in a partitioned switch are assigned properly.

Separate Directories for Each Partition

The manager of a partitioned switch should provide a separate telephone directory for each partition in the partitioned switch. The directory for each partition should contain only the extension numbers, attendant numbers, and dial access codes that users in that partition are allowed to dial. Providing separate directories for each partition would serve to minimize confusion for users, and would also increase the users' perceptions of being served by their "own" switch.

Voice-Terminal Display Area and Training Facility

The manager of a partitioned switch can provide a facility where the variety of voice terminals are displayed and demonstrated. Using such a facility, prospective tenants can see this equipment first-hand, acquire a better concept of the available choices, and actually learn to use the equipment.

Two Partitions for the System Manager's Organization

For many applications of shared telecommunications, the system manager and colleagues are employees of a linger organization. In many instances, it would be inappropriate to place the entire organization in Partition(s) 0. Usually, the rest of the organization can be assigned to Partition(s) 1. In this way, there is still communication between the system-management group and larger organization, but the rest of the organization is ***fictionally separated*** from other tenant organizations. This separation provides the expected privacy, perhaps "autonomy," (e.g., from service observing, agent override, Override, and Busy Verification of Lines) for the individual users in the rest of the organization.

Recent-Disconnect Announcements and Call Vectoring

Multiple "recent-disconnect" announcements can be desirable for a partitioned switch (for example, "You have reached an unworking number of the _ _ _ _ _ _ Corporation."). When Call Vectoring is used on a partitioned switch, as many as 84 different recent-disconnect announcements can be provided.

Whenever a voice terminal is taken out of service, the voice terminal can be removed using Procedure 000, Word 1. Once the voice terminal is removed, the extension number is temporarily assigned (in Procedure 000, Word 1) as a VDN (Vector Directory Number) that points to a specific partition's "recent-disconnect" vector (Procedure 031, Word1). Each recent-disconnect vector would contain a single "forced disconnect with announcement" step that ***specifies*** the actual tenant called, and provides that tenant's LDN (for example, "You have reached an non-working number of the Jericho Company. For assistance, please call 737-2100.")

Call Coverage to the Shared Attendant Queue

Call Vectoring can also redirect coverage calls to the attendant queue. Attendant coverage can be beneficial for some System 85s or DEFINITY Generic 2s. Using this coverage, the attendant group can serve as the final coverage point for an assortment of principals.

A VDN can be assigned as the final point in a coverage path. One of these VDNs can be assigned to a vector with a single "route to" step. The "route to" step within this coverage vector contains an Abbreviated Dialing list item that outpulses the attendant dial access code [usually "0," or a DID LDN (Listed Directory Number)].

Since "route to" steps can direct calls to DID LDNs, partitioned switches can also cover to the ***shared*** attendant queue. Each extension partition desiring attendant coverage can have a vector that directs calls to the LDN for the attendant partition assigned to that extension partition. In this way, attendant coverage is a partitioned function of the Tenant services environment.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

Abbreviated Dialing list items are not checked for legality at the time they are entered. Illegal list items are accepted into the list, but calls to these illegal numbers are denied when the digits are outpulsed by the switch. When a user attempts to use Abbreviated Dialing to place a call over another partition's trunk group or to an extension in another partition, the switch returns intercept treatment to the calling party.

A voice terminal user from any partition can be allowed to access the system list, but the switch will only allow completion of calls that are consistent with the dialing capability of that user's partition.

Each partition can have a group list that every voice terminal user in the partition can access. This would provide the equivalent of the system list on a per-partition basis. Again, the switch will deny calls using list items that are not consistent with a partition's dialing capabilities.

Attendant Call Waiting

System 85 and DEFINITY Generic 2 allow attendant calls to wait on local voice terminals, and always provide 2-burst tone for the called voice terminal. When allowed, 2-burst tone is provided for attendant calls directed to a voice terminal using the extension number. When allowed, 2-burst tone is also provided for attendant calls directed to a voice terminal using DXS (Direct Extension Selection).

Attendant Direct Extension Selection With Busy Lamp Field

Assignment of DXS (Direct Extension Selection) or Extension DXS is on a system-wide basis. Every attendant in a partitioned System 85 or DEFINITY Generic 2 is provided with the complete set of DCS buttons and the associated BLF (Busy Lamp Field) lamps. The BLF lamps are updated on every attendant console, but attendant calling is limited by the Tenant Services feature.

An attendant (in a partition other than Attendant Partition 0) is allowed to use DXS or Extended DXS to call a voice terminal in any partition that the attendant can call using the touch-tone dialing pad. When an attendant attempts to call a voice terminal in a partition that is not allowed by partitioning, the switch returns intrcept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to use DXS or Extended DXS to call any voice terminal in the switch.

Attendant Direct Trunk Group Selection

Assignment of Attendant Direct Trunk Group selection is on a system-wide basis. Every attendant console in a partitioned System 85 or DEFINITY Generic 2 is provided with the complete set of 24 DTGS buttons and the associated lamps. However, the Tenant Services feature limits the use of these buttons. Also, the switch only updates the lamps associated with a specific button for consoles that can access that button's trunk group.

An attendant (in a partition other than Attendant Partition 0) is allowed to use the DTGS feature to access any trunk group that the attendant can access by dialing the trunk-group access code. When an attendant tries to select a trunk group that is not allowed by partitioning, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to use every DTGS button on the attendant console. Also, the switch updates the lamps associated with every DTGS button on these consoles.

Attendant Control of Trunk Group Access

An attendant (in a partition other than Attendant Partition 0) can control access to trunk groups that are assigned to that attendant's partition. When control is allowed, the switch returns confirmation tone to the attendant (as would occur in an unpartitioned switch). However, when an attendant attempts to control a trunk group that is not assigned to the attendant's partition, the switch denies the attempt and returns intercept treatment to the attendant.

When Attendant Control of Trunk Group Access is activated toward a ***shared*** trunk group, the switch applies control to every extension partition sharing the trunk group. Controlled outgoing calls from an extension partition with an attendant partition assigned route to the assigned attendant partition. Controlled outgoing calls from an extension partition ***without*** an attendant partition assigned route to Attendant Partition 0. If Attendant Partition 0 is not assigned, these outgoing calls are denied.

An attendant in Attendant Partition 0 can control access to any trunk group* with an assigned button in the lowest two rows of Attendant DTGS buttons. (These two rows of buttons have "CONT" lamps.)

Attendant Interposition Calling and Transfer

An attendant (in a partition other than Attendant Partition 0) is only allowed to place interposition calls to attendants in the same partition or to attendants in Attendant Partition 0. If an attendant tries to place an interposition call to an attendant in any other partition, the switch denies the call by returning intercept treatment.

An attendant in Attendant Partition 0 is allowed to place an interposition call to an attendant in any partition.

Attendant Recall

For calls ***held*** on an attendant console, the Attendant Recall feature is naturally partitioned on System 85 and DEFINITY Generic 2. In a properly partitioned switch, an attendant cannot receive a call that breaks the rules of partitioning. The attendant must extend the call acceding to partitioning rules. So, for a call held on a console, both parties in the held call can access the attendant holding the call. And, the Attendant Recall feature always alerts the attendant who is holding the call.

For calls ***not held*** on an attendant console, the Attendant Recall feature is also partitioned. For these recalls, one of the talking parties places the other on hold and then dials the attendant group. In a properly partitioned switch, both of these parties can access the same attendant partition. So, when either party dials the attendant queue (using either the attendant access code or an LDN) the same partitioning checks used by the Dial Access to Attendant feature are made.

Attendant Release Loop Operation

The attendant-calling operations of this feature are naturally partitioned on System 85 and DEFINITY Generic 2. In a properly partitioned switch, an attendant cannot receive a call that breaks the rules of partitioning. The attendant must extend the call according to partitioning rules. And, the Attendant Release Loop Operation feature always returns the call to the same attendant partition of the console that extended the call.

* Except the Attendant Conference trunk group (if assigned).

The Release Loop Time Change operation is not partitioned. The ARL timed reminder interval can be changed by an attendant in any attendant partition. When this is done, the new timed reminder interval is applied to every attendant partition in the switch

AUDIX (Audio Information Exchange)

A partitioned System 85 or DEFINITY Generic 2 is not aware of AUDIX user permissions. When a voice terminal user dials the AUDIX extension number, the switch follows the usual rules for terminal-to-terminal calling to reach the AUDIX system. Therefore the AUDIX extension number must either belong to the user's extension partition or to Extension Partition 0.

After reaching the AUDIX system, messages can be left for, created by, or retrieved by any subscriber (regardless of the extension partition the subscriber belongs to).

The Call Transfer To (and From) AUDIX functions are partitioned. A voice terminal user (in an extension partition other than Extension Partition 0) can transfer these calls to extension numbers in the same extension partition or to extensions in Extension Partition 0. When the user tries to transfer these calls to any other extension partition (using an extension number), the switch returns intercept treatment to the transferring party.

A voice terminal user in Extension Partition 0 is allowed to transfer these calls (using an extension number) to any voice terminal in the switch.

Authorization Codes

In Procedure 281, Word 1, the amount of digits used in authorization codes is assigned on a system-wide basis. Therefore, every partition in a partitioned switch must use the same amount of digits.

In Procedure 282, Word 1, each authorization code is assigned to an extension partition. Users in different extension partitions are not allowed to share an authorization code, and the same code cannot be duplicated for use in more than one partition.

When a voice terminal user in a specific partition dials an authorization code, call-processing software checks to determine whether the code's associated partition matches the partition of the voice terminal used to place the call. If these partitions do not match, the switch returns intercept treatment to the calling party.

AAR (Automatic Alternate Routing)

The AAR feature provides private network routing (including tandem services) on System 85 and DEFINITY Generic 2.1 switches. The AAR feature is not partitioned. So, a partitioned System 85 or Generic 2.1 ***cannot save as a tandem node*** for any tenant's private network. However, with proper assignment, a partitioned switch ***can serve as an endpoint*** for one or more tenants' private networks.

To allow an extension partition to serve as an endpoint in a private network, the partition should be administered as a satellite in a Main / Satellite configuration (where the main also serves as a tandem node in the private network). The tie trunks connecting the main and the "satellite partition" must be ***dedicated to*** the partition to prevent unexpected access by other partitions. Under this arrangement, outgoing calls route from the satellite partition to the rest of the private network according to the partitioning assignments. Incoming calls route from the main to the satellite partition according to Extension Number Steering.

ACD (Automatic Call Distribution)

Automatic-in type routing to ACD splits is not partitioned. There are no checks in Procedure 115 to ensure that the partition of an automatic-in type trunk group matches the partition of the assigned split. It is the responsibility of the system manager to ensure that these partition numbers match.

Dial-repeating type routing to ACD splits is partitioned. When the digits of an associated extension number are either passed to the switch from the serving switch or dialed from inside the switch, the Tenant Services feature makes the necessary partitioning checks. Associated extension numbers are assigned to partitions using Procedure 000, Word 4.

There are no checks made in switch administration to ensure the internal partitioning of ACD splits. When administered from the Manager II, MAAP or SMT, a single split may contain agents from several extension partitions. Once an ACD call enters the split's queue, the call will terminate to the selected available agent regardless of the agent's extension partition assignment

In general practice, each ACD split provides a functional division of call answering responsibilities. For most (if not all) ACD applications, it is recommended that each split contain only agents assigned to the same extension partition. The System Manager should be careful to follow this guideline.

The ACD feature also allows a split supervisor (in a partition other than Extension Partition 0) to add and remove agents to / from the supervisor's ACD split. When this method for adding agents is used, the Tenant Services feature provides a check to ensure that the added agent belongs either to the supervises extension partition or to Extension Partition 0. If not, the switch returns intercept treatment to the split supervisor.

A split supervisor in Extension Partition 0 is allowed to add an agent from any partition to any ACD split in the switch.

An observer (in a partition other than Extension Partition 0) is allowed to use agent override to monitor the call handling activity of an agent in the same extension partition or in Extension Partition 0. When the observer attempts to activate agent override toward an agent in any other partition, the switch returns intercept treatment to the observer.

An observer in Extension Partition 0 is allowed to use agent override to observe any agent in the switch.

An observer (in a partition other than Extension Partition 0) is allowed to use service observing to monitor the call handling activity of an agent in the same extension partition or in Extension Partition 0. When the observer attempts to activate service observing toward an agent in any other partition, the switch returns intercept treatment to the observer.

An observer in Extension Partition 0 is allowed to use service observing to observe any agent in the switch.

Automatic Callback

A voice terminal user (in a partition other than Extension Partition 0) is allowed to activate Automatic Callback toward voice terminals in the same partition or in Extension Partition 0. When the user tries to activate Automatic Callback toward a voice terminal in any other partition, the switch returns intercept treatment.

A voice terminal user in Extension Partition 0 is allowed to activate Automatic Callback toward any voice terminal in the switch.

Automatic Circuit Assurance

An attendant in Attendant Partition 0 is allowed to activate or deactivate the Automatic Circuit Assurance feature. However, the switch will only direct ACA referral calls to the designated attendant console. The designated console for ACA referral calls must belong to Attendant Partition 0.

Attendants (in partitions other than Attendant Partition 0) are not allowed to activate ACA, deactivate ACA, or receive ACA referral calls.

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, Automatic Route Selection is a partitioned feature. In a partitioned switch, one or more partitions are assigned to one of 64 call categories in Procedure 320, Words 2 and 3. (Each extension partition or attendant partition can only be assigned to one call category.) In turn, each unique routing designator/call category pair maps to one of 64 ARS patterns in Procedure 314, Word 1. The result is that an ARS call can receive treatment by a different pattern depending on the partition's assigned call category. Further, once an ARS call enters the assigned pattern, the ARS call-processing software checks successive trunk groups in the pattern (in a similar manner to the algorithm described in the Route Advance interaction) to determine whether the call is allowed to use the trunk group.

The three ARS time-of-day plans are a system-wide resource, and are not partitioned. In an unpartitioned System 85 or DEFINITY Generic 2.1, any attendant is allowed to change the time-of-day plan using Manual Override. For a partitioned System 85 or Generic 2.1, any attendant can also change the time-of-day plan.

NOTE: For partitioned System 85 or DEFINITY Generic 2.1, it is recommended that the dial access code (Encode 60) be used to change the ARS routing plan, and that only the attendant(s) in Attendant Partition 0 are informed of the dial access code. This will provide better control. Otherwise, the routing plan might be changed at undesirable times.

Bridged Call

There are no tests in Procedure 052, Word 1 to ensure that the images of an appearance (or that the appearances of an extension) do not cross partition boundaries. It is the responsibility of the system manager to ensure that every image of each extension resides only on terminals that are used by a particular extension partition.

Busy Verification of Lines

An attendant (in a partition other than Attendant Partition 0) is allowed to verify any extension residing in Extension Partition 0 or residing in an extension partition that has been assigned to the attendant's partition. When the attendant tries to verify an extension in another partition, the switch returns intercept treatment to the attendant. An attendant in Attendant Partition 0 is allowed to version any extension in the switch.

Call Coverage

There are no tests in Procedure 011, Word 1 to ensure that every point in a coverage path belongs to the same extension paxtition. It is the responsibility of the system manager to ensure that coverage paths do not cross partition boundaries.

There are also no tests in Procedure 000, Word 2 to ensure that a coverage group is only assigned to extensions residing in the same extension partition. It is the responsibility of the system manager to ensure that a coverage group pertaining to one extension partition is not assigned to an extension in another extension partition.

Call Detail Recording

The Call Detail Recording features generally allow an attendant to activate or deactivate recording for trunk groups. However, in a partitioned System 85 or DEFINITY Generic 2, this operation is controlled. If an attendant (in a partition other than Attendant. Partition 0) tries to activate or deactivate recording for a trunk group which that attendant cannot access, the switch returns intercept treatment. An attendant in Attendant Partition 0 can activate or deactivate recording for any trunk group in the switch.

A data item is provided to indicate the extension partition of the extension originating or terminating a call. Whenever the extension number changes in the CDR record, the field is updated to reflect the extension partition of the new extension number. For an unpartitioned switch, this field contains an "A" (hexadecimal).

Call Forwarding—Busy and Don't Answer

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition.

Call Forwarding—Don't Answer

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition.

Call Forwarding—Follow Me

A voice terminal user (in a partition other than Extension Partition 0) is allowed to forward calls to another voice terminal in the same partition or in Extension Partition 0. The switch will return intercept treatment to a user attempting to forward calls to any other partition.

Call Forwarding—Off Net is allowed in a partitioned System 85 or DEFINITY Generic 2. This forwarding can be activated using a networking access code or using the access code of a trunk group that is dedicated to the user's extension partition.

Call Park

The call park zones for the Call Park feature are not partitioned. By default, the provided zones are equally accessible to voice terminal users in any extension partition.

Voice terminal access to the Call Park feature can be limited in the voice terminal class of service. To limit voice terminal access, assign a Miscellaneous Trunk Restrictions group containing the Call Park trunk group to a voice terminal class of service in Procedure 010, Word 3.

Answer-back channels for the Call Park feature are not partitioned. A parked call can be retrieved by dialing the answer-back access code from any voice terminal in the switch.

Call Pickup

There are no tests in Procedure 000, Word 2 to ensure that a Call Pickup group is only assigned to extensions residing in the same extension partition. The system manager should ensure that every member of each Call Pickup group belongs to the same extension partition.

When Call Pickup groups have been assigned to overlap partition boundaries, the call-processing software provides partitioning for the feature. If a Call Pickup group member in one partition tries to pickup a call to another group member residing in a different extension partition, intercept treatment is returned by the switch.

Call Vectoring

Trunk group-oriented routing to vectors is not partitioned. There are no checks in Procedure 031, Word 2 to ensure that the partition of an automatic-in trunk group matches the partition of the assigned VDN. It is the responsibility of the system manager to ensure that these partition numbers match.

Digit-oriented routing to vectors is partitioned. When a VDN's digits are passed to the System 85 or DEFINITY Generic 2 from the serving switch or dialed from inside the switch, the Tenant Services feature makes the necessary partitioning checks. Vector directory numbers are assigned to partitions using Procedure 000, Word 4.

The "queue to main split" and "check backup split" vector commands are not partitioned. For vectors containing either of these commands, there are no checks in call-processing software to ensure that the extension partition of the answering agent matches the extension partition of the VDN or the split supervisor. It is the responsibility of the System Manager to ensure that "queue to main split" and "check backup split" commands do not cause calls to cross partition boundaries.

The "route to" vector command can route calls to an extension in the VDN's extension partition or to Extension Partition 0. If a "route to" command is programmed to route calls to another extension partition, the switch will treat a final effective "route to" step as a "stop" step. Otherwise, the "route to" step is ignored, and vector processing continues with the next sequential step.

If a vector contains a "route to" step that routes calls to the shared attendant queue, the call will terminate to an attendant partition that is assigned to the VDN's extension partition. If no attendant partition is assigned to the VDN's extension partition, the switch returns Intercept Treatment to the calling party.

If a vector contains a "route to" step that routes calls outside the switch, the call uses the FRL assigned to the VDN and the Call Category assigned to the VDN's extension partition.

In Procedure 030, Word 1, one Abbreviated Dialing group list is assigned to control the destinations of "route to" commands. The controlling terminal for this group list (assigned in Procedure 059, Word 1) can belong to any extension partition. However, it is strongly recommended that this controlling terminal belong to Extension Partition 0.

The calling party announcements for the Call Vectoring feature are not partitioned. These announcements are available to every extension partition. VDNs in various extension partitions can terminate to several vectors containing "announcement" steps that all request the same recorded announcement.

Call Waiting

A partitioned System 85 or DEFINITY Generic 2 allows Call Waiting and provides 1-burst tone for terminal-to-terminal calls inside an extension partition. The switch also allows Call Waiting and provides 1-burst tone for terminal-to-terminal calls between an extension partition and Extension Partition 0.

Otherwise, terminal-to-terminal calls are not allowed to cross partition boundaries. In order to call a voice terminal in any other partition, a voice terminal user must dial the appropriate 7-digit number which routes the call over a CO (Central Office) trunk. When this is done, the switch allows Call Waiting and provides 2-burst tone for the called voice terminal.

The switch allows Call Waiting for incoming calls from the public network, and provides 2-burst tone for these calls.

System 85 and DEFINITY Generic 2 allow attendant calls to wait on local voice terminals, and always provide 2-burst tone for the called voice terminal. When allowed, 2-burst tone is provided for attendant calls directed to a voice terminal using the extension number. When allowed, 2-burst tone is provided for attendant calls directed to a voice terminal using DXS (Direct Extension Selection). Two-burst tone is also provided for attendant calls directed to a local voice terminal using the appropriate 7-digit public-network number (when partitioning requires this method of dialing).

CAS (Centralized Attendant Service)

A partitioned System 85 or DEFINITY Generic 2 can function as a main location in a CAS arrangement. However, incoming RLTs (release link trunks) are not assigned to specific partitions, so these centralized attendants must reside in Attendant Partition 0.

A partitioned System 85 or DEFINITY Generic 2 cannot serve as a branch location in a CAS arrangement.

Code Calling Access—Universal

The paging (chiming) zones for the Code Calling Access feature are not partitioned. Attendants in any attendant partition and voice terminal users in any extension partition have equal access to each paging zone.

Conference—Attendant Five Party

The Conference—Attendant Five Party feature can be accessed by an attendant in any attendant partition.

After a voice terminal or trunk user (the first conferee) calls an attendant (in a partition other than Attendant Partition 0) to establish a conference, the attendant can add conferees in the same partition by dialing the conferee's extension number. Voice terminal users in other Extension Partitions can also be added, but the attendant must dial the 7-digit private network number. Furthermore, transmission quality may deteriorate if more than two CO trunks or more than one CO trunk and two tie trunks are in the connection.

An extension in Extension Partition 0 can be included in any attendant conference.

An attendant in Attendant Partition 0 is allowed to add conferees from any partition in the switch or from any trunk group to an attendant conference.

An attendant recall by a conferee is usually directed to the attendant partition associated with that conferee's extension partition. However, when the establishing attendant has placed the conference on hold, an attendant recall by any of the conferees is directed to that attendant.

Conference—Attendant Six Party

The SN254 attendant conference circuits (as many as 13) are not partitioned. These trunks, when provided, are equally accessible to an attendant in any attendant partition needing them.

After a voice terminal or trunk user (the first conferee) calls an attendant (in a partition other than Attendant Partition 0) to establish a conference, the attendant is only allowed to add conferees that would otherwise have access to the first conferee who is not a member of Extension Partition 0.

An extension in Extension Partition 0 can be included in any attendant conference

An attendant in Attendant Partition 0 is allowed to add conferees from any partition in the switch or from over any trunk group to an attendant conference.

An attendant recall by a conferee is usually directed to the attendant partition associated with that conferee's extension partition. However, when the establishing attendant has placed the conference on hold, an attendant recall by any of the conferees is directed to that attendant.

Conference—Three Party

A voice terminal user (in a partition other than Extension Partition 0) is allowed to establish 3-party conferences that include participants that the user is otherwise allowed to call. If the user tries to add a conferee to the conference by dialing an extension number in another partition, the switch returns intercept treatment to the user.

A voice terminal user in Extension Partition 0 is allowed to establish 3-party conferences with any voice terminal or over any trunk in the switch.

Data Call Setup

Keyboard dialing is a partitioned function of the Data Call Setup feature. A data terminal user (in a partition other than Extension Partition 0) can use keyboard dialing to access data trunk groups that are dedicated to or shared with the partition. If the data terminal user tries to access any other data trunk group, the switch will return Intercept Treatment to the calling party.

A data terminal user in Extension Partition 0 can use keyboard dialing to access any data trunk group in the switch.

The ***mnemonic dialing*** list, assigned in Procedure 013 Words 1 and 2, is a system-wide resource. These 1000 names are shared by every partition in a partitioned switch.

Mnemonic dialing is a partitioned function of the Data Call Setup feature. A data terminal user (in a partition other than Extension Partition 0) can use mnemonic dialing to access data trunk groups that are dedicated to or shared with the partition. If the data terminal user tries to access any other data trunk group, the switch will return Intercept

Treatment to the calling party. (This Intercept Treatment is returned after a disallowed name has been converted to digits that are outpulsed by the switch.)

A data terminal user in Extension Partition 0 can use mnemonic dialing to access any data trunk group in the switch.

Default dialing numbers are not checked for legality at the time they are assigned, but calls to these disallowed numbers are denied when the digits are outpulsed by the switch. When a data terminal user attempts to use default dialing to place a call over another partition's trunk group, the switch returns intercept treatment to the calling party.

A data terminal user in Extension Partition 0 can use default dialing to access any data trunk group in the switch.

DCA (Data Communications Access)

Data Communications Access (analog access to a host computer) is a partitioned feature on System 85 and DEFINITY Generic 2.

Line-side computer access is partitioned using partitioned extension numbers. Each extension number is assigned to a modem and is usually included in a hunt group. In turn, the modem's extension number is assigned to an extension partition allowing data-terminal access for users in that partition and Extension Partition 0.

Trunk-side computer access is partitioned using partitioned trunk groups. Since the DCA trunk type (37) can be partitioned, access to these trunk groups can be dedicated to or shared by an extension partition. However, if a computer is accessed from outside the switch, ***trunk-side*** partitioning would have no effect. There are no partitioning checks between the incoming trunk group and the outgoing DCA trunk group.

Data Protection

The Data Protection feature is a switch-wide resource that can be shared by data-terminal users in every partition. Data Protection—Permanent is assigned to a class of service, and the 63 classes of service are shared by the various extension partitions. Meanwhile, Data Protection—Temporary is activated with a dial access code, and dial access codes are common to the various extension partitions.

Dedicated Switch Connections

There are no tests in Procedure 360, Word 1 to ensure that the endpoints of a dedicated switch connection belong to compatible extension partitions. It is the responsibility of the system manager to ensure that these connections are allowed.

Dial Access to Attendant

In a partitioned switch, the Dial Access to Attendant feature is limited. Each extension partition can be assigned to one attendant partition. (More than one extension partition can be assigned to the same attendant partition.) These associations between extension partitions and attendant partitions are used to determine whether the Dial Access to Attendant feature is allowed, and the system manager is required to establish these associations.

When a voice terminal user dials the "general" attendant access code (usually "0"), the call can only complete to an attendant in the associated attendant partition. If there is no attendant partition associated with the user's extension partition, the switch denies the call and returns intercept treatment to the calling party.

Local voice terminal users can dial an LDN to reach the attendant queue. When this is done, partitioning checks are made. When a voice terminal user (in a partition other than Extension Partition 0) dials an LDN, the call is allowed if the user's extension partition is assigned to the called attendant partition, or if the user is calling Attendant Partition 0. If not, the switch denies the call and returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can use an LDN to call any attendant partition in the switch.

When a voice terminal user (in a partition other than Extension Partition 0) dials a selected attendant access code, the call is allowed if the dialed attendant resides in the associated attendant partition or in Attendant Partition 0. If not, the switch denies the call and returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 is allowed to call a selected attendant in any partition.

DMI (Digital Multiplexed Interface)

DMI (DS1 access to a host computer) is a partitioned feature on System 85 and DEFINITY Generic 2. Since the trunk types (108 and 109) for DMI trunk groups can be partitioned, access to these trunk groups can be dedicated to or shared by an extension partition.

DS1 Interface

The DS1 Interface is a partitioned feature on System 85 and DEFINITY Generic 2. If the desired trunk type applies to DS1 service and the specific trunk group is partitioned, access to this trunk group can be dedicated to or shared by an extension partition.

Direct Inward Dialing

A DID (Direct Inward Dialing) trunk group can be dedicated to a specific extension partition or attendant partition. When this is done, the system manager should accurately convey the numbering plan (within the partition that is associated with the trunk group) to the serving CO (Central Office) so that calls from the public network can be routed properly.

> **NOTE:** Using this method, if any DID calls are improperly routed to the wrong partition, the switch will return intercept treatment (reorder tone by default) to the calling party.

To minimize the consumption of DID trunk groups in a partitioned System 85 or DEFINITY Generic 2, the partitioning of DID trunks can also be set up at the serving CO. Under this arrangement, numerous small DID trunk groups at the CO can be administered ***to converge to*** a single large DID trunk group (containing as many as 255 discrete trunks) at the partitioned switch. This large DID trunk group is assigned to Partition 0, and is shared by various partitions.When this is done, the system manager should coordinate the arrangement with the serving CO so that calls from the public network are properly partitioned.

NOTE: With this type of trunking configuration, the System 85 or DEFINITY Generic 2 has no knowledge of the partitioning arrangement at the serving CO. So, if any DID calls are improperly routed out of the serving CO, the switch would usually allow the call.

Direct Outward Dialing

A voice terminal user in a partitioned switch is allowed to place calls outside the switch using a DOD trunk group that is either dedicated to or shared by the user's extension partition. If the user's extension partition is not allowed to use the dialed trunk group, and if the switch cannot select an allowable alternate trunk group from the Route Advance list (if provided), the switch returns intercept treatment to the calling party.

DCS (Distributed Communications System)

A partitioned System 85 or DEFINITY Generic 2 can serve as an endpoint in a DCS cluster. As long as the partitioned switch is serving as an endpoint, an extension partition and an optional attendant partition within the partitioned switch can access the tandem node over dedicated trunk groups. However, a partitioned System 85 or DEFINITY Generic 2 should not serve as a tandem node. When this is done, the tandem does not provide trunk-to-trunk partitioning to the public or private network.

When DCS is used in conjunction with Tenant Services, be careful to ensure that the tenant's organization can access the dedicated DCS trunk group at both ends. Otherwise, DCS feature transparency is degraded. As an example, when a voice terminal user at Node A activates Call Forwarding toward a voice terminal in Node B, the activation is accepted by Node A with confirmation tone. However, if both ends of the tie trunk cannot be accessed by the same tenant organization, a call to the user at Node A will fail during the forwarding process. Meanwhile, the calling party receives the usual forwarding display and the forwarding party receives ring-ping.

Allowing a ***single*** partitioned System 85 or DEFINITY Generic 2 to serve as an endpoint in ***multiple*** DCS clusters can be difficult to implement. Within the partitioned switch, each extension partition must have unique extension numbers. Moreover, the endpoint partition's extension numbers must be unique in its DCS cluster. Also, the DCS tie trunks must have the same trunk-group numbers and trunk numbers at both ends.

Extension Number Portability

On a System 85 or DEFINITY Generic 2.1 switch, Extension Number Portability is allowed as long as the ported number is consistent with the numbering plan in the partitioned switch. For example, the ported extension number is not allowed to duplicate an extension number in another partition. Also, the extension numbers in every extension partition of the partitioned switch and in the associated portability subnetwork must contain the same number of digits. This is because AAR is not a partitioned feature. In an AAR arrangement, an extension partition cannot serve as a tandem node. This same restriction applies to an extension partition in a portability subnetwork. That is, an extension partition on a partitioned System 85 or DEFINITY Generic 2.1 cannot serve as a tandem node within the network.

On a DEFINITY Generic 2.2 switch, AAR is replaced by the WCR feature. With WCR, the Extension Number Portability feature works in essentially the same way as on Generic 2.1 switches, except that the WCR feature is partitionable and therefore some of the constraints imposed by AAR are lifted. Partition checking is not done on incoming to outgoing trunk calls.

HCA (Host Computer Access)

Host Computer Access (DCP access to a host computer) is a partitioned feature on System 85 and DEFINITY Generic 2.

Line-side computer access is partitioned using partitioned extension numbers. Each extension number is assigned to a data module and is usually included in a hunt group. In turn, the data module's extension number is assigned to an extension partition allowing data-terminal access for users in that partition and Extension Partition 0.

Trunk-side computer access is partitioned using partitioned trunk groups. Since the trunk types (103 and 104) can be partitioned, access to these trunk groups can be dedicated to or shared by an extension partition. However, if a computer is accessed from outside the switch, ***trunk-side*** partitioning would have no effect. There are no partitioning checks between the incoming trunk group and the outgoing HCA trunk group.

Hot Line

Hot Line numbers are not checked for legality at the time they are entered. Illegal Hot Line numbers are denied when the digits are outpulsed by the switch. When a user attempts to place a Hot Line call over another partition's trunk group or to an extension in another partition, the switch returns intercept treatment to the calling party.

Hunting

There are no tests in System 85 or DEFINITY Generic 2 software to prevent hunting to another partition. It is the responsibility of the system manager to ensure that the extensions in a hunt group do not cross partition boundaries.

Intercept Treatment

The Intercept Treatment feature is administered on a per-system basis. If the Intercept Treatment feature is programmed in Procedure 289, Word 1, partitioning denials for internal calls always cause the switch to return the type of intercept administered for internal calls to vacant access codes (usually, intercept tone). If the Intercept Treatment feature is not programmed in Procedure 289, Word 1, the switch always returns the default types of intercept treatment: intercept tone for internal calls and reorder tone for calls from the public network.

Intercept Treatment—Attendant

In a partitioned System 85 or DEFINITY Generic 2, this function of the Intercept Treatment feature routes intercepted calls to an associated attendant partition. Incoming trunk calls placed to ***invalid extension numbers*** are routed to an attendant partition assigned to the trunk group. (If an attendant partition has not been assigned to the trunk group in Procedure 270, Word 5, the intercepted call routes to Attendant Partition 0.) Incoming trunk calls placed to ***valid extension numbers*** that are restricted from receiving the calls (for example, by DID Restriction, Inward Restriction, or Terminal Restriction) are routed to an attendant partition assigned to the called extensions' partitions.

Intercept Treatment—Recorded Announcement

The recorded announcement messages for the Intercept Treatment feature are system-wide messages. There is no capability to assign messages with different content for different partitions.

Intercom—Automatic

There are no tests in Procedure 056, Word 1 to ensure that the two Automatic Intercom members of a nondial intercom group belong to the same extension partition. It is the responsibility of the system manager to ensure that intercom groups do not cross partition boundaries.

Intercom—Dial

There are no tests in Procedure 056, Word 1 to ensure that the members of a Dial Intercom group belong to the same extension partition. It is the responsibility of the system manager to ensure that intercom groups do not cross partition boundaries.

Intercom—Manual

There are no tests in Procedure 056, Word 1 to ensure that the Manual Intercom members of a nondia1 intercom group belong to the same extension partition. It is the responsibility of the system manager to ensure that intercom groups do not cross partition boundaries.

ISN (Information Systems Network) Interface

A partitioned System 85 or DEFINITY Generic 2 can serve as an endpoint in a tenant's data network. As long as the partitioned switch is serving as an endpoint to the ISN, an extension partition within the partitioned switch can access the ISN over dedicated trunk groups. However, a partitioned System 85 or DEFINITY Generic 2 should not reside between the ISN and another switch. Acting as a tandem, the partitioned switch does not provide trunk-to-trunk partitioning to the ISN.

Last Number Dialed

A voice terminal user (in an extension other than Extension Partition 0) can use the Last Number Dialed button to redial calls within the user's partition, to Extension Partition 0, to the associated attendant partition, to Attendant Partition 0, or over an associated trunk group. When the user tries to use this button to redial an illegal call, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can use the Last Number Dialed button to redial calls to any extension partition, to any attendant partition, or over any trunk group in the switch.

Leave Word Calling

Leave Word Calling messages can only be left for an extension in the same extension partition or for an extension in Extension Partition 0. When Leave Word Calling is blocked by partitioning, the switch returns intercept treatment to the calling party.

Demand Print

The Demand Print function of the Leave Word Calling feature is also partitioned as part of the Tenant Services feature.

When a demand printout is requested from a voice terminal (in a partition other than Extension Partition 0), System 85 or DEFINITY Generic 2 software checks to determine whether the extension used to retrieve the messages and the extension for which the messages were left are in the same extension partition. If not, the switch returns intercept treatment to the party requesting the demand printout.

From a voice terminal in Extension Partition 0, a demand printout can be requested for messages left to any extension in the switch.

Given the preceding operation, there are two approaches to implementing the Demand Print function on a partitioned switch. One approach is to provide a voice terminal and an associated printer for each extension partition. The other approach is to provide a voice terminal and an associated printer to Extension Partition 0, and share this facility with all of the extension partitions.

> **NOTE:** Access to a printer cannot be shared by several tenants and denied to the other tenants.

Look-Ahead Interflow

The Look-Ahead Interflow feature can be provided on a partitioned R2 V4 System 85 or DEFINITY Generic 2. When implemented, the Look-Ahead Interflow feature (and the supporting Call Vectoring feature) are provided for every extension partition in the switch.

At a sending (or tandeming) partitioned R2 V4 System 85 or DEFINITY Generic 2.1, the ARS partitioning provided by the Tenant Services feature [the ability to associate a routing designator (from 1 to 64) with an ARS pattern identified by a different number] can be applied to Look-Ahead Interflow calls as they are diverted outside the R2 V4 System 85 DEFINITY Generic 2 to the public network. For a Generic 2.2 switch, call category is determined by partition, time-of-day plan, and conditional routing count. Call category is then used with VNI to determine the routing pattern to be used.

At a receiving partitioned R2 V4 System 85 or DEFINITY Generic 2, the switch also observes partition boundaries when the ISDN—PRI routing uses a form of "digit-oriented" routing (Trunk Type 46, 47, or 120). As the receiving switch receives the digits from within the Look-Ahead D-channel message, the receiving switch makes the necessary partitioning checks (based on the VDN's association with an extension partition in Procedure 000, Word 4).

Loudspeaker Paging Access

The paging zones for the Loudspeaker Paging Access feature are not partitioned. By default, the provided zones are equally accessible to attendants in any attendant partition and to voice terminal users in any extension partition.

Voice terminal access to the Loudspeaker Paging Access feature can be limited or enhanced in the voice terminal class of service. To limit voice terminal access, assign a Miscellaneous Trunk Restrictions group containing the Loudspeaker Paging trunk group to a voice terminal class of service in Procedure 010, Word 3. To enhance voice terminal access, priority paging can be assigned to a voice terminal class of service in Procedure 010, Word 1.

Answer-back channels for the Loudspeaker Paging Access feature are not partitioned. A page can be answered by dialing the answer-back access code from any voice terminal in the switch.

Main/Satellite/Tributary

A partitioned System 85 or DEFINITY Generic 2 can serve as a satellite or a tributary in a Main / Satellite / Tributary arrangement. As long as the partitioned switch is serving as an endpoint in the arrangement, an extension partition and an optional attendant partition within the partitioned switch can access the main over dedicated trunk groups. However, a partitioned System 85 or DEFINITY Generic 2 should not serve as the main. When this is done, the main does not provide trunk-to-trunk partitioning to the public or private network.

Malicious Call Trace

An attendant in any attendant partition or a voice terminal user in any extension partition can activate the Malicious Call Trace feature in response to a malicious call.

When the Malicious Call Trace feature is activated on a partitioned switch, only the attendants in Partition 0 are alerted to the malicious call.

Voice recorder trunks are not partitioned. These trunks, when provided, are equally accessible to any partition needing them.

Manual Signaling

There are no tests in Procedure 053, Word 1 to ensure that the members of a Manual Signaling pair belong to the same extension partition. It is the responsibility of the system manager to ensure that Manual Signaling pairs do not cross partition boundaries.

Modem Pooling

The Modem Pooling feature is not partitioned. When a digital trunk group, an analog trunk group, and a conversion resource are provided, access to this analog-to-digital conversion is provided for every partition in the switch.

Multiple Listed Directory Numbers

DID Listed Directory Numbers

Primarily for use with partitioned switches; 999 DID LDNs are available in Release 2, Version 4. Each LDN is associated with one attendant partition in Procedure 270, Word 4. If this association is not made, calls to unassigned LDNs will route to Attendant Partition 0.

If the need arises to route an LDN call to any attendant in the partitioned switch, then the LDN must be associated with an attendant partition that contains every attendant console.

A local voice terminal user can dial an LDN to reach the attendant queue. When this is done, partitioning checks are made.

Non-DID Listed Directory Numbers

An attendant-completing (automatic) trunk group can also be assigned to terminate to a specific attendant partition. The non-DID LDN is published in the same manner as a DID LDN. The serving CO, upon receiving the non-DID LDN, seizes a trunk in the attendant-completing group and routes the call without sending digits to the System 85 or DEFINITY Generic 2. These trunk groups are assigned as type "attendant-completing" in Procedure 100, Word 1. Then, these trunk groups are associated with one specific attendant partition in Procedure 270, Word 5. [As many of the available trunk groups as desired (up to 982) can be assigned in this manner.]

NOTE: This method of routing LDN calls can provide LDN access to the partitioned switch without using shared trunk groups.

Music-on-Hold Access

The Music-on-Hold Access feature is not partitioned. When a music source is provided and Music-on-Hold is enabled in Procedure 275, Word 1, music is provided to held parties in every partition.

Multiple music sources are also a system-side resource. If an extension (regardless of extension partition) in a certain module places a call on hold, music is added to the held party's time slot by that module's music source (if provided). Otherwise, music is added to the time slot using the music source attached to the ***lowest numbered module*** that does have a music source.

Override

A voice terminal user (in a partition other than Extension Partition 0) is allowed to place Override calls to extensions in the same partition or in Extension Partition 0. When the user ties to place an Override call to an extension in any other partition, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can place an Override call to any other extension in the switch.

Personal CO (Central Office) Line

There are no tests in Procedure 057, Word 1 to ensure that every voice terminal (from 1 to 16) sharing a Personal CO Line belongs to the same extension partition. It is the responsibility of the system manager to ensure that these shared lines do not cross partition boundaries.

A Personal CO Line trunk group can be dedicated to each extension partition that uses the personal lines. (There is a maximum of 150 Personal CO Lines that can be spread among as many as 150 trunk groups.) When this is done, each trunk group can be assigned (in Procedure 270, Word 5) to the specific extension partition to which the trunk group terminates.

To minimize the consumption of Personal CO Line trunk groups in a partitioned System 85 or DEFINITY Generic 2, these discrete personal lines can also be set up ***to converge to*** one (or a few) large trunk groups at the partitioned switch. When this is done, the several trunk groups can be assigned to Extension Partition 0 in Procedure 270, Word 5.

> **NOTE:** This specific trunking configuration is not a "shared trunk group" in the legal sense. The calling activity over Personal CO Line trunk groups would not be averaged. Rather, each CO line in the trunk group is assigned to terminate to a specific voice terminal (or set of voice terminals), and can only be used by the assigned terminal(s). Each trunk facility is dedicated to a partition; the ***trunk-group numbers*** are shared.

Precedence Calling

The Precedence Calling feature cannot be used in a partitioned switch.

Priority Calling

A voice terminal user (in an extension partition other than Extension Partition 0) is allowed to place priority calls within the user's partition or to Extension Partition 0. When the user tries to activate Priority Calling toward an extension in any other partition, the switch returns intercept treatment to the calling party.

A voice terminal user in Extension Partition 0 can place a priority call to any other extension in the switch.

Whenever Priority Calling is allowed, the switch provides 3-burst ringing or 3-burst waiting tone for the called voice terminal.

Privacy—Attendant Lockout

The Attendant Lockout feature is not partitioned. This feature is assigned to the entire switch in Procedure 200, Word 1. If assigned, no attendant in the switch can reenter a call held on the console. If not assigned, every attendant in the switch can reenter a call held on the console.

Queuing

The Queuing feature is not affected by partitioning. If Queuing is administered for an outgoing trunk group and every trunk in the group is busy, then a call (that is otherwise allowed to use the trunk group) can queue on the trunk group. If Queuing is not administered for the trunk group and every trunk in the group is busy, the switch will return reorder tone to the calling party.

Recorded Telephone Dictation Access

The trunk group for the Recorded Telephone Dictation Access feature is not partitioned. By default, the provided trunks are equally accessible to voice terminal users in any extension partition.

Voice terminal access to the Recorded Telephone Dictation Access feature can be limited in the voice terminal class of service. This is done by assigning a Miscellaneous Trunk Restrictions group containing the dictation trunk group to a voice terminal class of service in Procedure 010, Word 3.

Remote Access

Primarily for partitioned switches, the maximum number of Remote Access trunks is increased from 45 to 6000. However, this increased capacity is available using any R2 V4 System 85 or DEFINITY Generic 2. Since the limit of trunks per trunk group remains at 255, at least 24 trunk groups would be required to provide the full capacity of 6000 Remote Access trunks. Also, when the full capacity of these trunks is provided, the amount of available ILNs (internal line numbers) would decrease from 32,703 to 26,703.

Like unpartitioned switches, a partitioned System 85 or DEFINITY Generic 2 provides a choice of two methods for limiting use of the Remote Access feature: barrier codes and authorization codes. Access using authorization codes is partitioned, and is the recommended form of security for Remote Access on partitioned switches. The barrier code, however, is assigned on a system-wide basis, and has the effect of bypassing partitioning.

Barrier Code Required

Remote Access users who access the switch with the barrier code ***are not subject to*** partitioning limitations. These users have the full dialing capabilities of a voice terminal user in Extension Partition 0. As an example, these Remote Access users can access any trunk group in the switch.

If these Remote Access users either "time out" to or dial the attendant queue after accessing the switch, the call is directed to the attendant partition that is associated with the Remote Access trunk group.

Authorization Codes Required

Remote Access users who access the switch with an authorization code ***are subject to*** partitioning limitations. These users have the dialing capabilities of the authorization code's extension partition. As an example, these users are allowed to access any trunk group that is associated with the authorization code's partition.

If these Remote Access users dial an authorization code and then either "time out" to or dial the attendant queue after accessing the switch, the call is directed to the attendant partition associated with the authorization code's extension partition.

Authorization Code Required, But Not Dialed

If a Remote Access user is required to dial an authorization code, but times out to the attendant queue, the call is directed to the attendant partition that is associated with the Remote Access trunk group.

No Barrier Code or Authorization Codes Required

Same as the barrier-code operation.

Restriction—Attendant Control of Voice Terminals

In general practice, the Attendant Control of Voice Terminals feature allows an attendant to activate and cancel temporary restrictions for a specific voice terminal or a predefined group of voice terminals. The Tenant Services feature limits the operation of this feature.

An attendant (in a partition other than Attendant Partition 0) is allowed to activate or cancel restrictions for specific extensions. However, this operation is only allowed when the extension resides in an extension partition that is assigned to the attendant's partition. If the attendant tries to activate or cancel a restriction for an extension in any other partition, the switch returns intercept treatment to the attendant.

An attendant (in a partition other than Attendant Partition 0) is also allowed to activate or cancel restrictions for controlled restriction group. However, this operation is only allowed when the controlled restriction group has been assigned to the attendant's partition in Procedure 270, Word 2. If the attendant tries to activate or cancel a restriction for any other controlled restriction group, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to activate and cancel restrictions for any extension or any controlled restriction group in the switch.

It is the responsibility of the system manager to assign extensions to controlled restriction groups. However, there are no tests in Procedure 000, Word 2 to ensure partitioning of the 63 groups. It is strongly recommended that every member of each group belong to the same extension partition.

Restriction—Miscellaneous Trunk Restrictions

Miscellaneous Trunk Restrictions can be assigned to a voice terminal class of service in the normal manner. However, the capacity of this feature has not been increased for use with the Tenant Services feature. There remains a capacity of four trunk groups in each of eight restriction groups. With this limited capacity, the Miscellaneous Trunk Restrictions feature is only useful for limited applications (such as, to restrict Loudspeaker Paging or Call Park).

Ringing—Distinctive

The switch provides 1-burst ringing for terminal-to-terminal calls inside an extension partition. The switch also provides 1-burst ringing for terminal-to-terminal calls between an extension partition and Exension Partition 0.

Otherwise, terminal-to-terminal calls are not allowed to cross partition boundaries. In order to call a voice terminal in any other partition, a voice terminal user must dial the appropriate 7-digit number which routes the call over a CO (Central Office) trunk. When this is done, the switch provides 2-burst ringing at the called voice terminal.

The switch provides 2-burst ringing for incoming calls from the public network in the usual manner.

In a partition System 85 or DEFINITY Generic 2, the switch provides the usual 2-burst ringing for direct attendant calls or attendant-extended calls to voice terminals (whenever these calls are allowed by the Tenant Services feature).

When Priority Calling, Override, Automatic Callback, or On-Hook Queuing are provided on the switch and allowed in the partitioned environment, the switch provides 3-burst ringing in the usual manner.

Ringing Transfer

There are no tests in Procedures 052, Word 2 and 054, Word 1 to ensure that the members of a Ringing Transfer relationship belong to the same extension partition. However, if the Bridged Call feature is correctly partitioned, the Ringing Transfer feature would also be fully partitioned. It is the responsibility of the system manager to ensure that these Ringing Transfer relationships do not cross partition boundaries.

Route Advance

When the Route Advance feature is invoked for an outgoing call in a partitioned switch, the call-processing software checks each alternate trunk group in the list to ensure that the calling party's extension partition is allowed to use that trunk group.

The details are as follows: When a user attempts to place a call over an outgoing trunk group, the switch checks to determine whether the user is allowed to use the trunk group and whether there is an available trunk in the group. If either of these conditions is not met, and if a Route Advance list ***is not assigned,*** the switch returns intercept treatment to the calling party (or returns reorder tone or queues the call if these options are available). If a Route Advance list ***is assigned*** to the originally dialed trunk group, the switch checks the first trunk group in the list to determine if the user is allowed to use that trunk group and whether there is an available trunk in the group. If either of these conditions is not met, the switch performs the same check on successive trunk groups in the list until: either the call is completed over an allowable and available trunk, or an available trunk cannot be found (the call is queued, or reorder tone is returned), or an allowable trunk group cannot be found (intercept treatment is returned).

The recommended approach for assigning Route Advance lists is to assign trunk groups that are dedicated to a partition as the initial trunk groups in the list. These trunk groups can be followed by shared trunk groups that are also accessible to the same partition However, them are no tests in Procedure 100, Word 4 to prevent other ways of assigning these lists. It is the responsibility of the system manager to ensure that the Route Advance patterns are designed in a practical manner.

Serial Calls

The Serial Calls feature is naturally partitioned on System 85 and DEFINITY Generic 2. In a properly partitioned switch, an attendant cannot receive a call that breaks the rules of partitioning. The attendant must extend each call according to partitioning rules. And, the Serial Calls feature always returns the call to the same attendant console that extended the initial call of the series.

Straightforward Outward Completion

The Straightforward Outward Completion feature generally allows an attendant to complete outgoing calls for a voice terminal user in the switch. However, in a partitioned System 85 or DEFINITY Generic 2, this option is controlled. If an attendant* (in a partition other than Partition 0) whose partition is assigned to the voice terminal user's partition tries to place a call over a trunk that partitioning would not allow the voice terminal user to access, the switch returns intercept treatment to the attendant.

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* An attedant whose partition is ***not*** assigned to the voice terminal user's partition cannot access that extension partition's trunk group.

An attendant in Partition 0 can place outgoing calls over any trunk for voice terminal users in any partition. When these attendants press the RELEASE button after making an outgoing connection, the outgoing call is left intact.

Terminal Busy Indications

There are no tests in Procedure 055, Word 31 to ensure that the signaling terminal and the signaled terminal(s) in a Terminal Busy Indications relationship belong to the same extension partition. If this partitioning is either desired or legally required, it is the responsibility of the system manager to ensure that these relationships do not cross partition boundaries.

Through Dialing

The Through Dialing feature generally allows an attendant to access outgoing trunks for voice terminal users in the switch. Once a trunk has been accessed by an attendant, a voice terminal user can dial the call. However, in a partitioned System 85 or DEFINITY Generic 2 this operation is controlled. If an attendant (in a partition other than Partition 0) whose partition is assigned to the voice terminal user's partition* tries to access a trunk that partitioning would not allow the voice terminal user to access, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 can access any outgoing trunk for voice terminal users in any partition. When these attendants press the RELEASE button after receiving outside dial tone, Through Dialing proceeds normally.

Timed Recall on Outgoing Calls

The Timed Recall on Outgoing Calls feature generally notifies an attendant when the duration of an outgoing trunk call (placed by a nonexempt voice terminal user) has exceeded the preset time interval for the trunk group used. The operation of Timed Recall attendant notification is modified for a partitioned switch. When the preset time interval is exceeded in a partitioned switch, the Timed Recall feature ***would first notify*** an attendant whose attendant partition is associated with the voice terminal user's extension partition. If the terminal user's partition is not assigned to an attendant partition, the switch ***will instead notify*** an attendant whose partition is associated with the trunk group used.

Timed Reminder

The Timed Reminder feature is naturally partitioned on System 85 and DEFINITY Generic 2. In a properly partitioned switch, an attendant cannot receive a call that breaks the rules of partitioning. The attendant must extend the call according to partitioning rules. And Timed Reminder always returns the call to the same attendant console that extended the call.

Transfer

A voice terminal user (in an extension partition other than Extension Partition 0) can transfer calls to extension numbers within the same partition or to extensions in Extension

Partition 0. When the user ties to transfer a call to any other extension partition (using an extension number), the switch returns intercept treatment to the transferring party.

A voice terminal user in Extension Partition 0 is allowed to transfer calls (using an extension number) to any voice terminal in the switch.

Trunk Group Busy/Warning Indicators to Attendant

A partitioned switch only updates the lamps associated with a DTGS (Direct Trunk Group Selection) button for the attendant consoles that are allowed to access that specific trunk group.

Trunk Verification—Attendant

An attendant (in a partition other than Attendant Partition 0) is allowed to verify trunks within trunk groups that are assigned to that attendants partition. When this is done, the feature operates normally. However, when an attendant attempts to verify a trunk that is not within a trunk group assigned to the partition, the switch will return intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to verify any trunk that can otherwise be verified using this feature.

Trunk Verification—Voice Terminal

The Trunk Verification—Voice Terminal feature is unaffected in a partitioned switch. The user of either the internal or the remote designated voice terminal can verify, maintenance busy, or maintenance unbusy any trunk that can otherwise be verified using this feature.

The designated terminal(s) can reside in any partition, but it is strongly recommended that these terminals be assigned to Extension Partition 0.

Trunk-to-Trunk Connections

An attendant (in a partition other than Attendant Partition 0) has limited access to the Trunk-to-Trunk Connections feature. When both trunk groups involved in the connection are assigned to the attendant's partition, then the Trunk-to-Trunk Connections feature is allowed. The switch returns intercept treatment to the attendant when this condition is not met.

An attendant in Attendant Partition 0 has full access to the Trunk-to-Trunk Connections feature. These attendants are allowed to connect any two trunks in the switch.

Unattended Console Service—Alternate Console Position

The Alternate Console Position feature is well behaved in a partitioned switch. Using the Alternate Console Position feature, an alternate console is physically connected to its primary console, and is activated with a 609A transfer panel. Since the switch performs partitioning checks before calls can be routed to/from the primary console, an alternate console would have the same abilities in a partitioned switch as the specific primary console to which it is connected. The system manager need only ensure that the operators of both consoles are members of the same partition.

Unattended Console Service—Call Answer From Any Voice Terminal

Call Answer From Any Voice Terminal is not fully partitioned. In response to the system-wide "gong alerting," the CAAVT access code can be dialed from an unrestricted voice terminal belonging to any extension partition. When a CAAVT call is answered using a voice terminal (in a partition other than Extension Partition 0), the answering party is allowed to transfer the call to an extension in the same extension partition or to an extension in Extension Partition 0. If the answering party tries to transfer the call to any other partition, intercept treatment is returned.

When a CAAVT call is answered using a voice terminal in Extension Partition 0, the answering party is allowed to transfer the call to any extension partition in the switch.

Unattended Console Service—Preselected Call Routing

Default Night Terminal

For an ***unpartitioned*** System 85 or DEFINITY Generic 2, the default night terminal is assigned on a system-wide basis in Procedure 275, Word 2.

For a ***partitioned*** System 85 or DEFINITY Generic 2, the system-wide administration is blocked. Instead, a default night terminal can be assigned to each attendant partition in Procedure 270, Word 3.

Trunk-to-Terminal Assignments

Using an ***unpartitioned*** System 85 or DEFINITY Generic 2 with full night service, an attendant can dial the Assign Terminal to Trunk access code to set trunk-to-terminal assignments for Preselected Call Routing.

Using a ***partitioned*** System 85 or DEFINITY Generic 2, this function of full night service is disabled. Instead, these associations can only be assigned in Procedures 116 and 150. However, there are no software checks in these procedures to insure the proper partitioning of these associations. It is the responsibility of the system manager to design allowable and practical trunk-to-terminal assignments.

Common Night Terminal

Using an ***unpartitioned*** System 85 or DEFINITY Generic 2, an attendant can dial the Assign Common Terminal access code to designate the common night terminal for the entire switch.

Using a ***partitioned*** System 85 or DEFINITY Generic 2, this functionality is modified. To assign a common night terminal for a partition, the partition's controlling attendant dials the Assign Common Terminal access code, the extension number of the partition's common night terminal, and then the number of the attendant partition.

Clear All Terminals

Using an ***unpartitioned*** System 85 or DEFINITY Generic 2, an attendant can dial the Clear All Terminals access code to cancel the common night terminal assignment and / or the current trunk-to-terminal assignments for Preselected Call Routing. When this is done, Preselected Call Routing sends incoming calls to the default night terminal (if assigned), and then to the CAAVT queue (if assigned).

Using a ***partitioned*** System 85 or DEFINITY Generic 2, this functionality is modified. A partition's controlling attendant can dial the Clear All Terminals access code, followed by the number of the attendant partition. When this is done, ***only*** the common night terminal assignment for that partition is canceled. (The trunk-to-terminal associations, assigned in Procedures 116 and 150, are not affected.) So, Preselected Call Routing would send incoming calls to the trunk-to-terminal association (if assigned), then to the partition's default night terminal (if assigned), and then to the CAAVT queue (if assigned).

Override Common Night Terminal

Using an ***unpartitioned*** System 85 or DEFINITY Generic 2, an attendant can dial the Override Common Night Terminal access code to bypass the common night terminal and the default night terminal, and divert calls directly to the CAAVT queue.

Using a ***partitioned*** System 85 or DEFINITY Generic 2, the Override Common Night Terminal function is disabled. This function is no longer necessary because CAAVT "gong alerting" is not partitioned.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, the AAR and ARS features are replaced by the WCR feature. On a partitioned switch, one or more partitions are assigned to one of the WCR call categories in Procedure 317, Word 1. With the WCR feature, the number of call categories increases to 256 (from the 64 available with the ARS feature). The call category and VNI are used to determine the routing pattern. As a result, a WCR network call can be handled by a different pattern based on the partition's assigned call category. Further, once a network call enters the assigned pattern, call-processing software checks successive preferences in the pattern (in a similar manner to the algorithm described in the Route Advance interaction) to determine whether the call is allowed to use the trunk group.

With WCR, time-of-day plans can *in effect* be partitioned by association with a call category. In this way, each partition can have its own set of time-of-day / call category definitions.

Hardware Requirements

The Tenant Services feature may require the following additional or special hardware.

To provide an attendant console at a remote module:

- 107A ORPI (Optically Remoted Peripheral Interface).

  NOTE: Since the 107A ORPI connects to the common control carrier to provide digital signaling for a remote attendant console, this equipment is used for an attendant console in either a remote traditional module or a remote universal module.

Feature Administration

The Tenant Services feature is assigned on a per-attendant partition and on a per-extension partition basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — TENANT SERVICES

PROCEDURE	WORD	PURPOSE	SMT
000	4	Assigns an extension or a range of extensions to a partition Unless explicitly translated, extensions are assigned to Extension Partition 0 by default.	Yes
210	2	Assigns an attendant partition (0 to 40) to an attendant console position (1 to 40). Also, assigns an attendant position as a partition's controlling attendant console.	Yes
270	1	Associates an attendant partition (0 to 40) with one or more extension partitions (0 to 999).	Yes
270	2	Assigns the characteristics of an attendant partition including attendant overflow parameters and associated controlled restriction groups.	Yes

(Continued)

ADMINSTRATION PROCEDURES TENANT SERVICES *(Continued)*			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
270	3	Assigns the default extension for preselected call routing to an attendant partition This procedure also displays the attendant partition's common night extension.	Yes
270	4	Assigns an attendant partition (0 to 40) to a listed directory number.	Yes
270	5	Associates a trunk group with one or more extension partitions and one attendant partition.	Yes
276	1	Assigns the Tenant Services feature to the feature-group class of service.	No
282	1	Associates an authorization code with an extension partition.	Yes
314	1	On System 85 and Generic 2.1 switches, assigns an ARS pattern number (1 to 64) to a call category (0 to 63) and a routing designator (1 to 64). On Generic 2.2 switches, this procedure is used to define string identified for digit analysis.	Yes
317	1	On Generic 2.2 switches, associates a call category with an attendant or extension partition.	N/A
317	2	On Generic 2.2 switches, associated a call category and VNI with a routing pattern.	N/A
320	2	On System 85 and Generic 2.1 switches, associates an ARS call category with an extension partition. This Word does not exist for Generic 2.2 switches.	Yes
320	3	On System 85 and Generic 2.1 switches, associates an ARS call category with an attendant partition This Word does not exist for Generic 2.2 switches.	Yes
350	1	Assigns the first digit of the feature dial access codes (if required).	No
350	2	Assigns the access codes for the Tenant Services feature. The applicable encodes are as follows. 97 Activate Unattended Console Service (for a partition) 98 Deactivate Unattended Console Service (for a partition).	No

Notes:

Terminal Busy Indications

Description

This feature provides a visual indication of the on-hook / off-hook status of a monitored voice terminal (called the signaling terminal), without a shared appearance. The signaling terminal can be a multiappearance voice terminal or a SLS (Straight Line Set).

The monitoring terminal, known as the signaled terminal, must be on a multiappearance voice terminal. The terminal busy indication lamp is assigned to an unused appearance or feature button on the signaled terminal. There must be a status lamp associated with the button used for terminal busy indication. A button on an attached adjunct such as a coverage module or a feature module can be used for the terminal busy indication lamp.

When the signaling terminal is on-hook, the terminal busy indication lamp on the signaled terminal is dark. When the signaling terminal goes off-hook, the terminal busy indication lamp on the signaled terminal lights. This indication is provided for any voice appearance on the signaling terminal. An alternate answering position, such as a secretary or receptionist can use this information when handling calls for a busy primary position.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since the feature was first introduced.

User Operations

None.

Considerations

Terminal Busy Lamp

Each Terminal Busy Indication lamp can only display the status of one voice terminal. It is recommended that one voice terminal signal no more than 17 multiappearance voice terminals.

Straight Line Sets

A multiappearance terminal can monitor the busy or idle status of a single-appearance terminal if the single-appearance terminal is administered as an SLS (straight line set) using Procedures 000, 051, and 052.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Bridged Call

The button assigned for the terminal busy indication on the signaled terminal is not a shared appearance button. This button cannot be used to bridge onto an active call or to place or receive calles, even if an appearance button (two lamps) is used for this function.

Data Call Setup

The Terminal Busy Indications feature applies to voice appearances only. If a signaling terminal is active on a data appearance but not on a voice appearance, the terminal busy indication lamp will show the station as being in the on-hook state.

Tenant Services

There are no tests in Procedure 055, Word 1 to ensure that the signaling and the signaled terminal(s) in a Terminal Busy Indications relationship belong to the same extension partition. If this partitioning is either desired or legally required, it is the responsibility of the system manager to ensure that these relationships do not cross partition boundaries.

Hardware Requirements

For either a traditional or universal module, a single-appearance voice terminal can be assigned as the signaling, but not as the signaled, voice terminal.

Feature Administration

Assignment of the Terminal Busy Indication feature is on a per-voice terminal basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — TERMINAL BUSY INDICATIONS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns an analog extension for use on multiappearance voice terminals.	Yes
051	1	Assigns an analog voice terminal as type SLS to operate as a signaling terminal.	Yes
052	1	Displays the voice terminal button types.	Yes
055	1	Administers a signaling and signaled terminal with lamp for Terminal Busy Indication.	Yes

The following is the applicable TCM path name used with the AP 16.

TCM SRCREEN — TERMINAL BUSY INDICATIONS	
PATH NAME	**PURPOSE**
terminal-change terminal station-busy	Displays or prints a report of Terminal Busy Indication assignments.

Notes:

Through Dialing

Description

With the Through Dialing feature, an attendant selects a trunk group (or network) for an outgoing call. The calling party then dials the address digits required to complete the call. This allows the attendant to retain control of trunk use while reducing attendant call-processing time. With Through Dialing, trunk groups can be accessed directly using the trunk group DAC (Dial Access Code), or one of the network routing features (AAR [Automatic Alternate Routing], ARS [Automatic Route Selection], or WCR [World Class Routing]) can be used to select the outgoing trunk group.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Complete a Through Dialing Request for a User by an Attendant

Without a trunk-group button available and maintaining supervision:

1. The voice terminal user calls an attendant. [PA lamp goes out, ATND lamp lights, and the ICI displays the extension number.]
2. Attendant presses the appropriate loop button. [Attendant and user are connected.]
3. The voice terminal user requests a call to an outside party.
4. Press **[START]** . [SPLIT lamp lights, calling party is split away from the attendant, and the dial tone is heard.]
5. Dial the trunk-group access code. [Second dial tone is heard, and the ANS lamp lights.]
6. Press **[HOLD]** . [Attendant is removed from the connection, calling party hears dial tone, ATND and SPLIT lamps go out, the ICI display goes out, and the HOLD and PA lamps light.]
7. The calling party dials the number of the party being called.

Without a trunk-group button available and not maintaining supervision:

1. The voice terminal user calls an attendant. [PA lamp goes out, ATND lamp lights, and the ICI displays the extension number.]

2. Attendant presses the appropriate loop button. [Attendant and user are connected.]
3. The voice terminal user requests a call to an outside party.
4. Press **[START]** . [SPLIT lamp lights, calling party is split away from the attendant, and dial tone is heard.]
5. Dial the trunk-group access code. [Second dial tone is heard, and the ANS lamp lights.]
6. Press **[RELEASE]** . [Attendant is removed from the connection; calling party hears dial tone; ANS, ATND, and SPLIT lamps go out; the ICI display goes out; and the PA lamp lights.]
7. The calling party dials the number of the party being called.

With a trunk-group button available and maintaining supervision:

1. The voice terminal user calls an attendant. [PA lamp goes out, ATND lamp lights, and the ICI displays the extension number.]
2. Attendant presses the appropriate loop button. [Attendant and user are connected.]
3. The voice terminal user requests use of the Through Dialing feature.
4. Press the appropriate trunk-group button. [SPLIT and ANS lamp lights, the calling party is split away from the attendant, and dial tone is heard.]
5. Press **[HOLD]** . [Attendant is removed from the connection, calling party hears dial tone, ATND and SPLIT lamps go out, the ICI display goes out, and the HOLD and PA lamps light.]
6. The calling party dials the number of the party being called.

With a trunk-group button available and not maintaining supervision:

1. The voice terminal user calls an attendant. [PA lamp goes out, ATND lamp lights, and the ICI displays the extension number.]
2. Attendant presses the appropriate loop button. [Attendant and user are connected.]
3. The voice terminal user requests use of the Through Dialing feature.
4. Press the appropriate trunk-group button. [SPLIT and ANS lamp lights, the calling party is split away from the attendant, and dial tone is heard.]
5. Press **[RELEASE]** . [Attendant is removed from the connection; calling party hears dial tone; ANS, ATND, and SPLIT lamps go out; the ICI display goes out; and the PA lamp lights.]
6. The calling party dials the number of the party being called.

Using a Network Routing feature:

1. The voice terminal user calls an attendant. [PA lamp goes out, ATND lamp lights, and the ICI displays the extension number.]
2. Attendant presses the appropriate loop button. [Attendant and user are connected.]
3. The voice terminal user requests use of the Through Dialing feature.
4. Press **[START]** . [SPLIT and ANS lamp lights, the calling party is split away from the attendant, and dial tone is heard.]
5. Dial the network dial access code. [Second dial tone is heard (if administered for the network), and the ANS lamp lights.]
6. Press **[RELEASE]** . [Attendant is removed from the connection; calling party hears dial tone; ANS, ATND, and SPLIT lamps go out; the ICI display goes out; and the PA lamp lights.]

   or

   Press **[HOLD]** . [Attendant is removed from the connection, calling party hears dial tone, ATND and SPLIT lamps go out, the ICI display goes out, and the HOLD and PA lamps light.]
7. The calling party dials the number being called.

Considerations

Straightforward Outward Completion and Through Dialing

The Straightforward Outward Completion feature is similar to the Through Dialing feature. Using Straightforward Outward Completion, the attendant (instead of the calling party) dials the desired digits after second dial tone. The attendant does not press the RELEASE button until ringback tone is returned.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Control of Trunk Group Access

The Through Dialing feature is one means of accessing outgoing trunk groups that are under attendant control.

AAR (Automatic Alternate Routing)

On System 85 and DEFINITY Generic 2.1 switches, Through Dialing can be used to extend private network calls via the AAR feature. When this is done, the attendant's FRL is used for the call rather than the calling extension FRL. When Through Dialing is used, trunks under attendant control can be selected. See also the CDR feature interaction.

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, Through Dialing can be used to extend public network calls via the ARS feature. When this is done, the attendants FRL is used for the call rather than the calling extension FRL. When Through Dialing is used, trunks under attendant control can be selected. See also the CDR feature interaction.

CDR (Call Detail Recording)

Whenever the attendant allows a terminal user to use the Through Dialing feature, CDR records the call as an attendant-assisted call.

FEAC (Forced Entry of Account Codes)

The FEAC function requires that an account code be entered before an outgoing call can be placed. For attendant completed calls, this requirement does not apply. However, for Through Dialing calls, the attendant is initiating the call but not completing it. In this case, when an account code is required before the trunk group or network DAC, the attendant must enter the account code before entering the DAC and returning control of the call to the calling station.

However, when either the AAR or WCR feature is used they can be optioned to accept an account code immediately after the network DAC. In this case, the user must enter the account code before the address digits.

Centralized Attendant Service

When a switch is a branch location in a Centralized Attendant Service network, the Centralized Attendant Service attendant can allow terminal users at the branch to Through Dial calls at the branch.

Last Number Dialed

The Last Number Dialed feature stores and redials the digits dialed by a voice terminal user during a Through Dialing connection. However, for a Through Dialing call, an attendant initially accesses a trunk group for the voice terminal user. Therefore, when digits are redialed using the Last Number Dialed feature, the trunk-group access digit(s) would not be included as outpulsed digits.

Restriction—Attendant Control of Voice Terminals

Controlled Outward Restriction can be bypassed by using the Through Dialing feature.

Restriction—Code Restriction

Through Dialing bypasses Code Restriction.

Restriction—Miscellaneous Trunk Restrictions

The Through Dialing feature operates normally from the restricted terminal at the attendant's discretion. If the attendant determines that the terminal should be allowed access to the trunk, the miscellaneous trunk restriction checks are bypassed when the attendant seizes the trunk and then releases to allow the terminal user to complete the dialing.

Restriction—Toll Restriction

Through Dialing bypasses Toll Restriction.

Restriction—Voice Terminal Restrictions

Through Dialing bypasses Outward Restriction.

Tenant Services

The Through Dialing feature generally allows an attendant to access outgoing trunks for voice terminal users in the switch. Once a trunk has been accessed by an attendant, a voice terminal user can dial the call. However, in a partitioned System 85 or DEFINITY Generic 2, this operation is controlled. If an attendant (in a partition other than Partition 0) whose partition is assigned to the voice terminal user's partition* tries to access a trunk that partitioning would not allow the voice terminal user to access, the switch returns intercept treatment to the attendant.

An attendant in Attendant Partition 0 can access any outgoing trunk for voice terminal users in any partition. When these attendants press the RELEASE button after receiving outside dial tone, Through Dialing proceeds normally.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, Through Dialing can be used to extend network calls via the WCR feature. When this is done, the attendants FRL is used for the call rather than the calling extension FRL. When Through Dialing is used, trunks under attendant control can be selected. See also the CDR feature interaction.

Hardware Requirements

None.

Feature Administration

The Through Dialing feature is provided in all switches. Feature assignment is not required.

* An attendant whose partition is ***not*** assigned to the voice terminal user's partition cannot access that extension partition's trunk group.

Notes:

Timed Recall on Outgoing Calls

Description

This feature provides control over the use of outgoing trunks when there is an excessive number of lengthy calls. Timed Recall automatically transfers control of the outgoing calls from selected terminals to an attendant after a predetermined time interval of 1 to 31 minutes. This feature applies to extensions with a specific class of service designated in switch translation (see Feature Administration). The switch sends a warning tone to the calling party 30 seconds before the transfer occurs. The warning tone is a 440-hertz tone with a duration of 1 second.

When a call is transferred to an attendant, it is identified by the Attendant Display feature. At this time, the attendant assumes control of the call. The attendant may talk with the calling and called party to decide whether the call should continue or not.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Answer a Transferred Call:

Press the appropriate loop button. [The PA lamp goes out. The ATND lamp lights.]

To Discontinue the Call:

Press **[CANC]** . [ATND lamp and ICI display go dark. The call is disconnected, and the trunk is released.]

To Continue the Call for Another Timed Interval:

Press **[RELEASE]** . [ATND lamp and ICI display go dark The 2-way connection is restored. The attendant is released from the call.]

To Allow the Call to Continue Indefinitely:

Press **[HOLD]** . [The call is held on the attendant console. The HOLD lamp lights. The attendant is removed from the connection.] (The call is no longer subject to Timed Recall.)

Considerations

Timed Interval

An individual timed interval can be assigned to each trunk group selected for timed recall. The timed interval can be any value between 1 and 31 minutes.

Recall Level

An individual recall level can be assigned to each trunk group. When the number of idle trunks in a group falls below the assigned level (from 0 to 7), Timed Recall is activated for the trunk group.

2-Way Trunk Groups

If the feature is assigned to a 2-way trunk group, the Timed Recall is in effect for ***both incoming*** and ***outgoing*** calls.

Attendant Calls

Outgoing calls placed by an attendant and incoming calls to an attendant console (over a timed 2-way trunk group) are exempt from recall.

Night Mode

If the attendant console is in the night mode, the Timed Recall feature is disabled.

Hard and Soft Processor Swaps

Stable calls on timed trunk groups will endure a hard processor swap.

During a hard processor swap, the switch ceases timing outgoing calls and does not place Timed Recalls to an attendant.

After a hard processor swap, the switch resumes timing controlled calls. However, the clock resets to zero. So, the switch does not place a Timed Recall until another complete time interval elapses.

The Timed Recall on Outgoing Calls feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Busy Verification of Lines

Busy verification is denied if the terminal line being verified is connected to a call which has already been switched to an attendant position by means of the Timed Recall on

Outgoing Calls feature. Busy verification is allowed on terminal-to-trunk calls that have not yet been switched to the attendant by this feature. The recall timing is suspended as long as the attendant remains bridged on the connection by the Busy Verification of Lines feature and is resumed when the attendant releases the line.

Conference—Attendant Five Party

The Timed Recall on Outgoing Calls feature does not apply to an attendant conference connection.

Conference—Attendant Six Party

The Timed Recall on Outgoing Calls feature does not apply to an attendant conference connection.

Conference—Three Party

A voice terminal may place an outgoing call and later add a second trunk to the call. The first trunk, already in use and timing, receives recall treatment (that is, times out and routes to the attendant) even after the addition of the second trunk to the call. Afterwards, the trunk with the shorter recall time is timed for recall treatment.

Data Protection

This feature prohibits the attendant from intervening or monitoring calls on outgoing trunks. Thus, Timed Recall on Outgoing Calls cannot be used if Data Protection is assigned.

Malicious Call Trace

During a malicious call, the Timed Recall feature is deactivated for a trunk that is involved in the malicious call.

An activating console receives modified Timed Recall operation during a Malicious Call Trace. If the activating console is position busy, the Priority lamp lights during a timed recall, but the tone does not sound. If the console is not position busy, the Priority lamp lights and the tone sounds during a timed recall.

A controlling console receives modified Timed Recall operation during a Malicious Call Trace. While the controlling attendant is performing the trace, the Priority lamp lights during a timed recall, but the tone does not sound. After the trace is deactivated, the Priority lamp lights and the tone sounds for a timed recall.

Privacy—Attendant Lockout

If Privacy—Attendant Lockout is activated, the attendant cannot release after a timed recall by pressing the HOLD button. The switch ignores any press of the HOLD button.

Tenant Services

The Timed Recall on Outgoing Calls feature generally notifies an attendant when the duration of an outgoing trunk call (placed by a nonexempt terminal user) has exceeded the preset time interval for the trunk group used. The operation of Timed Recall attendant notification is modified for a partitioned switch. When the preset time interval is exceeded in a partitioned switch, the Timed Recall feature ***would first notify*** an attendant whose attendant partition is associated with the terminal user's extension partition. If the terminal user's partition is not assigned to an attendant partition, the switch ***will instead notify*** an attendant whose partition is associated with the trunk group used.

Unattended Console Service—Preselected Call Routing

While the Preselected Call Routing feature is active, the Timed Recall on Outgoing Calls feature is deactivated.

Restricting Feature Use

Timed Recall is not allowed on extensions with the exempt status administered in the extension's class of service.

Hardware Requirements

None.

Feature Administration

Assignment of the Timed Recall on Outgoing Calls feature is on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — TIMED RECALL ON OUTGOING CALLS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
000	1	Assigns the class of service to an extension number.	Yes
010	1	Assigns Timed Recall Exempt to an extension class of service.	Yes
101	1	Specifies the time (0 to 31) and the level (0 to 7) for Timed Recall on Outgoing Calls.	No
204	1	Designates the desired alphanumeric display for recalls to the attendant.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — TIMED RECALL ON OUTGOING CALLS	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns Timed Recall Exempt to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Notes:

Timed Reminder

Description

The Timed Reminder feature automatically alerts the attendant after 30 seconds for the following types of calls:

- Calls on a switched loop waiting for the called party to answer
- Calls on a switched loop waiting to be connected to a busy extension number
- Incoming calls placed on hold on the console.

The attendant can reenter the call and decide whether to terminate the call or permit the waiting to continue. By reentering the call, the attendant lets the calling party know that the call has not been forgotten. The Timed Reminder feature enables the attendant to keep track of every call and provides callers with good service.

Timed Reminder operates in two phases: the 30-second timing phase and the attendant alerting phase. The timing phase begins when an attendant releases from a connection by pressing the RELEASE or the HOLD button. The attendant alerting phase begins after the timing phase elapses. The switch alerts the attendant by flashing the RING, BUSY, or HOLD lamp on the attendant console, whichever is applicable, and emitting a 1950-hertz tone.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

To Use Timed Reminder When the Called Voice Terminal Is Idle:

1. Attendant receives a call. [ATND lamp lights.]
2. Press **[START]** . [SPLIT lamp lights, and calling party is split away from the attendant.]
3. Dial the called voice terminal's extension. [ATND lamp goes out, and RING lamp lights.]
4. Press **[RELEASE]** . [The timing phase starts. SPLIT lamp goes out, and PA lamp lights.]
5. After 30 seconds, the timing phase completes. [RING lamp flashes, and a tone is generated.]

To Use Timed Reminder When the Voice Terminal Is Busy

If the Attendant Call Waiting feature isn't activated:

1. Attendant receives a call. [ATND lamp lights.]
2. Press **[START]** . [SPLIT lamp lights, and calling party is split away from the attendant.]
3. Dial the called voice terminal's extension. [ATND lamp goes out, BUSY lamp lights, and the attendant hears busy tone.]
4. Press **[CANC]** . [Attendant is reconnected to the calling party. BUSY and SPLIT lamps go out, ATND lamp lights, and busy tone removed.]
5. Verify that calling party will wait.
6. To activate the Timed Reminder feature, press **[HOLD]** . [Calling party hears a special ringback tone, ATND lamp goes out, and the HOLD and PA lamps light.]
7. The timing phase completes. [HOLD lamp flashes, and a tone is generated.]

If the Attendant Call Waiting feature is activated:

1. Attendant receives a call. [ATND lamp lights.]
2. Press **[START]** . [SPLIT lamp lights, and calling party is split away from the attendant.]
3. Dial the called voice terminal's extension. [The called voice terminal is busy, and the attendant is reconnected to the calling party. ATND lamp goes out, BUSY lamp lights, and the attendant hears confirmation tone.]
4. Verify that calling party will wait.
5. To begin the Timed Reminder feature, press **[RELEASE]** . [SPLIT lamp goes out, BUSY and PA lamps light, and the Call Waiting tone is transmitted to the called voice terminal.]
6. The timing phase completes. [BUSY lamp flashes, and a tone is generated.]

To Use Timed Reminder When the Call Uses a Trunk

When the call reaches the called voice terminal:

1. Attendant receives a call. [ATND lamp lights.]
2. Dial the trunk-group access code or the appropriate DTGS button. [SPLIT lamp lights.]
3. The trunk is seized. [Dial tone]
4. Attendant or calling party dials the called voice terminal's extension. [ATND lamp goes out, and ANS lamp lights.]
5. Press **[HOLD]** . [ANS and SPLIT lamps go out. PA and HOLD lamps light.]

6. Calling party is connected to called party.
7. To begin the Timed Reminder feature, one party goes on-hook.
8. The timing phase completes. [HOLD lamp flashes, and a tone is generated.]

The call does not reach the called voice terminal:

1. Attendant receives a call. [ATND lamp lights.]
2. Dial the trunk-group access code or the appropriate DTGS button. [SPLIT lamp lights.]
3. The trunk is seized. [Dial tone]
4. Attendant or calling party dials the called voice terminal's extension. [ATND lamp goes out, and ANS lamp lights.]
5. Press **[HOLD]** . [ANS and SPLIT lamps go out. PA and HOLD lamps light.]
6. Calling party is not connected to called party and the Timed Reminder feature begins.
7. The timing phase completes. [HOLD lamp flashes, and a tone is generated.]

To Reconnect to the Calling Party

From an idle loop:

Press the appropriate loop button. [RING and PA lamps go out, ATND lamp lights, and tone quits.]

From a busy loop without Attendant Call Waiting:

Press the appropriate loop button. [HOLD and PA lamps go out, ATND lamp lights, and tone quits.]

From a busy loop with Attendant Call Waiting:

Press the appropriate loop button. [BUSY and PA lamps go out, ATND lamp lights, and tone quits.]

To Cancel the Call to the Called Terminal

If the call is to be terminated:

The calling party can go on-hook, or the attendant can press **[RELEASE]** .

If the call is to be extended to another number:

1. Be sure the attendant and calling party are reconnected, and the ATND lamp is lit.
2. Press **[CANC]** .
3. Begin again as if attendant had just received the call. See uses of the Timed Reminder feature in the beginning of the "User Operations" section.

Considerations

Ending the Cycle

The timed reminder cycle is ended when:

- The called voice terminal goes on-hook and connects with the calling party.
- The calling party goes on-hook
- 30 seconds elapses.

Serial Calls

During activation of a Serial Call, an attendant presses the HOLD button to retain access to the switched loop. Timed Reminder timing begins when one call is finished, and the calling party is waiting for the next call to be initiated. After the timed interval elapses, the HOLD lamp flashes, and the attendant should return to the calling party by pressing the appropriate loop button. The attendant is now able to extend the call to the next call destination in the series.

Up to six calls per attendant console can be using the Timed Reminder feature at one time. The timed reminder cycle can be repeated as often as necessary.

Call Waiting Tone

The Timed Reminder call waiting tone is two 100-millisecond, 400-hertz beeps.

Hard and Soft Processor Swaps

When a hard processor swap occurs, unanswered and held calls are dropped by the switch.

Since these calls are dropped, the switch does not place Timed Reminders to an attendant during the hard swap.

The Timed Reminder feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Release Loop Operation

The Attendant Release Loop Operation feature is similar to the Timed Reminder feature, and both of these features can reside on the same switch. The Timed Reminder feature is always provided, while the Attendant Release Loop Operation feature is optional.

When both features reside on the same switch, these two features operate simultaneously and control two separate sets of calls. Incoming trunk calls are controlled by the Attendant Release Loop Operation feature, and the timed reminder interval for these calls is assigned in Procedure 275, Word 4. Other affected calls are controlled by the Timed Reminder feature, and the timed reminder interval for these calls is fixed at 30 seconds.

When only the Timed Reminder feature resides on the switch, ***all*** affected calls are controlled by the Timed Reminder feature, and the timed reminder interval is fixed at 30 seconds.

ACD (Automatic Call Distribution)

The Timed Reminder feature does not apply to calls that an attendant extends to an ACD split. Once an attendant-extended call enters the ACD queue, this call is not timed and no reminder will be given to the attendant.

Call Coverage

When the STA ID button is pressed, the extension number to which the call was redirected is indicated on the attendant display rather than the originally called extension number.

Call Forwarding—Busy and Don't Answer

If the attendant presses the STA ID button after the extended call forwards, the attendant display indicates the extension number that the call forwarded to rather than the originally called extension number.

Call Forwarding—Don't Answer

If the attendant presses the STA ID button after the extended call forwards, the attendant display indicates the extension number that the call forwarded to rather than the originally called extension number.

Call Forwarding—Follow Me

When the STA ID button is pressed, the attendant display indicates the extension number that the call forwaded to rather than the originally called extension number.

Call Vectoring

The Timed Reminder feature does not apply to calls that an attendant extends to a VDN (Vector Directory Number). Once an attendant-extended call enters vector processing, this call is not timed and no reminder will be given to the attendant.

Centralized Attendant Service

The Centralized Attendant Service (CAS) feature uses a recall timing process that is different from the Timed Reminder feature. The timed interval for Centralized Attendant Service is assignable in 2-second intervals up to a maximum of 62 seconds.

The Timed Reminder feature is used on a CAS call in two cases:

If an incoming trunk call is extended to a local extension (an extension on the main) rather than back to a branch extension.

or

After extending a call to a branch extension, the CAS attendant presses **[HOLD]** rather than **[RELEASE]** .

EUCD (Enhanced Uniform Call Distribution)

The Timed Reminder feature does not apply to calls that an attendant extends to an EUCD split. Once an attendant-extended call enters the EUCD queue, this call is not timed and no reminder will be given to the attendant.

Hunting

When the STA ID button is pressed, the extension number to which the call hunted is indicated on the attendant display rather than the originally called extension number.

Malicious Call Trace

The activating console receives modified Timed Reminder operation during a Malicious Call Trace. While the activating attendant is connected to the malicious call, the Ring lamp flashes during a timed reminder, but the tone does not sound. After the activating attendant disconnects from the malicious call, the Ring lamp flashes and the tone sounds during a timed reminder.

The controlling console also receives modified Timed Reminder operation during a Malicious Call Trace. While the controlling attendant is performing the trace, the Ring lamp flashes during a timed reminder, but the tone does not sound. After the trace is deactivated, the Ring lamp flashes and the tone sounds during a timed reminder.

Tenant Services

The Timed Reminder feature is naturally partitioned on System 85 and DEFINITY Generic 2. In a properly partitioned switch, an attendant cannot receive a call that breaks the rules of partitioning. The attendant must extend the call according to partitioning rules. And, Timed Reminder always returns the call to the same attendant console that extended the call.

Restricting Feature Use

The Timed Reminder feature cannot be restricted. However, the attendant can turn off the signal using a switch behind the front panel of the attendant console.

Hardware Requirements

None.

Feature Administration

The Timed Reminder feature is provided on all switches. Assignment or administration is not required.

Touch-Tone Calling Senderized Operation

Description

This feature reduces the time necessary to set up calls to distant locations equipped to receive touch-tone calling signals. The collected digits are sent to the distant office via touch-tone service signaling.

> **NOTE:** Touch-tone service signaling is also known as DTMF (Dual Tone Multifrequency) signaling.

If a distant location is not equipped to receive touch-tone calling signals, the switch generates and sends dial pulses.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

None.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

The Touch-Tone Calling Senderized Operation feature is a prerequisite for implementing Abbreviated Dialing.

AAR (Automatic Alternate Routing)

On System 85 and DEFINITY Generic 2.1 switches, when the Automatic Alternate Routing feature is used to route calls, each call requires a touch-tone calling sender. If a sender is not available, the switch denies the call.

ARS (Automatic Route Selection)

On System 85 and DEFINITY Generic 2.1 switches, when the Automatic Route Selection feature is used to route calls, each call requires a touch-tone calling sender. If a sender is not available, the switch denies the call.

ISDN—PRI (Primary Rate Interface)

There is no direct interaction between Touch-Tone Calling Senderized Operation and ISDN—PRI. These features are compatible through the interworking function. That is, when calls tandem between analog and ISDN facilities interworking takes care of the necessary conversion. Touch-tone calling senders are required on ISDN—PRI calls.

Look-Ahead Interflow

Since the Look-Ahead Interflow feature uses ISDN—PRI trunk facilities where the dialed digits are contained in the interflow SETUP message, Look-Ahead interflow calls ***do not*** rely on touch-tone signaling. However, because touch-tone senders are required on ISDN—PRI calls, they are also required for Look-Ahead Interflow.

WCR (World Class Routing)

On DEFINITY Generic 2.2 switches, when the World Class Routing feature is used to route calls, each call requires a touch-tone calling sender. If a sender is not available, the switch denies the call.

Hardware Requirements

The Touch-Tone Calling Senderized Operation feature requires the following additional or special hardware.

For Traditional Modules:

- SN251 receiver circuits (four circuits per circuit pack).
- SN252 sender circuits (four circuits per circuit pack)

For Universal Modules:

- TN748C tone detector circuits (four receiver and two sender circuits per circuit pack).

Feature Administration

The Touch-Tone Calling Senderized Operation feature is assigned on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES TOUCH-TONE CALLING SENDERIZED OPERATION		
PROCEDURE	WORD	PURPOSE
100	1	Administers the trunk type and trunk group number for touch-tone service. The applicable trunk-type encode includes: 55 Touch-tone sender.
101	1	Administer the type of signaling to be sent over a particular trunk group.
150	1	Assigns the SN252 or TN748C equipment location of a Touch-Tone Sender trunk to its trunk-group number.

Notes:

Touch-Tone Dialing

Description

Touch-Tone Dialing provides quick and easy dialing on a touch-tone dialing pad. This dialing method is standard on voice terminals and attendant consoles for System 85 and DEFINITY Generic 2. Moreover, Touch-Tone Dialing is required for the use of certain features. For example, a touch-tone dialing pad is required to dial access-codes beginning with the special character "*" or "#." Also, Remote Access requires that a dialing register be retained for additional dialing after a line or trunk seizure.

Buttons 0 through 9 are the equivalent of the same numbers on a rotary dial. The two extra buttons, * and #, support special functions such as forming part of access codes. These characters are not available with rotary dialing. When a touch-tone dialing button is pressed, a distinctive tone is generated for each button.

If a distant switching system is unable to accept touch-tone calling signals, the switch has the ability to generate dial pulses by using Touch-Tone Dialing to Dial Pulse Conversion.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

Touch-tone dialing is accomplished by pressing the appropriate button on a touch-tone dialing pad. The tone generated can be heard through the voice terminal receiver. Each button should be pressed firmly and released before the next button is pressed to avoid overlapping tones.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Extension Number Portability

Extension Number Portability requires routing decisions at several points in the digit-collection process. This results in additional call-setup time. Touch-tone dialing is recommended for portability subnetworks to reduce the call-setup time as much as possible.

Hardware Requirements

The Touch-Tone Dialing feature requires the following specific hardware:

For Traditional Modules:

- SN251 touch-tone dialing receiver / register.

For Universal Modules:

- TN748C tone detector.

Feature Administration

Assignment of the Touch-Tone Dialing feature is on a per-trunk group and on a per-terminal class of service basis.

On system 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — TOUCH-TONE DIALING			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the extension class of service to an extension number.	Yes
010	1	Assigns touch-tone dialing to a terminal class of service.	Yes
100	1	Assigns the trunk type of a touch-tone dialing trunk group to trunk-group number 17. The applicable trunk-type encode includes: 2 Touch-tone register.	No
150	1	Assigns the SN251 or TN748C equipment location of a touch-tone dialing trunk to trunk-group number 17.	No

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — TOUCH-TONE DIALING	
PATH NAME	**PURPOSE**
terminal-change class-of-service attributes	Assigns touch-tone dialing to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Notes:

Transfer

Description

The Transfer feature allows voice terminal users to transfer calls to other terminals or trunks without attendant assistance.

The ability to transfer a call without attendant assistance enhances the efficiency of the switch operation and projects a professional business image. This feature can save time for all parties involved especially when a call was initially misdirected or after finishing business with one party and another party is needed to conduct additional business.

Feature History and Development

This feature was first available for System 85 in Release 1.

Beginning with Issue 3.0 of DEFINITY Communications System Generic 2.1, an RLT (Release Link Trunk) can terminate to an ACD split, a VDN (Vector Directory Number), or an attendant console. This enhancement changes the operation of the Transfer feature. For RLTs that terminate to ACD splits or VDNs, any voice terminal with a Conference or Transfer button can be used to transfer incoming RLT calls from the CAS main to a branch location.

User Operations

The following are the user operating procedures for this feature.

Transferring a Call on a Terminal Without a TRANSFER or RECALL Button:

1. Be sure there is a 2-party connection.
2. Momentarily press the switchhook. [The second party is placed on soft hold, and recall dial tone is heard.]
3. Dial the third party. [Ringback tone]
4. If announcement is necessary, wait for the third party to answer and announce the call.
5. If no announcement is necessary, wait to hear ringback tone.
6. Go on-hook [Call is transferred to the third party.]

Transferring a Call on a Terminal With a RECALL (or R) Button, but Without a TRANSFER Button:

1. Be sure there is a 2-party connection.
2. Press **[RECALL]** . [The second party is placed on soft hold, and recall dial tone is heard.]
3. Dial the third party. [Ringback tone]
4. If announcement is necessary, wait for the third party to answer and announce the call.
5. If no announcement is necessary, wait to hear ringback tone.
6. Go on-hook. [Call is transferred to the third party.]

Transferring a Call on a Terminal With A TRANSFER Button (non-RLT call):

1. Be sure there is a 2-party connection.
2. Press **[TRANSFER]** or **[CONFERENCE]** . [The second party is placed on hard hold, and dial tone is heard]
3. Dial the third party. [Ringback tone]
4. If announcement is necessary, wait for the third party to answer and announce the call.
5. If no announcement is necessary, wait to hear ringback tone.
6. Press **[TRANSFER]** . [Call is transferred to the third party.]
7. Go on-hook.

Transferring a Call on a Terminal With a TRANSFER button (RTL call):

1. After receiving the calling party's instructions, press **[TRANSFER]** or **[CONFERENCE]** . * [Dial tone] (Calling party is placed on hold.)
2. Dial the requested number.

* For incoming RLT calls, the Conference button works the same as the Transfer button. That is, pressing the Conference button transfers an incoming RLT call back to the originally called branch location. Pressing the Conference button **does not** set up a 3-party conference.

3. If ringback (or Call Waiting ringback) is heard, press **[DISCONNECT]** , **[RELEASE]** ,

   or

   Go on-hook. (The RLT is released.)

   NOTE: If the called party does not answer before the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main.

   or

   If busy tone is heard, press **[TRANSFER]** or **[CONFERENCE]** (Voice terminal user is reconnected to the calling party.) and inform the calling party that the extension is busy.

   If the calling party does not want to wait for the called party to answer, press **[DISCONNECT]** , **[RELEASE]** ,

   or

   Go on-hook. (The RLT is released)

   or

   If the calling party wants to wait for the called party to answer, press **[TRANSFER]** or **[CONFERENCE]** . [Dial tone] (Calling party is placed on hold at the main.)

   Dial the Remote Hold dial access code. [Confirmation tone] (Calling party is placed on hold at the branch.)

   Press **[DISCONNECT], [RELEASE]** ,

   or

   Go on-hook. [Dial tone] (The RLT is released)

   NOTE: If the calling party is a station at a branch location, Remote Hold will be denied.

   NOTE: When the timed reminder interval expires, the branch seizes an RLT and sends the call back to the main. The terminal user can attempt to complete the call again.

Using Meet-Me Transfer to Join Two Calls With a Multiappearance Terminal:

1. Be sure there is an active call on one appearance.
2. Receive another call on an idle appearance. [Voice terminal rings.]

3. Press the **[HOLD]** button to place the active call on hold. [Green status lamp flutters.]
4. Select the ringing appearance. [The new call is now active.]
5. Press the **[TRANSFER]** button for the newly active call. [An idle appearance is automatically selected.]
6. Select the held appearance. [The original call is again active.]
7. Press the **[TRANSFER]** button for this active call. [The two calls have been transferred to each other.]

Considerations

Transfer Restrictions

A single-appearance voice terminal can transfer one party to a third party. A multiappearance terminal can transfer either one party or two parties to a third party.

Unless Trunk-to-Trunk Transfer is also provided, an incoming or outgoing trunk call cannot be transferred to another trunk call.

System 85 or Generic 2 software does not allow Trunk-to-Trunk Transfer unless one of the trunks involved in the transfer is handling ***an incoming call*** to the switch. If both trunks involved in the transfer were outgoing trunks, neither CO (Central Office) would be responsible to return disconnect supervision to the switch. So, without this safeguard, the switch would have no way of knowing when the transferred call was finished, and could not break down the connection. Therefore, when a voice terminal user desires to connect two parties using two outgoing trunks, the user should establish a 3-party conference and stay on the connection to maintain local supervision of the call.

Hard and Soft Processor Swaps

A stable transferred call will endure a hard processor swap.

During a hard swap, a voice terminal user cannot transfer the active call.

The Transfer feature operates normally during a soft processor swap.

Voice Terminals That Cannot Be Used to Transfer RLT Calls

Voice terminals that are not equipped with a Conference or Transfer button cannot be used to transfer an incoming RLT call from the CAS main to a branch location.

Interactions With Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

Attendant Call Waiting is denied toward an extension with the Transfer feature activated.

ACD (Automatic Call Distribution)

When an ACD agent transfers a call to another local agent, the transferred call is not considered work-related activity for the second agent. The second agent is not removed from the agent queue.

While an observer (using agent override) is connected to an agent's call, the Transfer feature is denied for use by the agent.

Bridged Call

The TRANSFER button is inoperable during a bridged appearance call for both the controlling voice terminal (extension that originated the bridged call) and the bridged voice terminal.

Call Coverage

The Transfer feature is denied from a multiappearance voice terminal during the Caller Response Internal of call coverage. The switch ignores the button press.

After a call goes to coverage, when the covering user presses the TRANSFER button, the temporary bridge appearance is removed from the principal's voice terminal.

When a local voice terminal user transfers an incoming trunk call to a local extension where dual coverage paths apply, the Call Coverage feature redirects the incoming trunk call according to the assigned criteria and path for ***internal*** calls.

CDR (Call Detail Recording)

The last voice terminal on a call is the number recorded by CDR.

Call Vectoring

Voice terminal users (including ACD agents) can transfer calls to a VDN. The transferred-to vector controls call processing for the transferred call. For example, the call could enter an ACD split's queue (including an AUDIX or Message Center queue) or the attendant queue and be processed according to the vector's programming.

Call Waiting

An analog voice terminal user attempting to activate Call Waiting during a Transfer operation toward a busy, single appearance voice terminal is denied Call Waiting if a call is being held in the *soft hold* state (on the calling terminal). Call Waiting is denied while soft hold is active, however, Call Waiting can be used if hard hold is active.

Centralized Attendant Service

The following CAS interaction applies to Issue 3.0 or later of DEFINITY Communications system, Generic 2.1.

For RLTs that terminate to ACD splits or VDNs, the answering positions should have display capable voice terminals so that the user can distinguish between RLT and non-RLT calls. This is because the user operations for the Transfer feature are different for RLT calls. The Transfer feature works normally for non-RLT calls. For more information, refer to the User Operations section of this feature description.

Data Call Setup

The Transfer feature does not allow the preindication function of the Data Call Setup feature. Consequently, a call requiring preindication (Modem Pooling or DS1 Interface features) fails if Transfer is used.

DCS (Distributed Communications System)

In a DCS environment without an Applications Processor, after an internode call transfers, the switch denies activation of Leave Word Calling.

EUCD (Enhanced Uniform Call Distribution)

When an EUCD agent transfers an EUCD call to another agent, the transferred call is not considered work-related activity for the second agent. The 106B status indicator shows the second agent as engaged in non-EUCD activity.

While an observer (using agent override) is connected to an EUCD agent's call, the Transfer feature is denied for use by the agent.

IPA (Interpartition Access)

A voice terminal user (in an extension partition other than Extension Partition 0) can transfer calls to extension numbers within the same partition group or to extensions in Extension Partition 0. When the user tries to transfer a call to any other partition group (using an extension number), the switch returns intercept treatment to the transferring party.

A voice terminal user in Extension Partition 0 is allowed to transfer calls (using an extension number) to any voice terminal in the switch.

Look-Ahead Interflow

At a receiving switch, the Look-Ahead Interflow feature and the Transfer feature are compatible. The answering voice terminal user at the receiving switch can normally transfer a Look-Ahead Interflow call to a third party inside or outside (if Trunk-to-Trunk Transfer is assigned) the receiving switch.

Loudspeaker Paging Access

A single-appearance voice terminal user in a 2-party talking connection cannot access Loudspeaker Paging unless an answer-back channel is available.

LWC (Leave Word Calling)

In a DCS environment without APs, after an internode call in transferred, activation of Leave Word Calling is denied.

Music-on-Hold Access

When Music-on-Hold Access is provided, if a multiappearance terminal is transferring a single party, the party receives music after being put on hold. However, if transferring two parties, the parties do not receive music when on hold. Instead, the two held parties maintain a talking connection.

Personal Central Office Line

When a call on a Personal Central Office Line is transferred to another party, the owner of the personal line temporarily loses control of the personal line. The owner cannot place or receive a call on that line until the Transfer call is finished. This can be avoided by setting up a 3-party conference and then placing the conference on hold. Then, at any time, the owner can reenter the connection and request use of the personal line.

Priority Calling

Priority Calling is denied toward a single-appearance terminal with a Transfer in progress. Priority Calling is also denied if the calling party (party attempting to transfer a call) has a call (the call to be transferred) held in the *soft* hold state.

When the Priority Calling feature is activated toward a multiappearance terminal with a Transfer in progress, the call completes to an idle appearance. If no appearance is idle, the priority call is denied.

Remote Access

A remote access call can be transferred by a local System 85 or Generic 2 station user. However, the remote access caller cannot use the Transfer feature.

Restriction—Miscellaneous Trunk Restrictions

A terminal restricted from accessing a trunk group, due to the Miscellaneous Trunk Restriction feature, can be transferred via the Transfer feature to a restricted trunk group. However, the restricted terminal cannot originate a transfer via the Transfer feature to a restricted trunk group.

Restriction—Voice Terminal Restrictions

The Transfer feature may transfer an incoming trunk call to an Inward restricted voice terminal.

An Outward restricted voice terminal can access a public network trunk if the restricted voice terminal calls an unrestricted voice terminal. The unrestricted voice terminal uses the Transfer feature to connect the restricted voice terminal to the public network trunk.

An origination-restricted single-appearance voice terminal may originate a call from a 2-party connection by first accessing the Transfer feature.

A voice terminal with the Terminal-to-Terminal Only Calling restriction activated can be transferred to the attendant by another voice terminal user via the Transfer feature. A Terminal-to-Terminal Only Calling restricted voice terminal may transfer a call to another voice terminal, but not to the attendant.

Ringing—Distinctive

When a local voice terminal user transfers an incoming trunk call to another local extension, the switch provides 1-burst ringing for the transferred-to voice terminal.

Serial Calls

When Serial Calls is in effect, pressing RECALL button at a local terminal recalls the attendant. Therefore, a terminal user cannot access the Transfer feature during a serial call.

Tenant Services

A voice terminal user (in an extension partition other than Extension Partition 0) can transfer calls to extension numbers within the same partition or to extensions in Extension Partition 0. When the user tries to transfer a call to any other extension partition (using an extension number), the switch returns intercept treatment to the transferring party.

A voice terminal user in Extension Partition 0 is allowed to transfer calls (using an extension number) to any voice terminal in the switch.

Unattended Console Service—CAAVT (Call Answer From Any Voice Terminal) and Preselected Call Routing

A single-appearance voice terminal user accessing the CAAVT feature or the preselected terminal when Preselected Call Routing is active has access to the Transfer feature even though Transfer might not be assigned to the callers extension class of service. The CAAVT user or preselected terminal may direct a call to another local terminal via Transfer. When directing a call to a trunk, the CAAVT user or preselected terminal must use a terminal with Transfer assigned. All multiappearance voice terminals have Transfer Capabilities.

Hardware Requirements

None.

Feature Administration

Assignment of the Transfer feature is on a extension class of service basis for single-appearance voice terminals. Multiappearance voice terminals are always provided with the feature.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — TRANSFER			
PROCEDURE	WORD	PURPOSE	SMT
000	1	Assigns the extension class of service to an extension number.	Yes
010	1	Assigns the Transfer feature to an extension class of service.	Yes
275	4	Assigns Trunk-to-Trunk Transfer to the system class of service.	Yes

The following are the applicable TCM path names used with the AP 16.

TCM SCREENS — TRANSFER	
PATH NAME	PURPOSE
terminal-change class-of-service attributes	Assigns the Conference—Three Party and Transfer features to an extension class of service.
terminal-change extensions attributes	Assigns the class of service to an extension number.

Notes:

Trunk Group Busy/Warning Indicators to Attendant

Description

This feature provides the attendant with a visual warning, indicating when the number of available trunks in a group diminishes to a preset level. A visual indication is also provided when every trunk in a group is busy.

The Trunk Group Busy and the Trunk Group Warning Indicators are useful when the Attendant Control of Trunk Group Access feature is provided. The warning indicators show the attendant when control of access to trunk groups may be necessary. The busy indicators show the attendant when a trunk in a busy group becomes available for a waiting voice terminal.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

None.

Considerations

Limits

A total of 24 busy trunk-group indicators and 12 warning indicators is provided. A high number of busy or warning indications can indicate a need to rearrange or add trunks.

Dial Access Restriction

Although dial access restriction (assigned in Procedure 100, Word 1) prevents an attendant from using DTGS (Direct Trunk Group Selection) to access a restricted trunk group, busy / warning indication can still be applied to a dial-access-restricted trunk group so that the attendant(s) can monitor the calling activity on the trunk group.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Conference—Attendant Five Party

The trunk-group warning indicator lamp lights when five conferees are connected to the conference. The trunk-group warning indicator lamp goes out when the attendant releases from the conference. The trunk-group warning indicator lamp lights only on the console that is controlling the conference.

Conference—Attendant Six Party

When the Conference—Attendant Six Party feature is provided, the associated trunk-group warning indicator is lighted when six conferees are connected to the conference circuit. The trunk-group warning indicator extinguishes when less than six conferees connect to a conference circuit and when the attendant releases from the conference. The trunk group warning indicator lights only on the console to which the conference circuit is connected.

Tenant Services

A partitioned switch only updates the lamps associated with a DTGS button for the attendant consoles that are allowed to access that specific trunk group.

Hardware Requirements

None.

Feature Administration

Assignment of the Trunk Group Busy/Warning Indicators to Attendant feature is on a per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES TRUNK GROUP BUSY/WARNING INDICATORS TO ATTENDANT		
PROCEDURE	WORD	PURPOSE
200	1	Assigns Direct Trunk Group Selection to the attendant console(s)
202	1	Assigns busy/warning indication to the attendant console(s). Sets the warning level in the adjacent field.

Trunk Verification — Attendant

Description

This feature gives an attendant the ability to test the operation of individual trunks. The attendant can identify defective trunks and report their condition for servicing. This results in better overall communications.

The types of trunks that can be verified are listed in this feature's "Considerations" section.

If the trunk is busy, the talking parties hear a warning tone before the attendant enters the call. The warning tone is a 440-hertz tone applied at 15-second intervals. The duration of the first burst of tone is 2 seconds. Thereafter, the duration is a ½ second.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Test the Condition of a Trunk

For an idle nontie trunk:

1. Press an idle loop button. [ATND lamp lights, and PA lamp goes out.]
2. Press **[VERFY]** . [VERFY lamp lights.]
3. Press **[START]** or press a Direct Trunk Group Selection button and skip the next step. [Dial tone]
4. Dial the trunk-group access code.
5. Dial the trunk number (for example, "028"). [Trunk seized, VERFY lamp goes out ANS lamp lights, and Dial tone heard.]

   NOTE: Using an R2 V1 switch, only two digits should be dialed for trunk numbers (for example, "06," or "99"). This is because R2 V1 switches cannot have more than 99 trunks per trunk group.

6. Dial distant voice terminal extension number. [Ringback tone]
7. Distant voice terminal user answers. (The connection is verified and evaluated.)
8. After verification, press **[RELEASE]** . [ANS and ATND lamp go out, console is released from all trunks and tones, and PA lamp lights.]

For a busy nontie trunk:

1. Press an idle loop button. [ATND lamp lights, and PA lamp goes out.]
2. Press **[VERFY]** . [VERFY lamp lights.]
3. Press **[START]** or press a Direct Trunk Group selection button and skip the next step. [Dial tone]
4. Dial the trunk-group access code.
5. Dial the trunk number (for example, "255"). [Trunk is seized and warning tone is heard by both parties on the active call and the attendant. Attendant is bridged onto the call, VERFY lamp goes out, and ANS lamp lights.] (The connection is verified and evaluated.)
6. After verification, press **[RELEASE]** . [ANS and ATND lamp go out, console is released from all trunks and tones, and PA lamp lights.]

For a nonautomatic idle tie trunk:

1. Press an idle loop button. [ATND lamp lights, and PA lamp goes out.]
2. Press **[VERFY]** . [VERFY lamp lights.]
3. Press **[START]** or press a Direct Trunk Group Selection button and skip the next step. [Dial tone]
4. Dial the trunk-group access code.
5. Dial the trunk number (for example, "007"). [Trunk seized, VERFY lamp goes out, RING lamp lights, and dial tone heard.]
6. Dial distant voice terminal extension number. [Ringback tone]
7. Distant voice terminal user answers. [RING lamp goes out.] (The connection is verified and evaluated.)
8. After verification, press **[RELEASE]** . [ATND lamp goes out, console is released from all trunks and tones, and PA lamp lights.]

For a busy tie trunk:

1. Press an idle loop button. [ATND lamp lights, and PA lamp goes out.]
2. Press **[VERFY]** . [VERFY lamp lights.]
3. Press **[START]** or press a Direct Trunk Group Selection button and skip the next step. [Dial tone]
4. Dial the trunk-group access code.
5. Dial the trunk number. [Trunk is seized and warning tone is heard by both parties on the active call and the attendant. Attendant is bridged onto the call and VERFY lamp goes out.] (The connection is verified and evaluated.)
6. After verification, press **[RELEASE]** . [ANS and ATND lamp go out, console is released from all trunks and tones, and PA lamp lights.]

For an automatic idle tie trunk:

1. Press an idle loop button. [ATND lamp lights, and PA lamp goes out.]
2. Press **[VERFY]** . [VERFY lamp lights.]
3. Press **[START]** or press a Direct Trunk Group Selection button and skip the next step. [Dial tone]
4. Dial the trunk-group access code.
5. Dial the trunk number. [Trunk seized, VERFY lamp goes out, RING lamp lights, and ringback tone is heard.]
6. Distant voice terminal user answers. [RING lamp goes out.] (The connection is verified and evaluated.)
7. After verification, press **[RELEASE]** . [ATND lamp goes out, console is released from all trunks and tones, and PA lamp lights.]

For a Busy Trunk at a Distant DCS Node:

1. Press an idle loop button. [ATND lamp lights, and PA lamp goes out]
2. Press **[VERFY]** . [VERFY lamp lights.]
3. Press **[START]** or press DTGS button to reach the distant node and skip the next step. [Dial tone]
4. Dial the access code of the tie trunk group to reach the distant node.
5. Press **[VERFY]** again. [VERFY lamp goes out.]
6. Dial the access code of the trunk group to be verified at the distant node.
7. Dial the trunk number (for example, "173"). [Trunk is seized and warning tone is heard by both parties on the active call and the attendant. Attendant enters the call.] (The connection is verified and evaluated.)
8. After verification, press **[RELEASE]** . [Console is released from the trunk and tones, and PA lamp lights.]

Considerations

The following list identifies the Procedure 100, Word 1 trunk types and whether they can be tested.

Trunk Types and Whether They Can Be Tested		
Code	**Test**	**Trunk Type**
		Special trunks:
2	No	Touch-tone register
5	No	Conference—Six Party
6	No	Special queue
		All 2-way Advanced Private Line Termination (APLT) trunks:
12	Yes	Delay dial in/(wink start or delay dial) and dial tone out
13	Yes	Wink start in/(wink start or delay dial) and dial tone out
14	Yes	Delay dial in/dial tone out
15	Yes	Wink start in/dial tone out
		Regular Central Office (CO) trunks:
16	Yes, if busy	1-way automatic incoming attendant-completing
17	Yes	1-way outgoing DOD
18	Yes	1-way out DOD with party test
19	Yes	2-way automatic incoming attendant-completing/DOD
20	Yes	2-way with party test
		Foreign Exchange (FX) trunks:
21	Yes, if busy	1-way automatic incoming attendant-completing
22	Yes	1-way outgoing DOD
23	Yes	1-way out DOD with party test
24	Yes	2-way automatic attendant-completing in/DOD
25	Yes	2-way with party test
		Wide Area Telecommunications System (WATS) trunks:
26	Yes, if busy	1-way automatic incoming attendant-completing
27	Yes	1-way outgoing DOD or toll terminal access for TSPS
28	Yes	1-way out DOD with party test
		Direct Inward Dialing (DID) trunks:
30	Yes	Immediate start DID
31	Yes	Wink start DID
		Tie trunks:
32	Yes, if busy	1-way in dial repeating
33	Yes	1-way out automatic
34	Yes	1-way dial out repeating
35	Yes, if busy	1-way in automatic
36	Yes	2-way dial repeating both ways
37	Yes	2-way dial repeating in/automatic out
38	Yes	2-way automatic in/dial repeating out
39	Yes	2-way automatic both ways
40	Yes, if busy	1-way in, dial repeating, and delay dial
41	Yes	2-way wink start in/delay dial, wink start out
42	Yes, if busy	1-way in, wink start
43	Yes	1-way out, delay dial, wink start
44	Yes	2-way dial repeating, delay dial in
45	Yes	2-way dial repeating, delay dial in/automatic out

Trunk Types and Whether They Can Be Tested (Contd)		
Code	**Test**	**Trunk Type**
		Tie trunks (Contd)
46	Yes	2-way dial repeating in/delay dial, or wink start out
47	Yes	2-way dial repeating delay dial in/delay dial, or wink start out
		Special trunks and interfaces:
50	Yes	Remote access; 2-way dial tone in/ground start and dial tone out
51	No	Telephone Dictation Interface
52	No	Recorded Announcement Interface
53	No	Code Calling Interface
54	No	Loudspeaker Paging Interface
55	No	Touch-tone sender
57	No	CAS release link trunk-outgoing from branch; 1-way automatic out
58	No	ANI Interface
62	No	Music-on-Hold Interface
65	No	SN241 Contact Interface
66	No	CAS release link trunk-incoming at main; 1-way automatic in
67	No	Audio
68	No	UCD/DDC delay recorded announcement trunk (R2 V1 only)
		Special tie trunks (Main/ Satellite)
70	Yes, if busy	1-way in immediate start
71	No	1-way out immediate start
72	Yes	2-way immediate start both ways
73	Yes, if busy	1-way in wink start
74	No	1-way out wink start
75	Yes	2-way wink start both ways
76	Yes, if busy	1-way in delay dial
77	No	1-way out delay dial
78	Yes	2-way delay dial both ways
		Special trunks:
90	No	ACD/EUCD first announcement/Vectoring recorded announcement
91	No	ACD/EUCD second announcement
92	No	ACD/EUCD origin announcement
93	No	Malicious call trace recorder
		Data Trunk types:
100	No	Tone detector
101	No	Analog data modem pool
102	No	Digital data modem pool
103	No	Host access PDM
104	No	Host access TDM

Trunk Types and Whether They Can Be Tested (Contd)		
Code	**Test**	**Trunk Type**
		Data Trunk types (Contd):
105	No	AP32 DCPI
106	No	EIA 4 port
107	No	ISN/EIA port
108	No	DMI wink in/automatic out
109	No	DMI wink in/wink out
120	Maybe	ISDN dynamic
		Special Loop Start Central Office (CO) trunks:
316	No	1-way automatic in
317	No	1-way outgoing DOD
319	No	2-way automatic in/DOD
		Special Loop Start FX trunks:
321	No	1-way automatic in
322	No	1-way outgoing DOD
324	No	2-way automatic in/DOD
		Special Loop Start WATS trunks:
326	No	1-way automatic in
327	No	1-way outgoing DOD/or toll terminal access for TSPS
		Special Loop Start Remote Access trunks:
350	No	Remote Access

Maintenance Busy Trunks

The attendant cannot take a defective trunk out of service (maintenance busy-out). However, a designated voice terminal using the Trunk Verification—Voice Terminal feature can be used for this purpose. Also a trunk that has been maintenance busied-out can't be verified.

Verify Button and Lamp

When the switch is equipped with both the Trunk Verification by Attendant and the Busy Verification of Lines features, the same VERFY button and lamp are used for both features.

Reorder Tone

If the attendant receives reorder tone and the BUSY lamp lights, this indicates that the trunk is:

- Connected to a line that has data protection active
- Not in a stable state within the switch
- Not in a 2-party connection
- Connected to an attendant console (unless the calling party is a remote access trunk)

- Being held in a Call Park state.

The attendant should wait a few seconds and retry the check.

Intercept Tone

The switch returns intercept tone when the attendant tries to access an invalid dial access code or test an invalid trunk type. The VERFY lamp also extinguishes after attempting to test an invalid trunk type. The switch returns intercept tone when attempting to access an idle 1-way incoming trunk.

Dial Access Restriction

Dial Access Restriction (assigned to a trunk group in Procedure 100, Word 1) prevents attendants, voice terminal users, and data terminal users from directly accessing a trunk group by dialing the trunk-group access code. However, dial access restriction ***does not*** prevent an attendant from accessing a restricted trunk group to verify a trunk.

Hard and Soft Processor Swaps

If an attendant is verifying a stable trunk connection when a hard processor swap occurs, this 3-party connection will endure the hard swap.

An attendant cannot verify a trunk during a hard processor swap.

The Trunk Verification—Attendant feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Call Waiting

If the trunk is being held or answered by a terminal using the answer-hold code of Attendant Call Waiting, TVA (Trunk Verification—Attendant) is denied.

ACA (Automatic Circuit Assurance)

A distant trunk which has an ACA record associated will be updated.

Call Park

A trunk that is in call park cannot be busy verified using TVA.

Call Waiting

If the trunk is being held or answered by a terminal using the answer-hold code of Call Waiting, TVA is denied.

Conference—Attendant Five Party

Trunk Verification may not be used on a trunk that is involved in an attendant conference connection.

Conference—Attendant Six Party

Trunk Verification may not be used toward a trunk that is involved in an attendant conference connection.

Conference—Three Party

TVA is denied when attempted toward a conference call established by the Conference—Three Party feature.

Data Protection

If the trunk under test is data protected or connected to a terminal which is data protected, TVA is denied.

DCS (Distributed Communications System)

In a DCS environment, an attendant at one node can verify a trunk in another node. The procedures for verifying distant trunks are slightly different than the usual verification procedures (see the "User Operations" of this feature description.)

Hold

If the trunk to be verified is being held by the Hold feature, TVA is denied.

ISDN (Integrated Services Digital Network)/PRI (Primary Rate Interface)

The trunk verification features work in the same way for ISDN trunks as they do for other types of trunks, except for the ISDN Dynamic trunk types. These trunks reflect a default trunk type based on the far end of the trunk connection.

Malicious Call Trace

The Trunk Verification—Attendant can assist in tracing malicious calls that originate from or tandem through distant switches in the private network. After an attendant at the distant switch is called by the controlling attendant at the local switch, the distant attendant can enter the call using the Trunk Verification—Attendant feature and then activate Malicious Call Trace at the distant end.

> **NOTE:** Since the warning tone provided by trunk verification could arouse suspicion by the malicious caller, the verification should be deactivated as quickly as possible.

Modem Pooling

The Trunk Verification—Attendant feature cannot be used to test modem pooling trunk types (types 101 and 102). The Trunk Verification—Voice Terminal feature can, however, be used for this purpose.

Override

If a 3-way connection has been established using the Override feature, TVA is denied.

Priority Calling

If the trunk is being held or answered by a terminal using the answer-hold code of Priority calling, TVA is denied.

Serial Calls

TVA is denied toward a trunk involved in a serial call.

Tenant Services

An attendant (in a partition other than Attendant Partition 0) is allowed to verify trunks within trunk groups that are assigned to that attendant's partition. When this is done, the feature operates normally. However, when an attendant attempts to verify a trunk that is not within a trunk group assigned to the partition, the switch will return intercept treatment to the attendant.

An attendant in Attendant Partition 0 is allowed to verify any trunk that can otherwise be verified using this feature.

Trunk Verification—Voice Terminal

During a 3-way connection established using this feature, TVA is denied.

Hardware Requirements

None.

Feature Administration

Assignment of the Trunk Verification by Attendant feature is on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be adminstered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — TRUNK VERIFICATION—AITENDANT			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
103	1	Assigns bridge-on capability to a trunk group for use in trunk verification.	Yes
177	1	Prior to R2 V4, displays the equipment location of a trunk within a trunk group (by dial access code).	No
178	1	Beginning with R2 V4, displays the equipment location of a trunk within a trunk group (by dial access code or by trunk-group number).	No
200	1	Enables the Trunk Verification by Attendant feature.	No
203	1	Assigns the VERFY button to the attendant console(s). The applicable encode is as follows: 7 VERFY Button.	No

Trunk Verification — Voice Terminal

Description

This feature gives the user of a designated extension the ability to test the operation of individual trunks. The user can identify and remove defective trunks from service. The defective trunk can be repaired and returned to service quickly. As a result, better communications service is provided.

The types of trunks that can be verified are listed in this feature's "Considerations" section.

If a trunk is busy, the talking parties hear a waning tone before the testing party is added to the connection. The warning tone is a 440-hertz tone applied at 15-second intervals. The first burst of tone lasts for 2 seconds. Subsequent bursts of tone last for ½ second.

Trunk verification from a voice terminal can be performed at a remote switch using a dial repeating tie trunk.

Feature History and Development

This feature was first available for System 85 in Release 1. Subsequent enhancements include the following:

- Two-way tie trunks can be busied out. First available in Release 2, Version 3.

User Operations

The following are the user operating procedures for this feature.

To Activate Trunk Verification

For a local nonautomatic trunk that is idle:

1. Go off-hook. [Dial tone]
2. Dial the trunk test access code. [Second dial tone]
3. Dial the trunk-group access code.
4. Dial the trunk number (for example, "025"). [Trunk seized, and dial tone is heard.]

    NOTE: Using an R2 V1 switch, only two digits should be dialed for trunk numbers (for example, "37," or "99"). This is because R2 V1 switches cannot have more than 99 trunks per trunk group.

5. Dial a voice terminal number. [Ringback tone]

6. The called party answers. (The connection is verified and evaluated.)
7. After verification, go on-hook.

For a local automatic trunk that is idle:

1. Go off-hook. [Dial tone]
2. Dial the trunk test access code. [Second dial tone]
3. Dial the trunk-group access code.
4. Dial the trunk number (such as, "255"). [Trunk seized, and ringback tone is heard.]
5. When the attendant answers, ask the attendant to dial a voice terminal number. [Ringback tone]
6. The called party answers. (The connection is verified and evaluated.)
7. After verification, go on-hook.

For a local trunk that is busy:

1. Go off-hook. [Dial tone]
2. Dial the trunk test access code. [Second dial tone]
3. Dial the trunk-group access code.
4. Dial the trunk number (such as, "004"). [Trunk is seized. Warning tone is heard by both parties on the active call and the testing voice terminal. Voice terminal is bridged onto the call.] (The connection is verified and evaluated.)
5. After verification, go on-hook.

For a local Modem Pooling conversion resource that is idle:

— From the voice terminal of a DCP voice/data station:

1. Go off-hook. [Dial tone]
2. Dial the trunk test access code. [Second dial tone]
3. Dial the access code of the digital Modem Pooling trunk group. [Third dial tone]
4. Dial the trunk number of the Modem Pooling member. [Fourth dial tone]
5. Dial the number of an off-premises data end point (such as a host computer). [Ringback, followed by ready tone.]
6. Press the **[DATA]** button on your digital voice terminal. [Login prompt from host computer]
7. Go on-hook.
8. After verification, press the **[DISCONNECT]** button on the data module,

   or

Press **[BREAK]** on the data terminal's keyboard.

— From the voice terminal of a BRI voice / data station with a data line appearance of the designated extension.:

1. Without going off-hook, press the **[Data/Send/Off]** button. [The red in-use lamp lights on the **[Data/Send/Off]** button. [DIAL:] appears on the display.]
2. Dial the trunk test dial access code of the ***digital member*** of the modem pooling trunk group pair. [[DIAL:] appears on the display.]
3. Dial the trunk number of the modem pool member. [[DIAL:] appears on the display.]
4. Dial the number of an off-premises data end point (such as a host computer). [[CALLING:] appears on the display.]
5. Press the [Data / Send / Off] button again. [The red in-use lamp stays on and the green status lamp blinks. When the far end returns answer tone, the green status lamp goes to a steady on state and the red in-use lamp remains in the steady on state. The data terminal screen displays:

   [CONNECTED - MODE 2] and [FAR END SPEED - 19200] .]

   NOTE: The line appearance seized may not be the one assigned to the data button. Also, the figures for data mode (mode 2) and data rate (19200) shown in these examples are used for example purposes only. The mode and rate that will appear on your display will reflect the actual state of the far end of your connection and may differ from the examples shown.

6. Press **[RETURN]** on the data terminal keyboard and proceed with appropriate log on procedures.
7. After verification, press the **[Data/Send/Off]** button on the BRI voice terminal to end the data call.

For a nonautomatic tie trunk at a remote switch that is idle:

1. Go off-hook. [Dial tone]
2. Dial the access code of the tie trunk group. [Second dial tone]
3. Dial the trunk test access code. [Third dial tone]
4. Dial the trunk-group access code.
5. Dial the trunk number. [Trunk is seized, and dial tone is heard.]
6. Dial a voice terminal number. [Ringback tone]
7. The called party answers. (The connection is verified and evaluated.)
8. After verification, go on-hook.

For an automatic tie trunk at a remote switch that is idle:

1. Go off-hook. [Dial tone]
2. Dial the access code of the tie trunk group. [Second dial tone]
3. Dial the trunk test access code. [Third dial tone]
4. Dial the trunk-group access code.
5. Dial the trunk number. [Trunk is seized, and ringback tone is heard.]
6. When the attendant answers, ask the attendant to dial the voice terminal number. [Ringback tone]
7. The called party answers. (The connection is verified and evaluated.)
8. After verification, go on-hook.

To Remove (Maintenance Busy) a Defective Trunk From Service:

1. Go off-hook. [Dial tone]
2. Dial the maintenance busy access code. [Second dial tone]
3. Dial the trunk-group access code.
4. Dial the trunk number. (If there is an active call on the trunk, it is disconnected [Confirmation tone]
5. Go on-hook

To Make (Maintenance Unbusy) a Trunk Available:

1. Go off-hook. [Dial tone]
2. Dial the maintenance unbusy access code. [Second dial tone]
3. Dial the trunk-group access code.
4. Dial the trunk number. [Confirmation tone]
5. Go on-hook.

Considerations

The following list identifies the Procedure 100, Word 1 trunk types and whether they can be tested.

Trunk Types and Whether They Can Be Tested		
Code	**Test**	**Trunk Type**
		Special trunks:
2	No	Touch-tone register
5	No	Conference—Six Party
6	No	Special queue
		All 2-way Advanced Private Line Termination (APLT) trunks:
12	Yes	Delay dial in/(wink start or delay dial) and dial tone out
13	Yes	Wink start in/(wink start or delay dial) and dial tone out
14	Yes	Delay dial in/dial tone out
15	Yes	Wink start in/dial tone out
		Regular Central Office (CO) trunks:
16	No	1-way automatic incoming attendant-completing
17	Yes	1-way outgoing DOD
18	Yes	1-way out DOD with party test
19	Yes	2-way automatic incoming attendant-completing/DOD
20	Yes	2-way with party test
		Foreign Exchange (FX) trunk:
21	No	1-way automatic incoming attendant-completing
22	Yes	1-way outgoing DOD
23	Yes	1-way out DOD with party test
24	Yes	2-way automatic attendant-completing in/DOD
25	Yes	2-way with party test
		Wide Area Telecommunications System (WATS) trunks:
26	No	1-way automatic incoming attendant-completing
27	Yes	1-way outgoing DOD or toll terminal access for TSPS
28	Yes	1-way out DOD with party test
		Direct Inward Dialing (DID) trunks:
30	Yes	Immediate start DID
31	Yes	Wink start DID

Trunk Type and Whether They Can Be Tested (Contd)

Code	Test	Trunk Type
		Tie trunks:
32	No	1-way in dial repeating
33	Yes	1-way out automatic
34	Yes	1-way dial out repeating
35	No	1-way in automatic
36	Yes	2-way dial repeating both ways
37	Yes	2-way dial repeating in / automatic out
38	Yes	2-way automatic in / dial repeating out
39	Yes	2-way automatic both ways
40	No	1-way in, dial repeating and delay dial
41	Yes	2-way wink start in / delay dial, wink start out
42	No	1-way in, wink start
43	Yes	1-way out, delay dial, wink start
44	Yes	2-way dial repeating, delay dial in
45	Yes	2-way dial repeating, delay dial in / automatic out
46	Yes	2-way dial repeating in / delay dial, or wink start out
47	Yes	2-way dial repeating delay dial in / delay dial, or wink start out
		Special trunks and interfaces:
50	Yes	Remote access; 2-way dial tone in / ground start and dial tone out
51	No	Telephone Dictation Interface
52	No	Recoded Announcement Interface
53	No	Code Calling Interface
54	No	Loudspeaker Paging Interface
55	No	Touch-tone sender
57	No	CAS release link trunk-outgoing from branch; 1-way automatic out
58	No	ANI Interface
62	No	Music-on-hold Interface
65	No	SN241 Contact Interface
66	No	CAS release link trunk-incoming at main; 1-way automatic in
67	No	Audio
68	No	UCD / DDC delay recorded announcement trunk (R2 VI only)
		Special tie trunks (Main/ Satellite):
70	No	1-way in immediate start
71	No	1-way out immediate start
72	Yes	2-way immediate start both ways
73	No	1-way in wink start
74	No	1-way out wink start
75	Yes	2-way wink start both ways
76	No	1-way in delay dial
77	No	1-way out delay dial
78	Yes	2-way delay dial both ways

Trunk Types and Whether They Can Be Tested (Contd)		
Code	**Test**	**Trunk Type**
		Special trunks:
90	No	ACD/EUCD first announcement/Vectoring recorded announcement
91	No	ACD/EUCD second announcement
92	No	ACD/EUCD origin announcement
93	No	Malicious call trace recorder
		Data Trunk types:
100	No	Tone detector
101	Yes*	Analog data modem pool
102	Yes*	Digital data modem pool
103	No	Host access PDM
104	No	Host access TDM
105	No	AP32 DCPI
106	No	EIA 4 port
107	No	ISN/EIA port
108	No	DMI wink in/automatic out
109	No	DMI wink in/wink out
120	Maybe	ISDN dynamic
		Special Loop Start Central Office (CO) trunks:
316	No	1-way automatic in
317	No	1-way outgoing DOD
319	No	2-way automatic in/DOD
		Special Loop Start FX trunks:
321	No	1-way automatic in
322	No	1-way outgoing DOD
324	No	2-way automatic in/DOD
		Special Loop Start WATS trunks:
326	No	1-way automatic in
327	No	1-way outgoing DOD/or toll terminal access for TSPS
		Special Loop Start Remote Access trunks:
350	No	Remote Access

* Can be tested (by special procedures) begninning with R2 V3 switches. Cannot be tested on earlier switches.

Testing Terminal Types

The following terminals can use this feature:

- A terminal with an appearance of the designated extension
- A remote maintenance terminal
- A backup control voice terminal associated with CAS (Centralized Attendant Service).

Idle 1-way Incoming Trunk

An idle 1-way incoming trunk cannot be seized and tested. The trunk must be busy, and bridge-on must be allowed for the trunk group.

Reorder Tone

Reorder tone is heard if the trunk is in an unstable, transient condition. The user should wait a few seconds and retry the check.

Busy Tone

Busy tone is heard for the following reasons:

- The trunk under test belongs to a trunk group which has bridge-on restriction.
- The trunk under test is setting up a call.
- The trunk under test has a 3-way call.
- The trunk under test is a busy tie trunk.
- An RLT under test is busy.
- The call on the trunk under test is data protected.

  The call on the trunk under test is a trunk-to-trunk call. This feature cannot test a trunk-to-trunk connection.

Intercept Tone

Intercept tone is heard if trunk verification is not performed from an image of the designated extension, if an invalid trunk-group access code is dialed, or the trunk is not of a type that can be put on maintenance busy.

Dial Access Restriction

Dial Access Restriction (assigned to a trunk group in Procedure 100, Word 1) prevents voice terminal users, data terminal users, and attendants from directly accessing a trunk group by dialing the trunk-group access code. However, dial access restriction ***does not*** prevent the designated extension from accessing a resticted trunk group to verify a trunk.

Tandem Tie Trunk Switching

In order to test a remote tandem tie trunk, switching at a distant switch must be administered using Procedure 275, Word 1.

Busying Out Trunks

A trunk cannot be busied-out from another node.

If a 2-way trunk is busied-out, it has to be busied-out from both ends.

Hard and Soft Processor Swaps

If the user of the designated extension is verifying a stable trunk connection when a hard processor swap occurs, this 3-party connection will endure the hard swap.

The voice terminal user cannot verify a trunk during a hard processor swap.

The Trunk Verification—Voice Terminal feature operates normally during a soft processor swap.

Interactions With Other Features

The following System 85 and Generic 2 feature affect or are affected by the operation of this feature.

Attendant Call Waiting

If the trunk is being held or answered by a terminal using the answer-hold code of Attendant Call Waiting, TVVT (Trunk Verification—Voice Terminal) is denied.

Bridged Call

Any image (including image of shared appearances) of the extension designated for trunk verification, is the equivalent of the designated extension and can be used for the Trunk Verification Voice Terminal feature.

Call Park

A trunk that is in call park cannot be busy verified using the TVVT feature.

Call Vectoring

Entering a VDN in Procedure 285 as the designated internal extension or the remote maintenance terminal for Trunk Verification is denied. When this is attempted, an administration error will occur.

Call Waiting

If the trunk is being held or answered by a terminal using the answer-hold code of Call Waiting, TVVT is denied.

Conference—Attendant Five Party

Trunk Verification may not be used on a trunk that is involved in an attendant conference connection.

Conference—Attendant Six Party

Trunk Verification may not be used toward a trunk that is involved in an attendant conference connection.

Conference—Three Party

If a conference call has been established using Conference—Three Party, TVVT is denied.

Data Protection

If the trunk under test is data protected or connected to a terminal that is data protected, TVVT is denied.

Hold

If the trunk to be verified is being held by the Hold feature, TVVT is denied.

ISDN—BRI (Basic Rate Interface)

The trunk verification feature is compatible with ISDN—BRI terminals with the following exceptions and consideration.

- An appearance designated as a *data appearance* cannot be used for normal (voice call) trunk verification.
- A trunk that is connected to an active ISDN—BRI data call cannot be verified. These calls are provided data protection, whether or not they involve designated data appearances or the Data Protection feature is applied.

ISDN—PRI (Primary Rate Interface)

The trunk verification features work in the same way for ISDN trunks as they do for other types of trunks, except for the ISDN Dynamic trunk types. These trunks reflect a ***default*** trunk type based on the far end of the trunk connection.

Last Number Dialed

The Last Number Dialed feature does not store or redial trunk-test dial sequences. Once a trunk has been verified, redialing the digit sequence would not be useful.

Loudspeaker Paging Access

A Loudspeaker Paging trunk cannot be tested.

Modem Pooling

The Trunk Verification—Voice Terminal feature can be used to test Modem Pooling conversion resources on R2 V3 and later switches.

When keyboard dialing is used to place a data call, the System 85 or Generic 2 automatically reserves a Modem Pooling conversion resource and, if necessary, inserts the resource into the connection. As the switch reserves a resource, the ***digital*** Modem Pooling trunk-group number and the individual trunk number of the selected conversion resource are displayed in the RINGING call-progress message. In response to trouble reports, these values (after converting the trunk-group number to the trunk-group access code) can be used to verify the Modem Pooling conversion resource.

Override

If a 3-way connection has been established using the Override feature, TVVT is denied.

Priority Calling

If the trunk is being held or answered by a terminal using the answer-hold code of Priority Calling, TVVT is denied.

Serial Calls

TVVT is denied toward a trunk involved in a serial call.

Tenant Services

The Trunk Verification—Voice Terminal feature is unaffected in a partitioned switch. The user of either the internal or the remote designated extension can verify, maintenance busy, or maintenance unbusy any trunk that can otherwise be verified using this feature.

The designated terminal(s) can reside in any partition, but it is strongly recommended that these terminals be assigned to Extension Partition 0.

Trunk Verification—Attendant

During a 3-way connection established using this feature, TVVT is denied.

Hardware Requirements

The TELTONE M106-05 Remote Access Unit can be provided as an option to customers which have control over both ends of one or more 2-way tie trunks, such as with an Electronic Tandem Network (ETN). This allows the user to verify and/or maintenance busy/unbusy both ends from a central location.

One unit must be provided at each system requiring remote testing. The distant end of the trunk is tested or busied/unbusied by dialing up the remote access unit and patching through to the TVVT feature. The local end is tested or busied/unbusied normally.

Feature Administration

Assignment of the Trunk Verification by Voice Terminal feature is on a system class of service basis and within the system on an extension basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Communications System Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — TRUNK VERIFICATION—VOICE TERMINAL			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
100	1	Denies access to a trunk group (except by the designated extension) using the trunk-group dial access code.	No
103	1	Assigns bridge-on availability for testing to a trunk group for use with trunk verification (Field 7).	Yes
285	1	Assigns a designated internal extension and / or a remote maintenance terminal for trunk verification.	Yes
350	1	Assigns the first digit of feature dial access codes (if required).	No
350	2	Assigns the Trunk Test and Maintenance Busy dial access codes. The applicable feature encodes (field 1) are as follows: 42 Maintenance busy a trunk 43 Maintenance unbusy a trunk 44 Trunk test from a terminal.	No

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — TRUNK VERIFICATION—VOICE TERMINAL	
PATH NAME	**PURPOSE**
terminal-change system parameters (select the Access-Codes option)	Assigns a designated internal extension for trunk verification.

Trunk-to-Trunk Connections

Description

This feature allows an attendant to connect an incoming or outgoing trunk call to an outgoing trunk. The attendant console holds outgoing trunk-to-outgoing trunk connections in order to monitor the connection and release the trunk when the call is finished.

Local organizations can reduce costs by using the toll facilities [e.g., WATS, FX (foreign exchange), CO (central office), or tie trunk facility] at a central location. Incoming local calls can be connected to toll facilities, to local organizations, or extended to remote locations by the local attendant.

For example, a sales person can call an attendant using the LDN (listed directory number) and request a business call using this feature. The call can be extended to a distant destination using a toll facility.

An attendant can also establish a trunk-to-trunk connection between two employees working away from the home office. The attendant can call one employee over a trunk and ask the employee to stand by for the second party. The attendant then calls the second employee over another trunk and informs the employee of the call. The attendant can then connect the two employees. The call is held at the attendant console.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Extend an Incoming Trunk Call to an Outgoing Trunk

Without direct trunk group selection buttons when an automatic outgoing tie trunk isn't accessed:

1. Be sure there is an outside call. [ATND lamp and ICI display are lit.]
2. Press **[START]** . [The SPLIT lamp lights, the calling party is split from attendant, and dial tone is heard.]
3. Dial the appropriate trunk-group access code. [The ANS lamp lights, and second dial tone is heard.]
4. Dial the desired number. (If AAR or ARS are used, dialing an authorization code might be necessary.) [Ringback tone]

5. Press **[RELEASE]** . [The SPLIT, ANS, and ATND lamps go out, the ICI display goes out, and the PA lamp lights.]
6. Calling party is connected to the dialed number via an incoming trunk-to-outgoing trunk connection. [Ringback tone is heard by the calling party.]

Without direct trunk group selection buttons when an automatic outgoing tie trunk is accessed:

1. Be sure there is an outside call. [ATND lamp and ICI display are lit.]
2. Press **[START]** . [The SPLIT lamp lights, the calling party is split from the attendant, and dial tone is heard.]
3. Dial the appropriate trunk-group access code. [The ANS lamp lights, and second dial tone is heard.]
4. An automatic outgoing tie trunk is accessed. [Ringback tone is heard, and RING lamp lights.]
5. Attendant at the distant facility answers and extends the call. [The RING lamp goes out, the ANS lamp lights, and ringback tone is heard by the attendant.]
6. Press **[RELEASE]** . [The SPLIT, ANS, and ATND lamps go out, the ICI display goes out, and the PA lamp lights.]
7. Calling party is connected to the dialed extension via an incoming trunk-to-outgoing trunk connection. [Ringback tone is heard by the calling party.]

With direct trunk group selection buttons when an automatic outgoing tie trunk isn't accessed:

1. Be sure there is an outside call. [ATND lamp and ICI display are lit.]
2. Press the appropriate trunk-group button. [The SPLIT and ANS lamps light, and the calling party is split away from the attendant.]
3. Dial the desired number. (If AAR or ARS is used, dialing an authorization code might be necessay.) [Ringback tone]
4. Press **[RELEASE]** . [The SPLIT, ANS, and ATND lamps go out, the ICI display goes out, and the PA lamp lights.]
5. Calling party is connected to the dialed number via an incoming trunk-to-outgoing trunk connection. [Ringback tone is heard by the calling party.]

With direct trunk group selection buttons when an automatic outgoing tie trunk is accessed:

1. Be sure there is an outside call. [ATND lamp and ICI display are lit.]
2. Press the appropriate trunk-group selection button. [The SPLIT and ANS lamps light, and calling party is split away from the attendant.]
3. An automatic outgoing tie trunk is accessed. [Ringback tone is heard, and RING lamp lights.]

4. Attendant at the distant facility answers and extends the call. [The RING lamp goes out, the ANS lamp lights, and the ringback tone is heard by the attendant.]
5. Press **[RELEASE]** . [The SPLIT, ANS, and ATND lamps go out, the ICI display goes out, and the PA lamp lights.]
6. Calling party is connected to the dialed extension via an incoming trunk-to-outgoing trunk connection. [Ringback tone is heard by the calling party.]

To Connect an Outgoing Trunk Call to an Outgoing Trunk

Without direct trunk group selection buttons when an automatic outgoing tie trunk isn't accessed:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights.]
2. Press **[START]** . [Dial tone]
3. Dial the appropriate outgoing trunk-group access code. [ANS lamp lights, and second dial tone is heard.]
4. Dial the desired first trunk party number. (If AAR or ARS is used, dialing an authorization code might be necessary.) [Ringback tone]
5. When the called party answers, request the called party to stand by for the call from the second trunk party.
6. Press **[START]** [ANS lamp goes out, SPLIT lamp lights, the called party is split away from the attendant, and the attendant hears dial tone.]
7. Dial the appropriate outgoing trunk-group access code for the second trunk group. [ANS lamp lights, and second dial tone is heard.]
8. Dial the desired second trunk party number. (If AAR or ARS is used, dialing an authorization code might be necessary.) [Ringback tone]
9. When the called party answers, request the called party to stand by for the call from the first trunk party.
10. Press **[HOLD]** . (The attendant can handle other calls.) [ATND and SPLIT lamps go out, HOLD lamp lights, and the first and second trunk parties are connected. If no calls are waiting to be answered, the PA lamp lights.]

Without direct trunk group selection buttons when an automatic outgoing tie trunk is accessed:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights.]
2. Press **[START]** . [Dial tone]
3. Dial the appropriate outgoing trunk-group access code. [ANS lamp lights, and second dial tone is heard.]
4. An automatic outgoing tie trunk is accessed. [Ringback tone is heard, and RING lamp lights.]

5. Attendant at the distant facility answers and extends the call. [The RING lamp goes out, the ANS lamp lights, and ringback tone is heard by the attendant.]
6. When the called party answers, request the called party to stand by for the call from the second trunk party.
7. Press **[START]** . [ANS lamp goes out, SPLIT lamp lights, the called party is split away from the attendant, and the attendant hears dial tone.]
8. Dial the appropriate outgoing trunk-group access code. [ANS lamp lights, and second dial tone is heard.]
9. An automatic outgoing tie trunk is accessed. [Ringback tone is heard, and RING lamp lights.]
10. Attendant at the distant facility answers and extends the call. [The RING lamp goes out, the ANS lamp lights, and ringback tone is heard by the attendant.]
11. When the called party answers, request the called party to stand by for the call from the first trunk party.
12. Press **[HOLD]** . (The attendant can handle other calls.) [ATND and SPLIT lamps go out, HOLD lamp lights, and the first and second trunk parties are connected. If no calls are waiting to be answered, the PA lamp lights.]

With direct trunk group selection buttons:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights.]
2. Press the appropriate outgoing trunk-group button. [ANS lamp lights, and dial tone is heard.]
3. Dial the desired first trunk party number. (If AAR or ARS is used, dialing an authorization code might be necessary.) [Ringback tone]
4. When the called party answers, request the called party to stand by for the call from the second trunk party.
5. Press the appropriate outgoing trunk-group button. [SPLIT lamp lights, the called party is split away from the attendant, and dial tone is heard.]
6. Dial the desired second trunk party number. (If AAR or ARS is used, dialing an authorization code might be necessary.) [Ringback tone]
7. When the called party answers, request the called party to stand by for the call from the first trunk party.
8. Press **[HOLD]** . (The attendant can handle other calls.) [ATND and SPLIT lamps go out, HOLD lamp lights, and the first and second trunk parties are connected. If no calls are waiting to be answered, the PA lamp lights.]

With direct trunk group selection buttons when an automatic outgoing tie trunk is accessed:

1. Press an idle loop button. [PA lamp goes out, and ATND lamp lights.]

2. Press the appropriate outgoing trunk-group button. [ANS lamp lights, and dial tone is heard.]
3. An automatic outgoing tie trunk is accessed. [Ringback tone is heard, and RING lamp lights.]
4. Attendant at the distant facility answers and extends the call. [The RING lamp goes out, the ANS lamp lights, and the ringback tone is heard by the attendant.]
5. When the called party answers, request the called party to stand by for the call from the second trunk party.
6. Press the appropriate outgoing trunk-group button. [SPLIT lamp lights, the called party is split away from the attendant, and dial tone is heard.]
7. An automatic outgoing tie trunk is accessed. [Ringback tone is heard, and RING lamp lights.]
8. Attendant at the distant facility answers and extends the call. [The RING lamp goes out, the ANS lamp lights, and the ringback tone is heard by the attendant.]
9. When the called party answers, request the called party to stand by for the call from the first trunk party.
10. Press **[HOLD]** . (The attendant can handle other calls.) [ATND and SPLIT lamps go out, HOLD lamp lights, and the first and second trunk parties are connected. If no calls are waiting to be answered, the PA lamp lights.]

To Monitor an Outgoing Trunk Call to an Outgoing Trunk:

1. Be sure an outgoing trunk call has been extended to another outgoing trunk. [HOLD lamp is lit for the appropriate loop button.]
2. Press the appropriate loop button. [HOLD lamp goes out, and ATND lamp lights.]
3. To allow the call to continue unmonitored, press **[HOLD]** . [ATND lamp goes out, and HOLD lamp lights. If no calls are waiting to be answered, the PA lamp lights.]

To Release an Outgoing Trunk Call to Another Trunk if Disconnect Supervision is Not Present:

1. Be sure the parties have gone on-hook.
2. Press the appropriate loop button. [HOLD lamp goes out, and ATND lamp lights.]
3. To disconnect the second trunk, press **[CANC]** . [ANS lamp goes out.]
4. To disconnect the first trunk, press **[RELEASE]** . [ATND lamp goes out.]

Considerations

Intercept Tone

Intercept tone is heard if an invalid access code is dialed or if the incoming trunk is restricted from accessing the selected trunk.

Calls in Queue

Music or a recorded message is heard if the call is placed in queue after dialing the trunk-group access code.

Undesirable Connections

Because of customer needs or traffic requirements, some trunk-to-trunk connections may be considered undesirable. These connections can be restricted by service technicians via the MAAP (Maintenance and Administration Panel) or DEFINITY Manager II. In this way, a trunk group can be restricted from connecting to other trunk groups.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Data Communications Access

Trunk-to-trunk restrictions applied to DCA trunks restrict trunk calls from direct access to the host computer. Attendant assistance is required to reach DCA ports from the off-premises locations. These attendant-extended calls are denied Data Protection.

Tenant Services

An attendant (in a partition other than Attendant Partition 0) has limited access to the Trunk-to-Trunk Connections feature. When both trunk groups involved in the connection are assigned to the attendant's partition, then the Trunk-to-Trunk Connections feature is allowed. The switch returns intercept treatment to the attendant when this condition is not met.

An attendant in Attendant Partition 0 has full access to the Trunk-to-Trunk Connections feature. These attendants are allowed to connect any two trunks in the switch.

Restricting Feature Use

Specific Trunk-to-Trunk Connections can be restricted by applying trunk-to-trunk restriction to the trunk group. The service technician performs this operation using the MAAP or the DEFINITY Manager II.

Hardware Requirements

None.

Feature Administration

Assignment of the Trunk-to-Trunk Connection feature is on a per-system class-of-service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — TRUNK-TO-TRUNK CONNECTIONS			
PROCEDURE	WORD	PURPOSE	SMT
110	1	Assigns restricted trunk-group dial access codes to trunk groups.	No
111	1	Administers restricted trunk groups for trunk-to-trunk restrictions.	No
275	1	Assigns the Trunk-to-Trunk Connections feature to the system class of service.	Yes
275	4	Assigns trunk-to-trunk transfer to the system class of service.	Yes

Notes:

Unattended Console Service — Alternate Console Position

Description

This feature is one of three Unattended Console Service features. These features are designed to work together to provide flexibility and enhance attendant services under a wide variety of circumstances. While designed to work together, each of the Unattended Console Service features can function separately and independently from the others. The Unattended Console Service features are:

- Unattended Console Service — Alternate Console Position
- Unattended Console Service — Call Answer From Any Voice Terminal
- Unattended Console Service — Preselected Call Routing.

The way in which these features work together to supplement each other's coverage is shown in Figure 127-1.

When active, the Alternate Console Position version of this feature directs calls for one attendant console to an alternate console. The alternate console must be identical to and have the same features as the primary console.

This feature is useful at night or during periods when consoles have been removed from service. Also, providing attendant services at another location during certain periods (e.g., a security office) provides convenience and efficiency of operation. The alternate console position provides all the attendant capabilities of the primary attendant console.

Feature History and Development

This feature was first available for System 85 in Release 1. Them have been no changes to this feature since Release 1.

User Operations

The following is the user operating procedure for this feature.

To Activate an Alternate Console Position:

The attendant at the primary console operates a transfer switch (on/off switch mounted on a wall or desk). Calls for the primary console route to the alternate console.

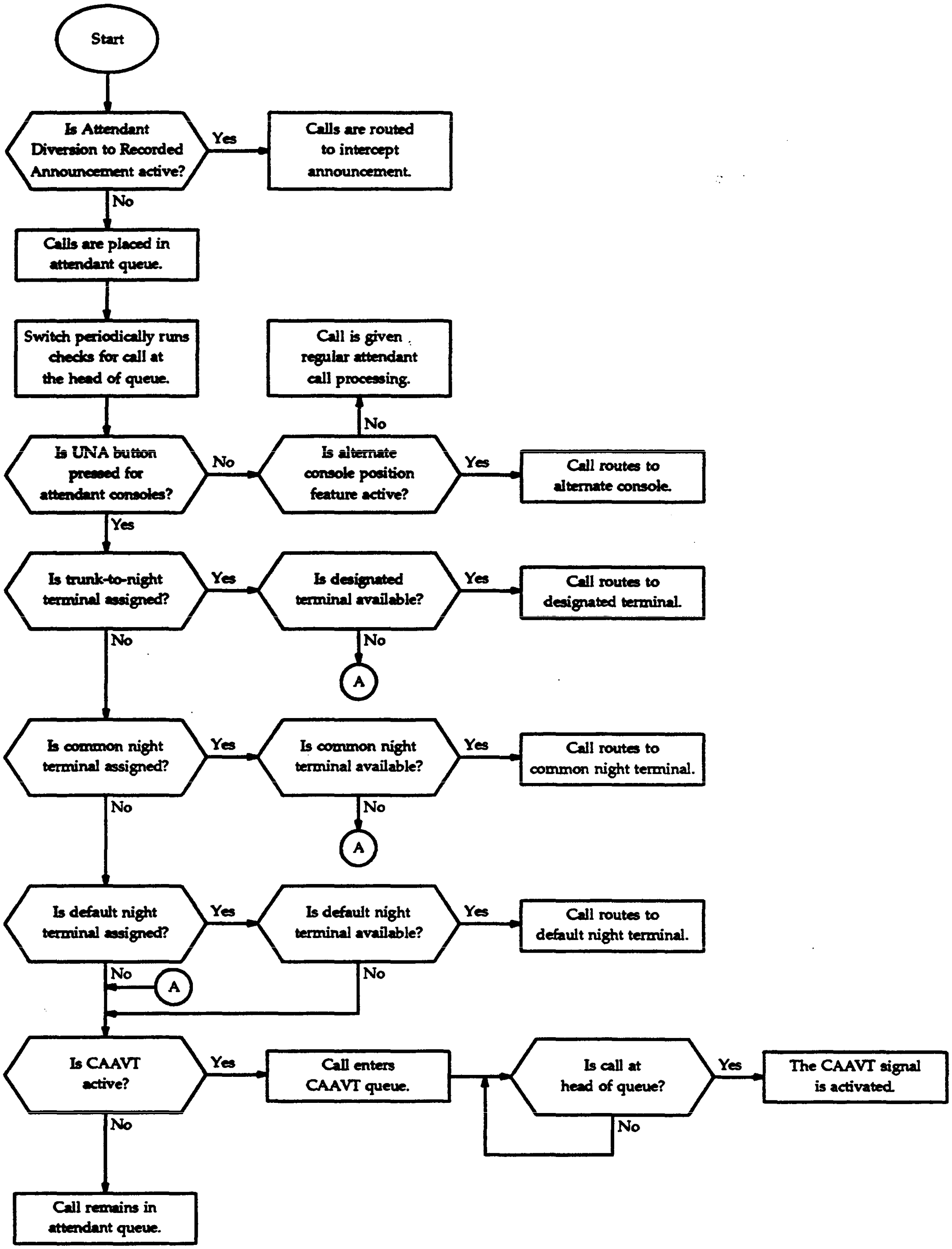


Figure 127-1. Interrelation of Unattended Console Service Features

Considerations

Alternate Console Requirements

An alternate console can be provided for each regular console. Each alternate position requires a transfer switch, relay panel, and attendant console.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Control of Trunk Group Access

When the Alternate Console Position feature and the Attendant Control of Trunk Group Access feature are provided and activated concurrently, calls to a controlled trunk are routed to the alternate console position for subsequent processing.

Dial Access to Attendant

The Alternate Console Position feature provides an alternate attendant position which can be used in lieu of a regular (primary) attendant position. Calls normally directed to the primary position route to the alternate position.

Malicious Call Trace

While the Alternate Console Position feature is activated, the alternate attendant position (instead of the regular attendant position) is alerted to trace malicious calls. If the alternate attendant responds to an alert first (by pressing the MCT CONT button), this attendant can also trace the call.

Tenant Services

The Alternate Console Position feature is well behaved in a partitioned switch. Using the Alternate Console Position feature, an alternate console is physically connected to it's primary console, and is activated with a 609A transfer panel. Since the switch performs partitioning checks before calls can be routed to/from the primary console, an alternate console would have the same abilities in a partitioned switch as the specific primary console to which it is connected. The system manager need only ensure that the operation of both consoles are members of the same partition.

Restricting Feature Use

Position buttons on the alternate console control the operation of the alternate console. For example:

- Pressing the position busy (PBSY) button restricts the alternate console position from receiving traffic. The PBSY button affects only the console on which it is operated.
- Pressing the position unattended (UNA) button restricts the alternate console position from receiving traffic. Operation of the UNA button places the switching system in the night mode and diverts all attendant-seeking traffic to a preselected voice terminal.

Hardware Requirements

The Alternate Console Position feature requires the following additional or special hardware for a traditional or universal module:

- An alternate console, 6017-type key, or equivalent
- A 609A transfer panel.

Feature Administration

The Alternate Console Position feature is a hardware implemented feature and requires no administrative procedures.

Unattended Console Service — Call Answer From Any Voice Terminal

Description

This feature is one of three Unattended Console Service features. These features are designed to work together to provide flexibility and enhanced attendant services under a wide variety of circumstances. While designed to work together, each of these features can function separately and independently from the others. The Unattended Console service features are:

- Unattended Console Service — Alternate Console Position
- Unattended Console Service — Call Answer From Any Voice Terminal
- Unattended Console Service — Preselected Call Routing.

The way in which these features work together to supplement each other's coverage is shown in Figure 128-1.

The CAAVT (Call Answer From Any Voice Terminal) version of the Unattended Console Service feature group allows any unrestricted voice terminal user to answer calls made to the attendant when the attendant is not on duty and a specific voice terminal has not been designated to answer the calls.

The incoming attendant-seeking calls activate a distinctive gong, bell, or chime. If necessary, the answering voice terminal user can transfer the call to another voice terminal.

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

User Operations

The following are the user operating procedures for this feature.

To Answer a Call From an Unrestricted Voice Terminal:

1. Go off-hook. [Dial tone]
2. Dial the CAAVT access code. [Incoming call is connected.]

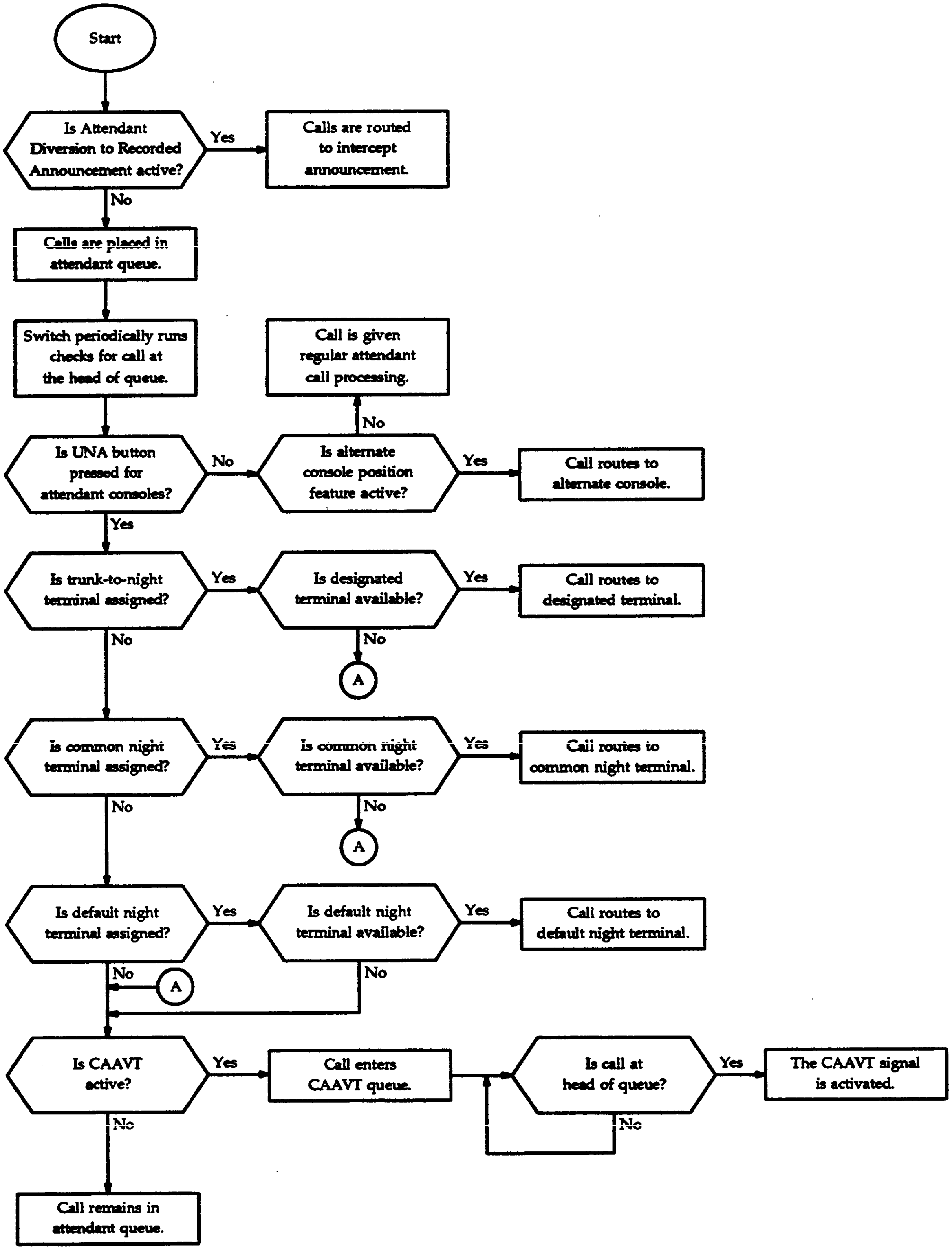


Figure 128-1. Interrelation of Unattended Console Service Features

Considerations

Backup Function

The CAAVT (Call Answer from Any Voice Terminal) feature operates when extensions have been assigned. The CAAVT feature also operates when a common or default voice terminal, assigned for the Preselected Call Routing feature, is busy on another call.

Intercept Treatment

If the voice terminal user receives intercept tone after dialing the CAAVT access code, one or more of the following applies:

- The extension used is Inward, Origination, or Terminal-to-Terminal Only restricted.

- Another user has already answered the call, and the CAAVT queue is empty.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Control of Trunk Group Access

When Call Answer from Any Voice Terminal and Attendant Control of Trunk Group Access are provided and activated concurrently, calls to a controlled trunk are routed to intercept tone.

Call Detail Recording

While CAAVT is active, calls answered by dialing the CAAVT access code are extended using the 3-Party Conference and Transfer features. Therefore, Forced Entry of Account Codes applies to these calls acceding to the options assigned to the answering "night" terminal.

Centralized Attendant Service

When a Centralized Attendant Service branch switch is in the Call Answer from Any Voice Terminal mode of operation, incoming tie trunk and Remote Access trunk calls, as well as Listed Directory Number calls, reach the Call Answer from Any Voice Terminal queue. These calls do not receive Intercept Treatment.

Conference—Attendant Five Party

If the attendant releases a Conference—Attendant Five Party call and then changes to an Unattended Console Service feature, attendant recall is denied. The conference must be held on the console prior to starting the Unattended Console Service feature to provide attendant service. When attendant recall is used from a conference held on the attendant console, the recall is activated toward the attendant console where the call is held even if Unattended Console Service is in effect.

Conference—Attendant Six Party

When a conference call that was established via the Conference—Attendant Six Party feature (established when the console is in the normal or "daytime" mode) is released by the attendant and then the Call Answer from Any Voice Terminal feature is activated, an attendant is no longer available to service the conference call. If a terminal user, who is also a conferee, presses the RECALL button to recall an attendant, the recall is disallowed. The console must hold the conference prior to activating the CAAVT feature to provide attendant service once the CAAVT feature is activated.

Conference—Three Party

A single-appearance voice terminal extension assigned as the default extension (under Preselected Call Routing), has access to the Conference—Three Party feature when functioning in its Unattended Console Service role, even though Conference—Three Party might not be assigned to its extension class of service. However, a Call Answer From Any Voice Terminal user (not using the Preselected Call Routing default extension), must be using a terminal with Conference—Three Party assigned to use the Conference—Three Party feature. All multiappearance voice terminals have the Conference—Three Party capability.

Intercept Treatment

If an attendant activates Attendant Diversion to Recorded Announcement and then an attendant activates Unattended Console Service, calls placed to the attendant queue do not enter the CAAVT queue. Instead these callers will receive the recorded announcement. If this operation is not desired, the attendant should deactivate Attendant Diversion to Recorded Announcement before activating Unattended Console Service.

Last Number Dialed

The LND (Last Number Dialed) feature does not store or redial the CAAVT access code at a voice terminal that responds to the general night-service alert. Instead, the previously stored digits will remain in LND memory.

Malicious Call Trace

While the CAAVT feature is active, the switch does not sound the CAAVT signaling device in response to a malicious call. Instead, the switch will activate a voice recorder and alert the attendant consoles in the normal manner. After an attendant returns to an attendant position, the trace can be performed and the Malicious Call Trace feature can be deactivated.

Restriction—Voice Terminal Restrictions

The switch denies dialing the Call Answer from Any Voice Terminal access code from an Inward restricted terminal, Terminal-to-Terminal Only Calling terminal, or Origination restricted terminal.

Tenant Services

The Call Answer from Any Voice Terminal feature is not partitioned. In response to the system-wide "gong alerting," the CAAVT access code can be dialed from an unrestricted voice terminal belonging to any extension partition.

When a CAAVT call is answered using a voice terminal (in a partition other than Extension Partition 0), the answering party is allowed to transfer the call to an extension in the same extension partition or to an extension in Extension Partition 0. If the answering party tries to transfer the call to any other partition, intercept treatment is returned.

When a CAAVT call is answered using a voice terminal in Extension Partition 0, the answering party is allowed to transfer the call to any extension partition in the switch.

Transfer

A single-appearance voice terminal extension assigned as the default extension (under Preselected Call Routing), has access to the Transfer feature when functioning in its Unattended Console Service role, even though Conference—Three Party might not be assigned to its extension class of service. However, a Call Answer from Any Voice Terminal user (not using the Preselected Call Routing default extension), must be using a terminal with Transfer assigned to use the Transfer feature. All multiappearance voice terminals have the Transfer capability.

Hardware Requirements

The Call Answer from Any Voice Terminal feature requires the following additional or special hardware.

For Traditional Modules:

- A customer-provided signaling device operated by a 20-hertz signal
- An SN229 interface circuit.

For Universal Modules:

- A customer-provided signaling device operated by 20-hertz signal
- A TN746 interface circuit.

Feature Administration

Assignment of the Call Answer from Any Voice Terminal feature is on a system class-of-service basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal).

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES UNATTENDED CONSOLE SERVICE — CAAVT			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
275	2	Assigns CAAVT to the system class of service and administers the SN229 equipment location.	Yes
350	1	Assigns the first digit of the feature dial access code (if required).	No
350	2	Assigns the feature dial access code. The applicable recodes are as follows: 16 CAAVT code 47 CAAVT—Activate for CAS Backup.	No

Unattended Console Service — Preselected Call Routing

Description

This feature is one of three Unattended Console Service features, which are designed to work together to provide flexibility and enhance attendant services under a wide variety of circumstances. While designed to work together, each of the Unattended Console Service features can function separately and independently from the others. The Unattended Console Service features are:

- Unattended Console Service — Alternate Console Position
- Unattended Console Service — Call Answer from Any Voice Terminal
- Unattended Console Service — Preselected Call Routing

The way in which these features work together to supplement each other's coverage is shown in Figure 129-1.

The Preselected Call Routing version of the Unattended Console Services feature group directs attendant-seeking calls to designated extension numbers whenever the console is unattended. Distinctive 3-burst ringing is used at the designated voice terminal to alert the user to the nature of the incoming call.

Feature History and Development

This feature was first available on System 85 in Release 1.

Dial access activation and deactivation of Unattended Console Service are provided for the R2 V4 and DEFINITY Generic 2 switches.

Preselected Terminals

The attendant can designate and cancel the assignment of extension numbers for two of the three preselected terminal types. There are three types of preselected terminal assignments available.

Common Terminal

The common terminal (also referred to as the common night terminal), is assigned by the attendant. When the common terminal is assigned, all attendant-seeking calls are routed to the designated extension number.

Default Terminal

The default terminal is assigned by the switch administrator using one of the switch administration tools (see the Feature Administration section). When a common terminal has not been assigned, attendant seeking calls are directed to the default terminal extension number.

Trunk-to-Terminal

The trunk-to-terminal function is assigned by the attendant. When assigned, this function directs calls on a specific incoming trunk to the designated extension number.

Override vrs Clearing Preselected Terminal Assignments

Override Common Terminal

An attendant are dial the Override Common Terminal access code to bypass the common terminal and the default terminal, and divert calls directly to the CAAVT queue. However, on a partitioned switch, the Override Common Terminal function is disabled. This function is not used because CAAVT "gong alerting" is not partitioned.

Clear All Terminals

An attendant can dial the Clear All Terminals access code to cancel the common terminal assignment and/or the current trunk-to-terminal assignments for Preselected Call Routing. When this is done, Preselected Call Routing sends incoming calls to the default common terminal (if assigned) and then to the CAAVT queue (if assigned).

On a partitioned switch, this functionality works differently. A partition's controlling attendant can dial the Clear All Terminals access code, followed by the number of the attendant partition. When this is done, only the common terminal assignment for that partition is canceled. (The trunk-to-terminal associations, assigned in Procedure 116, Word 1 or Procedure 150, are not affected). In this case, Preselected Call Routing would send incoming calls to the trunk-to-terminal association (if assigned), then to the partition's default terminal (if assigned), and then to the CAAVT queue (if assigned).

User Benefits

The user benefits derived from this service are:

- Improved customer services by providing holiday, weekend, or night service
- Prevention of lost calls
- Flexibility in meeting customer needs.

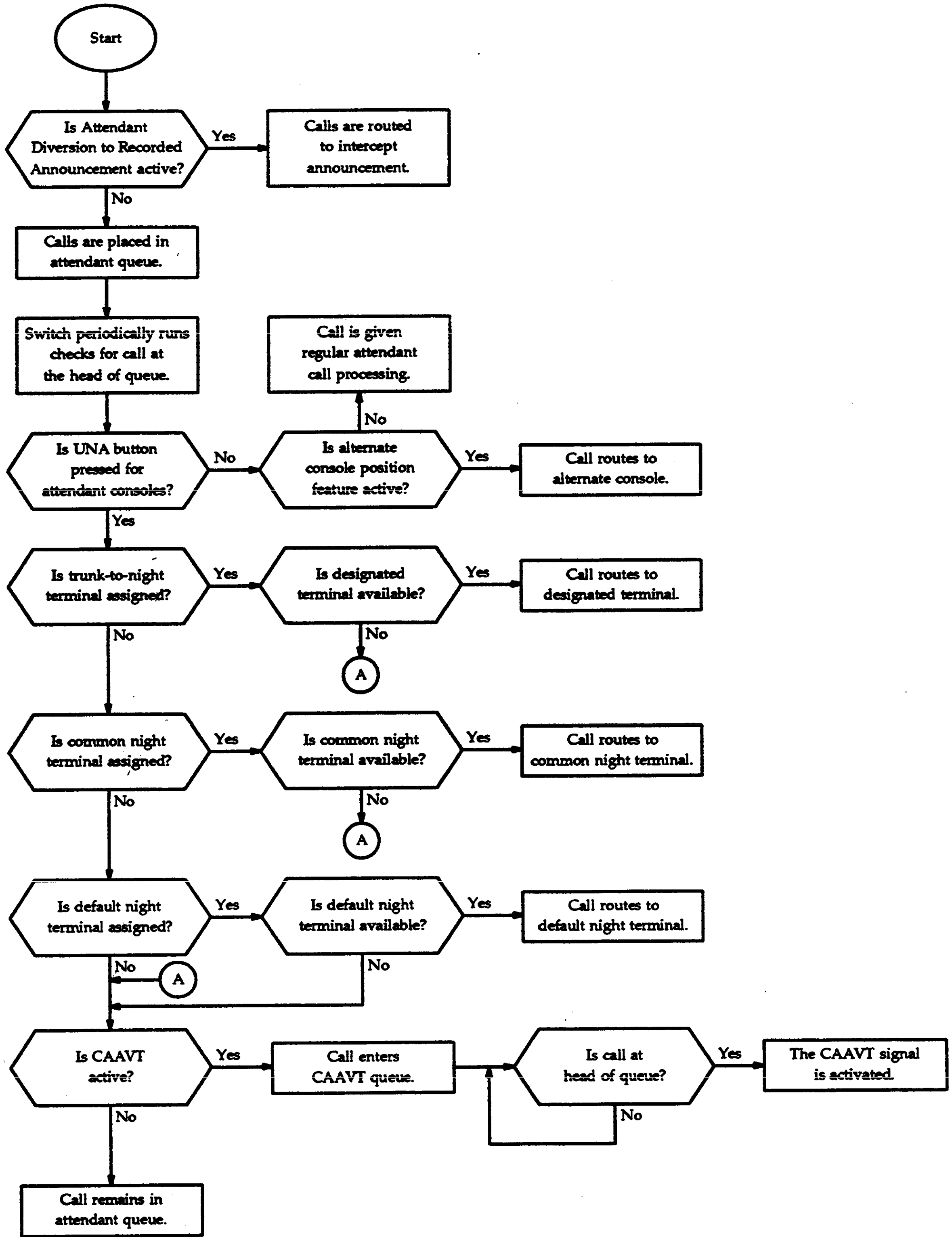


Figure 129-1. Interrelation of Unattended Console Service Features

User Operations

The following are the user operating procedures for this feature.

To Activate Preselected Call Routing:

Press the **[UNA]** (position unattended) button. [UNA lamp flashes.]

To Activate Preselected Call Routing Using the Access Code (R2 V4):

1. Press an idle loop button. [PA lamp goes out.]
2. Press the **[START]** button. [Dial tone]
3. Dial the Activate Unattended Console Service access code. [The switch returns confirmation tone, and UNA lamp (if provided) flashes.]
4. Press the **[RELEASE]** button. [PA lamp lights.]

To Deactivate Preselected Call Routing Using the Access Code (R2 V4):

1. Press an idle loop button. [PA lamp goes out.]
2. Press the **[START]** button. [Dial tone]
3. Dial the Deactivate Unattended Console Service access code. [The switch returns confirmation tone, and UNA lamp (if provided) goes out.]
4. Press the **[RELEASE]** button. [PA lamp lights.]

To Assign the Common Terminal:

1. Press an idle loop button. [PA lamp goes out.]
2. Press **[START]** . [Dial tone]
3. Dial the Assign Common Terminal access code. [Second dial tone]
4. Dial the terminal's extension number,

   or

   Press the appropriate DXS button. [The switch returns confirmation tone, and the UNA lamp lights steadily.]
5. Press **[RELEASE]** . [PA lamp lights.]

To Establish a Trunk-to-Voice Terminal Assignment:

1. Press an idle loop button. [PA lamp goes out.]
2. Press the **[START]** button. [Dial tone]
3. Dial the Assign Terminal to Trunk access code. [Second dial tone]

4. Dial the terminal's extension number,

   or

   Press the appropriate DXS button. [Third dial tone]

5. Dial the trunk group's access code,

   or

   Press the appropriate DTGS button. [Silence]

6. Dial the number of the trunk to be assigned to the terminal (for example, "007"). [Confirmation tone]

   NOTE: Using an R2 V1 switch, only two digits should be dialed for trunk numbers (for example, "07", or "99"). This is because R2 V1 switches cannot have more than 99 trunks per trunk group.

7. Press the **[RELEASE]** button. [PA lamp lights.]

To Cancel the Common Terminal and/or the Trunk-to-Terminal Assignments:

1. Press an idle loop button. [PA lamp goes out.]
2. Press **[START]** . [Dial tone]
3. Dial the Clear All Terminals access code. [Confirmation tone]
4. Press **[RELEASE]** .

To Disable the Common Terminal:

1. Press an idle loop button. [PA lamp goes out.]
2. Press **[START]** . [Dial tone]
3. Dial the Override Common Terminal access code. [Confirmation tone]
4. Press **[RELEASE]** . [PA lamp lights.]

Considerations

Calls in Progress

Calls alredy in progress when Unattended Console Service is activated or canceled are not disturbed.

Extension Assignment

One common extension number and one default extension number can be assigned. The default extension number is assigned with the MAAP, SMT, or the TCM feature on System 85 switches and with the DEFINITY Manager II on DEFINITY Generic 2 switches.

Override Common Terminal Function

On an ***unpartitioned*** System 85 or DEFINITY Generic 2, an attendant can dial the Override Common Terminal access code to bypass the common terminal and the default terminal, and divert calls directly to the CAAVT queue. On a partitioned System 85 or DEFINITY Generic 2, the Override Common Terminal function is disabled. This function is not necessary on a partitioned switch because CAAVT "gong alerting" is not partitioned.

Hard Processor Swaps and Power Failures

After a power interruption, the current common extension number may be lost.

After a hard processor swap or a power failure, the switch returns to service with Unattended Console Service active. At this time, an attendant can press the UNA button to restore normal service. A soft processor swap does not invoke the Unattended Console Service feature.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Attendant Control of Trunk Group Access

When Preselected Call Routing is provided and activated concurrent with Attendant Control of Trunk Group Access, calls directed to a controlled trunk group are routed to intercept tone.

Attendant Interposition Calling and Transfer

If an attendant interposition calls another attendant with Preselected Call Routing active, the call is placed in the called attendant's priority queue. When Preselected Call Routing is deactivated, the call is removed from the priority queue and alerting is provided. The call does not route to the preselected voice terminal.

ACD (Automatic Call Distribution)

Beginning with Issue 1.6 of R2 V3 and Issue 1.2 of R2 V4, an ACD split can indirectly perform night service for the attendant queue. The designated terminal, the common terminal, or the default terminal for preselected call routing can activate Call Forwarding—Follow Me to forward all calls to an associated extension number of an ACD split. When this is done, every call to the terminal enters the ACD split's queue and receives normal ACD processing.

Automatic Circuit Assurance

When the switch is in the preselected call routing mode, partial Automatic Circuit Assurance operation allows trunk referrals to be recorded on the audit trail without originating referral calls to the attendant.

Call Coverage

Calls routing to a preselected voice terminal when the attendant console is in the unattended mode do not redirect to coverage when that terminal has coverage assigned. Only direct calls to the preselected voice terminal assigned coverage redirect.

CDR (Call Detail Recording)

When the switch is in the unattended mode, CDR records calls routed to the common terminal.

While Preselected Call Routing is active, routed to a preselected voice terminal are extended using the 3-Party Conference and Transfer features. Therefore, Forced Entry of Account Codes applies to these calls according to the options assigned to the preselected voice terminal.

Call Forwarding—Follow Me

When in the unattended console (night service) mode, the extension designated to receive attendant-seeking-calls can activate Call Forwarding—Follow Me. As long as the forwarded to station is on-net, there is not problem. However, calls should not be forwarded to an off-net station when in the unattended console mode. Attendant-seeking-calls will forward to an on-net station but not to an off-net station. If forwarding to an off-net station is attempted, attendant- seeking-calls will not forward but will wait in the attendant queue until the call is abandoned or an attendant console is activated. See also the ACD feature interaction discussed earlier in this section.

Call Park

When Preselected Call Routing is active, if a trunk party is placed in Call Park and the 2-minute timer for ringback tone times out, the trunk call is routed to the assigned preselected voice terminal.

Call Vectoring

Vector processing is not available for incoming calls to the default voice terminal for Unattended Console Service. Entering a VDN in Procedure 275, Word 2 as the default extension number is not allowed. When this is attempted, an administration error will

Entering a VDN as the "Trunk-to-Night Terminal" extension number for a trunk group in Procedure 116, Word 1 or Procedure 150 is not allowed. When this is attempted, an administration error will occur.

An attendant is not allowed to establish a "Trunk-to-Night Terminal" assignment by dialing a VDN as the night-terminal extension. When this is attempted, the switch returns intercept tone.

Beginning with Issue 1.2 of R2 V4, Call Vectoring can indirectly perform night service for the attendant queue. The designated terminal, the common terminal, or the default terminal for preselected call routing can activate Call Forwarding—Follow Me to forward all calls to a VDN (Vector Directory Number). When this is done, every call to the common terminal enters the VDN's vector processing and receives the normal treatment programmed within the vector.

Call Waiting

Call Waiting is not available when a call is already waiting on the busy preselected voice terminal, or when the terminal user has placed another call on hold. The next call to that terminal forwards to the Call Answer from Any Voice Terminal feature, if provided. The incoming queue holds all other calls.

With Call Waiting, the following could happen when going on-hook with a call waiting and one or more calls to the busy preselected voice terminal are in queue: the user may be connected to the next call in the queue instead of the waiting call. This occurs because the queue scan is performed every 100 milliseconds, whereas the Call Waiting scan is performed every 2 seconds. To ensure a connection with call waiting the terminal user presses the RECALL button and dials the answer-hold access code. The terminal user then connects to the waiting party.

Conference—Attendant Five Party

If the attendant releases a Conference—Attendant Five Party call and then changes to an Unattended Console Service feature, attendant recall is denied. The conference must be held on the console prior to starting the Unattended Console Service feature to provide attendant service. When attendant recalI is used from a conference held on the attendant console, the recall is activated toward the attendant console where the call is held even if Unattended Console Service is in effect.

Conference—Attendant Six Party

When a conference call that was established via the Conference—Attendant Six Party feature (established when the console is in the normal or "daytime" mode) is released by the attendant and then the Preselected Call Routing feature is activated, an attendant is no longer available to service the conference call. If a terminal user, who is also a conferee, presses the RECALL button to recall an attendant the recall is disallowed. The console must hold the conference prior to activating the Preselected Call Routing feature to provide attendant service once the Preselected Call Routing feature is activated.

Conference—Three Party

When the switch is in the Preselected Call Routing mode of operation, the Conference—Three Party feature is always enabled for the preselected terminal, regardless of the voice terminal's extension class of service. However, the single-appearance preselected voice terminal must have Conference—Three Party assigned to its class of service to enable the user to direct incoming call to outgoing trunks. All multiappearance voice terminals have the Conference—Three Party capability.

Extension Number Portability

The preselected voice terminal extension must reside at the local switch. It cannot be ported to another node and still serve as a preselected voice terminal at the ported-from node.

Intercept Treatment

If an attendant activates Attendant Diversion to Recorded Announcement and then an attendant activates Unattended Console Service, calls placed to the attendant queue do not complete to the preselected voice terminal. Instead, these callers will receive the recorded announcement. If this operation is not desired, the attendant should deactivate Attendant Diversion to Recoded Announcement before activating Unattended Console Service.

Main/Satellite

When a Direct Inward Dialed call to the main receives intercept-to-attendant treatment and the preselected Call Routing feature is active, the call routes to the preselected voice terminal at the main switch. When a Direct Inward Dialing call to a satellite receives intercept treatment and there is neither a recorded announcement nor an attendant console, the call routes to the satellite preselected voice terminal.

Malicious Call Trace

While the Preselected Call Routing feature is active, preselected voice terminals are not alerted to trace malicious calls. Instead, the switch will activate a voice recorder and alert the attendant consoles in the normal manner. After an attendant returns to an attendant position, the trace can be performed and the Malicious Call Trace feature can be activated.

Power Failure Transfer

When full (flexible) night station service is activated and a power failure occurs, the terminal line(s) assigned by the attendant is lost from memory. When commercial power is restored and reinitialization occurs, night station service calls are routed to the default terminal.

Restriction—Voice Terminal Restrictions

The Preselected Call Routing feature takes precedence over the Voice Terminal Restrictions (Inward, Termination, Terminal-to-Terminal Only Calling, and Manual Terminating Line) feature. Therefore, incoming trunk calls can terminate on the preselected voice terminal, and the preselected terminal can initiate outgoing trunk calls with these restrictions active. This is the only facet of the restrictions that is overridden. For example, all calls, except incoming trunk calls, are denied unless the calls are attendant-completed for voice terminals with Manual Terminating Line Restriction.

Tenant Services

Default Terminal

For an ***unpartitioned*** System 85 or DEFINITY Generic 2, the default terminal is assigned on a system-wide basis in Procedure 275, Word 2.

For a ***partitioned*** System 85 or DEFINITY Generic 2, the system-wide administration is blocked. Instead, a default common terminal can be assigned to each attendant partition in Procedure 270, Word 3.

Trunk-to-Terminal Assignments

Using an ***unpartitioned*** System 85 or DEFINITY Generic 2 with full night service, an attendant can dial the Assign Terminal to Trunk access code to set trunk-to-terminal assignments for Preselected Call Routing.

Using a ***partitioned*** System 85 or DEFINITY Generic 2, this function of full night service is disabled. Instead, these associations can only be assigned in Procedure 116, Word 1 or Procedure 150. However, there are no software checks in these procedures to ensure the proper partitioning of these associations. It is the responsibility of the system manager to design allowable and practical trunk-to-terminal assignments.

Common Terminal

On an ***unpartitioned*** System 85 or DEFINITY Generic 2, an attendant can dial the Assign Common Terminal access code to designate the common terminal.

On a ***partitioned*** System 85 or DEFINITY Generic 2, this functionality is modified. To assign a common terminal for a partition, the partition's controlling attendant dials the Assign Common Terminal access code, the extension number of the partition's common terminal, and then the number of the attendant partition.

Clear All Terminals

On an ***unpartitioned*** System 85 or DEFINITY Generic 2, an attendant can dial the Clear All Terminals access code to cancel the common terminal assignment and/or the current trunk-to-terminal assignments for Preselected Call Routing. When this is done, Preselected Call Routing sends incoming calls to the default common terminal (if assigned) and then to the CAAVT queue (if assigned).

Using a ***partitioned*** System 85 or DEFINITY Generic 2, this functionality is modified. A partition's controlling attendant can dial the Clear All Terminals access code, followed by the number of the attendant partition. When this is done, ***only*** the common terminal assignment for that partition is canceled. (The trunk-to-terminal associations, assigned in Procedure 116, Word 1 or Procedure 150, are not affected.) So, Preselected Call Routing would send incoming calls to the trunk-to-terminal association (if assigned), then to the partition's default terminal (if assigned), and then to the CAAVT queue (if assigned).

Override Common Terminal

On an ***unpartitioned*** System 85 or DEFINITY Generic 2, an attendant can dial the Override Common Terminal access code to bypass the common terminal and the default terminal, and divert calls directly to the CAAVT queue.

On a ***partitioned*** System 85 or DEFINITY Generic 2, the Override Common Terminal function is disabled. This function is no longer necessary because CAAVT "gong alerting" is not partitioned.

Timed Recall on Outgoing Calls

When the Preselected Call Routing feature is active, the Timed Recall on Outgoing Calls feature is deactivated.

Transfer

When the switch is in the Preselected Call Routing mode of operation, the Transfer feature is always enabled for the preselected terminal, regardless of the preselected terminal's extension class of service. However, the single-appearance preselected voice terminal must have the Transfer assigned to its class of service to enable the user to direct incoming calls to outgoing trunks. All multiappearance voice terminals have the Transfer capability.

Hardware Requirements

None.

Feature Administration

Assignment of the Preselected Call Routing feature is on a per-system basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES UNATTENDED CONSOLE SERVICE — PRESELECTED CALL ROUTING			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
116	1	Assigns the default terminal to a trunk group.	No
150	1	Assigns the default terminal to a trunk group.	No
203	1	Assigns the UNA button to the attendant console(s).	No
275	2	Assigns Preselected Call Routing to the system class of service. Also, use this procedure to assign the default terminal extension and to display the current common terminal extension.	Yes
350	1	Assigns the first digit of the feature dial access codes (if required).	No
350	2	Assigns the feature dial access codes. The applicable encodes are as follows: 22 Clear all terminals 23 Assign common terminal 24 Override common terminal 25 Assign terminal to trunk 97 Activate Unattended Console Service 98 Deactivate Unattended Console Service.	No

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN UNATTENDED CONSOLE SERVICE — PRESELECTED CALL ROUTING	
PATH NAME	**PURPOSE**
terminal-change system parameters (select the Night-Service option)	Assigns Preselected Call Routing to the system class of service and assigns the default terminal extension number.

Unified Messaging

Description

Unified Messaging is an integrated, cost-effective family of distinct messaging services working together to meet a customer's total organizational messaging requirements.

Unified Messaging allows prompt internal and external communications for voice and data. This feature is designed to break the barriers to efficient phone communications including busy signals, unanswered calls, repeated call backs, waiting on hold, and missed communications. These barriers are broken by having efficient alerting and notification of new messages that can be universally accessed from the messaging services.

Messaging Principles

The integration is achieved through five messaging principles. The five principles are:

- **Universal Mailbox**

  Provides single access for all messages in the system.

- **Integrated Preparation**

  Provides the ability to create messages on one service for distribution to disparate mailboxes in other services. A distribution group for mail can consist of subscribers to any of the messaging services.

- **Integrated Notification**

  Allows the AMWL (Automatic Message Waiting Lamp) and/or mail counter to notify all messaging services of incoming messages. Their source and nature are identified.

- **Universal Retrieval**

  Allows retrieval of messages from mailboxes using a printer, voice terminal, or data terminal.

- **Universal Connectivity**

  Permits intra- and inter-premises communication over both public and private networks. This is made possible by and adoption of the CCITT (International Telegraph and Telephone Consultative Committee) standards for message handling services.

Available Messaging Services

The messaging services available with System 85 and DEFINITY Generic 2 are:

- AUDIX (Audio Information Exchange)
- Call Coverage
- Electronic Mail

  Electronic Mail consists of EDC (Electronic Document Communications) Mail and UNIX® System Mail
- Leave Word Calling
- Message Center Service

AUDIX (Audio Information Exchange)

The AUDIX message-handling system uses stored voice prompts and announcements to guide subscribers through their messaging operations. AUDIX provides subscribers with the ability to send and receive voice messages to or from other AUDIX subscribers. AUDIX subscribers can also receive messages from nonsubscriber. See the AUDIX chapter of this manual.

Call Coverage

Call Coverage provides alternate answering points for calls that would otherwise go unanswered. For a principal (user with Call Coverage active), Call Coverage provides automatic redirection of calls that meet specified conditions to a coverage path. The principal, coverage path, and coverage criteria (conditions) combine to make a coverage group. See the Call Coverage chapter of this manual.

Electronic Mail

Electronic Mail is either of two applications that allow users to electronically transmit messages and documents to other users in the network. The two electronic-mail applications provided with System 85 and DEFINITY Generic 2 are:

- EDC (Electronic Document Communications):

  EDC gives the user the ability to store, edit, retrieve, and send materials to other users of integrated voice/data stations on a scheduled basis (at specified dates and times). Recipients can include non-EDC users. EDC also lets users update their status for retrieval at the Message Center.
- UNIX System Mail:

  UNIX System Mail uses the UUCP (UNIX System-to-UNIX System Copy) as a communications link between systems. With the UNIX System V operating system resident on the 3B5 AP, mail can be exchanged with users on other UNIX systems.

LWC (Leave Word Calling)

The LWC feature allows internal callers (internal to the local switch or to the Distributed Communications System environment) to leave messages for internal principals (the called parties) without the assistance of a secretary or Message Center agent The LWC feature stores a standard message on the AP (Applications Processor), AUDIX adjunct, or the switch. Leave Word Calling on the switch provides economical message handling on System 85s or DEFINITY Generic 2s with a low volume message service requirement, where an AP or an AUDIX adjunct is not necessary. The storage facility used for LWC messages is an administrable option. See the Leave Word Calling chapter of this manual.

MCS (Message Center Service)

The MCS feature provides routing of calls to centralize agents who give a personalized greeting and take and relay messages. A Message Center provides a central exchange point for messages and ensures proper coverage of calls made to voice terminals. The Message Center is an ACD (Automatic Call Distribution) or EUCD (Enhanced Uniform Call Distribution) split, and must be the last point in a Call Coverage path (see 999-700-559).

Feature History and Development

This feature was first available for System 85 in Release 2, Version 3.

User Operations

Each different messaging feature and service has its own user operating procedures. There are no separate operations for Unified Messaging. See the user manuals for EDC and MCS user operations and the feature chapters in this manual for AUDIX, Call Coverage, and LWC user operations.

Considerations

Loss of DCIU Link

The DCIU (Data Communications Interface Unit) is critical to messaging services that use an AP or adjunct. When the DCIU connection is down, messaging operation cannot be performed with these services. For LWC (on the AP) the user receives a "messages unavailable" message. No attempt is made to display AUDIX status.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the operation of this feature.

Unified Messaging as Viewed by an EDC User

Table 130-A shows the interactions while sending or receiving messages using EDC.

TABLE 130-A. EDC Interactions With Unified Messaging services

EDC Users		
Service	**EDC Users Can Receive Mail From:**	**EDC Users Can Send Mail To:**
EDC	Complete EDC functionality.	Complete EDC functionality.
LWC	Recipient must be an EDC and MCS subscriber. A LWC message can be demand- or auto-forwarded to the EDC mailbox from MCS.	Users cannot send a text message from EDC into integrated LWC on the switch.
AUDIX	Recipient must be an EDC and AUDIX subscriber. An AUDIX message remains in AUDIX and the EDC user receives the notification "AUDIX= y" on the status line of the terminal, "You have voice mail" on the message line, and an audible alert at the terminal.	Recipient must be an EDC and AUDIX subscriber. An EDC message remains in EDC and the AUDIX user receives the verbal notification, "You have Electronic Text messages."
MCS	Recipient must be an EDC and MCS subscriber. An MCS message can be demand- or auto-forwarded to the EDC mailbox.	Recipient must be an MCS subscriber. Nonprivate messages less than or equal to 296 characters go to the subscribers MCS mailbox. If the message is more than 296 characters, it goes to the recipient's assigned printer. An MCS Agent can receive itinerary updates from EDC subscribers. Also, there is notification of text messages.
UNIX System	Recipient must be an EDC subscriber. Incoming mail is placed in the EDC mailbox and terminates as EDC mail.	Location indicated on EDC mail must be a known UNIX System machine name.
*Leave Word Calling on an AP stores mmessages on the AP in the MCS mailbox. Integrated Leave Word Calling stores messages in the switch's database.		

When a new message arrives while the user is using EDC at a data terminal, the user can be alerted to the new message in seven ways. The seven ways are:

- If applicable, the MWL lights.
- The single new mail counter on the status line increments for EDC or UNIX system mail.
- For MCS messages, the MCS: y/n indicator changes from "n" to "y."
- For AUDIX messages, the AUDIX y/n indicator changes from "n" to "y."
- For MCS messages, the message line displays, "You have MCS messages."

- For AUDIX messages, the message line displays, "You have AUDIX messages."
- There is an audible indication from the terminal.

When the new message arrives while the user is accessing EDC at a data terminal, the user can retrieve the message by:

- Using an EIA terminal to display the message
- Using a BCT terminal to display the message
- Letting another user display the message via either an EIA or BCT terminal.

Unified Messaging as Viewed By an MCS User

Table 130-B shows the interactions while sending or receiving messages using MCS.

TABLE 130-B. MCS Interactions With Other Unified Messaging Services

MCS Users		
Service	**MCS Users Can Receive Mail From:**	**MCS Users Can Send Mail To:**
EDC	Recipient must be an MCS and EDC subscriber. Message less than or equal to 296 characters go to subscriber's message box. If more than 296 characters, they go to sender's assigned printer. An MCS Agent can receive itinerary updates from EDC subscribers.	Recipient must be an MCS and EDC subscriber. Messages left in the MCS mailbox can be auto- or demand-forwarded to the EDC mailbox.
LWC	Automatically places mail in the MCS mailbox Messages may be forwarded to the EDC mailbox.	Recipient must have Integrated Leave Word Calling on the switch.
AUDIX	Recipient must be an MCS and AUDIX subscriber. The AUDIX message remains in the AUDIX mailbox, and the MCS user receives the notification, "You have voice mail waiting."	Recipient must be an MCS and AUDIX subscriber. The MCS message remains in the MCS mailbox and the AUDIX user receives the verbal notification, "You have new Electronic Text messages."
MCS*	Complete MCS functionality.	Complete MCS functionality.
UNIX system	Same as EDC Interactions.	Same as EDC Interactions.
* Private documents and messages will not be sent to the MCS printer. Users must call the MCS Agent.		

The MCS user can be alerted of new messages by the MWL.

The MCS user has several methods of retrieving messages. At the end of each method of retrieval is the notification of whether there is mail from either AUDIX or EDC. The methods of retrieving messages are:

- Printing the message
  - Scheduled: the printing of the messages at a specified time of the day. If Auto Print is used, no notification of AUDIX or EDC messages is provided.
  - Demand the printing of the messages by dialing into the MCS.
- Calling the MCS Agent for verbal delivery of messages
- Via an alternate messaging service (such as, EDC, UUCP). The user can activate forwarding to these services, and MCS messages can be waiting in the alternate service.
- Using the digital display available on any display capable device (such as a 515 BCT, 7406D, or 7506)
  - Only messages less than 40 characters can be retrieved, otherwise the user has to call their MCS Agent.
- Using an EIA terminal to retrieve the message
- Using auto or demand forward of the message from MCS to EDC's mailbox.

Unified Messaging as Viewed From an AUDIX User

Table 130-C shows the interactions while sending or receiving messages using AUDIX from any touch-tone voice terminad anywhere in the world.

Users are alerted to new messages through the MWL.

TABLE 130-C. AUDIX Interactions With Other Unified Messaging Services

AUDIX Users		
Service	**AUDIX Users Can Receive Mail From:**	**AUDIX Users Can Send Mail To:**
EDC	Recipient must be an AUDIX and EDC subscriber. An EDC message remains in the EDC mailbox and the AUDIX user receives the verbal notification, "You have new Electronic Text messages."	Recipient must be an AUDIX and EDC subscriber. An AUDIX message remains in the AUDIX mailbox, and the EDC user receives the notification "AUDlX=y" on the status line of the terminal, "You have voice mail" on the message line of the terminal, and an audible alert at the terminal.
LWC*	Automatically places message in the AUDIX mailbox. Message is delivered in its entirety.	Recipient must have Integrated Leave Word Calling on the switch.
AUDIX	Complete AUDIX functionary.	Complete AUDIX functionality.
MCS	Recipient must be an AUDIX and MCS subscriber. MCS messages remain in MCS mailbox and AUDIX users receive a verbal message, "You have new MCS messages."	Recipient must be an AUDIX and MCS subscriber. AUDIX messages remain in the AUDIX mailbox and MCS users receive notice, "You have voice mail."
UNIX system	Same as EDC Interactions.	Same as EDC Interactions.

* Leave Word Calling on an AP stems messages on the AP in the MCS mailbox. Integrated Leave Word Calling stems messages in the switch's database.

Hardware Requirements

As with User Operations, each of the messaging features are services have their own hardware requirements. Unified Messaging has no special hardware requirements apart from those of the messaging features and services used.

Feature Administration

No separate administration is needed for Unified Messaging. Unified Messaging relies upon the administration of AUDIX Call Coverage, and Leave Word Calling at the switch and EDC Mail and MCS at the AP.

Notes:

Uniform Call Distribution

Description

The use of UCD (Uniform Call Distribution) groups provides an economical alternative to Direct Inward Dialing for departments that receive a high volume of incoming calls. Selected terminal users can be organized into a group, and the group is accessed by an LDN (Listed Directory Number).

This feature reduces call-completion time and attendant assistance for many incoming calls. There is one listed directory number for the group. Calls to a UCD group can be via Direct Inward Dialing, non-Direct Inward Dialing, private-switched network, and dial repeating or automatic tie trunks, as well as from terminals or attendants. All calls to the group extension, other than attendant-originated calls, are initially directed to the UCD queue associated with the group.

Feature History and Development

This feature was first available on System 85 in Release 1.

The UCD feature is not available on switches after Release 2, Version 1. This feature was replaced by the EUCD (Enhanced Uniform Call Distribution) feature and the ACD (Automatic call Distribution) feature.

> **NOTE:** See Appendix B for a tabular comparison of the various call distributors provided in Release 2 System 85 and DEFINITY Generic 2.

LDN (Listed Directory Number)

A listed directory number links to a UCD group by associating the LDN with the extension number of the first or controlling terminal in the group. The controlling or primary for the group is the first terminal, and the first terminal provides control functions such as call forwarding for the group. The switch directs incoming calls for the listed directory number to a group queue.

Hunting

A circular hunt routine extends the call to a group member. The hunt sequence starts with the extension number after the last extension to receive a group call. The hunt then proceeds to test all extension numbers in the hunt sequence. If all terminals are busy, the call remains in queue, and the hunt routine runs again in 2 seconds.

UCD Terminals

Each terminal (including the controlling terminal) in a UCD group can receive calls either as a group member or as an individual terminal. For calls originating within the switch, a unique group extension number (the associated extension number) identifies the group.

For incoming calls (call originating outside the switch), the type of incoming trunk identifies the UCD group to which a call will be routed:

- For automatic-in type trunks, the call routes to the UCD group on which that trunk group terminates.
- For dial repeating type trunks, the call routes to the group dialed.

This is similar to the way DID calls complete to individual terminals (including individual members of a UCD group).

Busy-Out Function

Individual members can control the availability of their terminals for group calls. To busy-out a single terminal, the group member goes off-hook, dials the UCD Terminal Busy access code, and goes on-hook.

The controlling terminal can make the entire group unavailable for group calls. To busy-out the group, the controlling terminal user goes off-hook, dials the UCD Group Busy access code, and goes on-hook.

Recorded Announcement

When there is a delay in completing a call, an optional recorded announcement informs the calling party of the delay.

This delay announcement can be repeated (Procedure 275, Word 4). When repeated, the calling party hears silence between cycles on the announcement machine.

FADS (Force Administration Data System)

The Force Administration Data System (FADS) feature provides incoming call traffic information.

User Operations

The following are the user operating procedures for this feature.

To Busy-Out an Individual Group Member:

1. Go off-hook. [Dial tone]
2. Dial the UCD Terminal Busy access code. [Confirmation tone]
3. Go on-hook.

To Busy-Out an Entire Group From the Controlling Terminal:

1. Go off-hook. [Dial tone]
2. Dial the UCD Group Busy access code. [Confirmation tone]
3. Go on-hook.

Considerations

Status Indicators

Status indicator lamps (provided by SSI [System Status Indicator] units) are available as queue warning level, trunk status, or system reload indicators. These units can be desk or wall mounted.

As many as 128 trunk status lamps can be provided with the switch to display the status of trunks terminating to DDC groups or UCD groups. The status lamp indicates three separate trunk states: busy (lamp on), idle (lamp off), and alerting (fluttering lamp).

A queue warning level lamp lights when the number of calls waiting in queue exceeds a preset Warning level between 1 and 31.

Power Failure Interrupts

After a power interruption, an automatic reloading occurs of switch translations from the tape into memory. The system reload indicator lights. All group members are unavailable for group calls after a tape reload. Each group member must dial the UCD terminal idle access code in order to receive group calls.

Recorded Announcement

If the recorded announcement is provided and there is more than one UCD or DDC group assigned in the switch, all groups have the same recorded announcement

When the 13A announcement set is used, the Recorded Announcement option for Intercept Treatment feature and the delay recorded announcement for UCD can use separate channels on the same announcement machine.

Message Center

The Message Center feature uses UCD groups. Hence, the use of UCD groups by Message Center decreases the number of UCD groups available for other UCD functions.

Interactions With Other Features

The following System 85 features affect or are affected by the operation of this feature.

Attendant Call Waiting

An attendant call to a UCD group does not queue. The switch attempts to complete the call ahead of any calls that may be in queue. If no idle line is found in the group, the call waits on the controlling terminal if Attendant Call Waiting is provided. However, when an attendant places a call to a UCD individual terminal and that terminal is busy, the call waits on the busy individual terminal if Attendant Call Waiting is provided.

Attendant Direct Extension Selection With Busy Lamp Field

An attendant can use the appropriate DXS (Direct Extension Selection) buttons to place or extend calls to the listed directory number of a UDC group. However, since a group's queue is never really "busy", the BLF (Busy Lamp Field) lamps adjacent to these DXS buttons are never lit.

Automatic Callback

Any UCD group member can use the Automatic Callback feature. When Automatic callback is activated toward a UCD group number, callback occurs only when the UCD controlling terminal and the calling terminal becomes idle.

Busy Verification of Lines

An attendant can use the Busy Verification of Lines feature to check the busy/idle condition of a terminal in a UCD group. However, this feature cannot check for a line "made busy" to group calls. If activating Busy Verification of Lines toward a UCD group number, only the controlling terminal line is verified (no hunting takes place).

Call Coverage

Coverage cannot be assigned to an associated extension number of a UCD group. However, the switch allows the assignment of coverage to individual extensions of group members for calls addressed directly to that extension

The switch cannot redirect a call to coverage after the UCD feature has distributed the call to a group member. Therefore, the UCD group must be assigned as the final point in the coverage path. When assigning a UCD group as the final point in a coverage path, the group number is the assigned coverage point.

CDR (Call Detail Recording)

On an incoming UCD call, the trunk identification number is recorded as the calling number.

Call Forwarding—Follow Me

Call Forwarding—Follow Me, when activated for a UCD group, routes all UCD calls to a designated terminal, the attendant queue, the centralize attendant queue, or to another UCD or DDC group queue immediately after dialing. If a call is already in queue when this feature is activated, the call remains in queue for 7 seconds before forwarding.

- Only the controlling terminal or attendant can activate or deactivate Call Forwarding—Follow Me for the UCD group. Noncontrolling group terminals cannot activate or cancel Call Forwarding—Follow Me even if authorized by their extension class of service.
- Only calls to the UCD group number forward when activating Call Forwarding—Follow Me. Calls to an individual terminal or controlling terminal number do not forward.

- The stop hunt option should be provided in the extension class of service for each UCD group extension. Otherwise, a call forwarded to an individual UCD member's extension is treated as a call to the UCD group (hunting is used).

Call Waiting

When Call Waiting is assigned to an individual UCD group terminal, calls to the terminal are allowed to wait if the called terminal line is busy. The group terminal user can be connected to the waiting call by going on-hook, whereby the terminal is alerted and connected to the call upon answer. These calls have preference over UCD calls in queue waiting to be answered. Do not assign this feature to the controlling extension number of a UCD group.

DDC (Direct Department Calling)

If the switch uses UCD and DDC, the memory size of the switch and the call traffic requirements determine the number of combined groups and voice terminals per group. The switch provides up to 28 groups with a maximum of 40 voice terminals per group. An extension number cannot belong to both a UCD and a DDC group; the groups are mutually exclusive.

DCS (Distributed Communications System)

In a Distributed Communications System environment, calls to a UCD group cannot forward between nodes.

FADS (Force Administration Data System)

The FADS feature only measures the first 12 groups. For maximum traffic measurement efficiency, it is recommended that the first 12 UCD/DDC groups be assigned as UCD groups. Groups 1 through 4 should contain 40 members, and groups 5 through 12 should contain 24 members.

Hold

Going on-hook to return to a held call may result in the voice terminal being connected to the next call in the distribution-group queue. Using the Call Hold access code is the recommended method for returning to a held call.

Hunting

Group members should have the stop hunt option active. This prevents calls forwarded to an individual's extension number from hunting.

Main/Satellite/Tributary

Extension numbers that use extension number steering cannot be the controlling extension for a UCD group.

Priority Calling

When activating Priority Calling toward a UCD group terminal, the call waits on that terminal. When activating Priority Calling toward a UCD associated extension number, the call waits on the controlling terminal (the call does not enter the group queue).

Restriction—Attendant Control of Voice Terminals

If any Attendant Control of Voice Terminals restiction (Outward, Terminal-to-Terminal Only Calling, Termination, or Total) is assigned to a UCD controlling terminal or a UCD group extension, then the restriction assigned is applied to the entire UCD group.

Restricting Feature Use

Performing the busy-out procedure removes a group member from the UCD group.

The controlling terminal can make the entire UCD group unavailable for UCD calls by dialing an access code.

An attendant can restrict access to a UCD group by activating the Attendant Control of Voice Terminals feature for the group number.

Hardware Requirements

The UCD feature requires the following hardware for a traditional module.

- 30A8 system status indicator lamp panel to display the queue warning status for eight UCD groups
- SN241 contact interface (eight circuits per circuit pack)
- SN231 auxillary trunk for delay recorded announcement (four circuits per circuit pack)
- 13A announcement set to provide the recorded announcement (only one channel can be used for UCD/DDC) or KS-65270 digital announcer (single-channel announcement set) or KS-65272 4-channel digital announcer.

Feature Administration

Assignment of the UCD feature is on a per-trunk group basis and on a extension class of service basis. The customer can partially administer this feature using the SMT (System Management Terminal) or the TCM (Terminal Change Management) feature. This feature can also be administered using the Manager IV.

The following are the applicable MAAP and SMT procedures.

MAAP and SMT Procedures — Uniform Call Distribution

Procedure	Word	Purpose	SMT
000	1	Administers the equipment location and class of service of each member's extension.	Yes
000	2	Administers hunt-to assignments to each member's extension.	Yes
001	1	Assigns associated extension numbers to the primary extension.	Yes
010	1	Assigns UCD/DDC membership to an extension class of service. Also, use this procedure to assign the stop hunt option to a UCD/DDC member's class of service.	Yes
025	1	Administers the characteristics of the UCD/DDC group.	Yes
025	4	Administers UCD/DDC trunk indicator lamps.	Yes
100	1	Administers the trunk type and trunk-group assignments for UCD delay recorded announcement, contact interface, and queuing trunk groups. The applicable trunk-type encodes include: 6 Special queue 65 SN241 Contact Interface 68 UCD delay recorded announcement.	No
115	1	Assigns the UCD/DDC group terminations to incoming trunk groups.	No
150	1	Assigns the delay recorded announcement trunk to an SN231 equipment location.	No
155	1	Assigns the contact interface trunk group to an SN241 equipment location.	No
204	1	Designates the desired alphanumeric display for UCD/DDC calls that reach an attendant.	No
275	4	Assigns the UCD/DDC delay recorded announcement to the system class of service.	Yes

(Continued)

MAAP and SMT Procedures Uniform Call Distribution *(Continued)*			
Procedure	**Word**	**Purpose**	**SMT**
350	1	Assigns the first digit of the feature dial access codes and extension number groups (if required).	No
350	2	Assigns the feature dial access codes. The applicable recodes are as follows 51 UCD/DDC terminal busy 52 UCD/DDC terminal idle 54 UCD/DDC lamp test 58 UCD/DDC group busy 59 UCD/DDC group unbusy 70 UCD/DDC status toggle.	No
354	1	Administers groups of extension numbers.	No

The following are the applicable TCM path names used with the AP 16.

TCM Screens — Uniform Call Distribution	
Path Names	**Purpose**
terminal-change terminal display unit	Displays or prints the group members being monitored by the 106B display unit.
terminal-change class-of-service attributes	Assigns UCD/DDC membership to an extension class of service. Also, use this screen to assign the stop hunt option to a UCD/DDC member's class of service.
terminal-change extensions attributes	Assigns the extension class of service to an extension number, to administer the associated extension numbers to the primary extension, and to administer the hunt-to assignment to an extension.
terminal-change group call-distribution attributes	Administers the characteristics of a UCD group. Also assigns "circular" to the type-of-hunting field.
terminal-change group call-distribution members	Adds or removes members to/from a UCD group.

Visually Impaired Attendant Service

Description

The Visually Impaired Attendant Service feature provides additional devices to enable visually impaired attendants to operate the attendant console. These devices include:

- A grooved faceplate guide
- A light-sensitive pen
- A tone generator.

Refer to Figures 132-1 and 132-2 for drawings of these devices.

The attendant receives an audible tone through the headset, whenever the light pen is held over a lighted or flashing console lamp.

The tone generator provides additional distinctive audible signals, based on the ICI (Incoming Call Identification) translations, to identify the type of incoming call. Table 132-A describes the six different audible signals associated with the ICI display. See the Attendant Display feature for more information on ICI.

TABLE 132-A. Distinctive Audible Signals

Alerting Rate	Alerting Signal	Signal Meaning
Rate 1	Two burst, 0.15 seconds each	Incoming LDN Call
Rate 2	Repeated short bursts, 50 milliseconds each	Dial "0" Call
Rate 3	Two short bursts, repeated after 0.2 seconds	Attendant Recall
Rate 4	Three short bursts, repeated after 0.1 seconds	ICI Option
Rate 5	Short burst followed by long burst (0.25 seconds)	ICI Option
Rate 6	Short burst, long burst, short burst	ICI Option

Feature History and Development

This feature was first available for System 85 in Release 1. There have been no changes to this feature since Release 1.

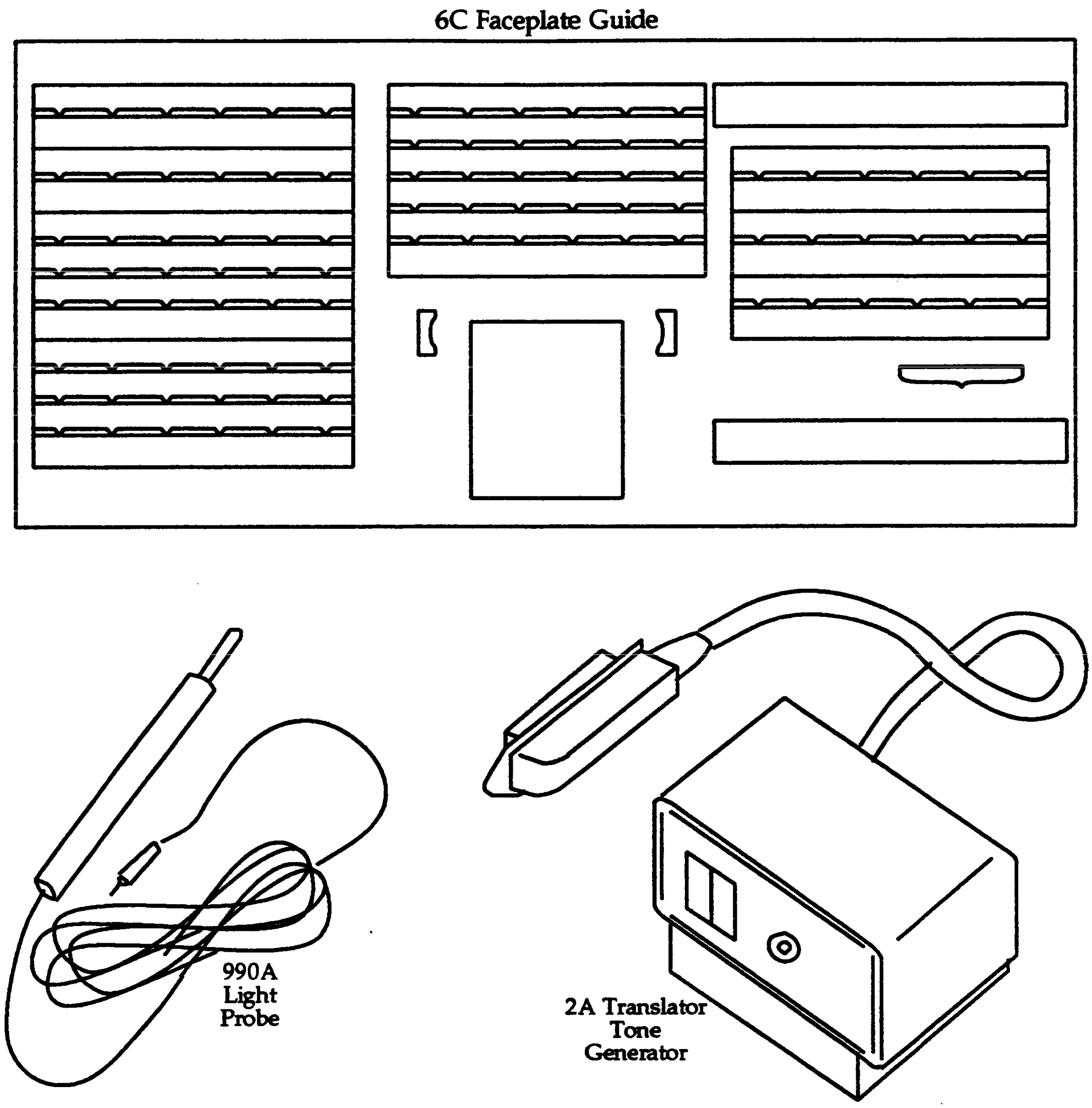


Figure 132-1. Visually Impaired Attendant Console Devices

User Operations

The following are the user operating procedures for this feature.

To Extend an Incoming LDN (Listed Directory Number) Call:

1. Distinctive audible tone identifies the incoming call.
2. Press the **[ANSWER]** button. [Ringing stops.]
3. After obtaining the desired extension, press the **[START]** button. [Calling party is split from the console, distinctive signal stops, dial tone.]

4. Dial the desired extension number or trunk-group access code,

or

Use DXS (Direct Extension Selection) or Direct Trunk Group Selection if available. [Call-progress tone]

5. Depending on calling instructions:

Press **[RELEASE]** to disconnect from the call,

or

Press **[HOLD]** to hold the call on the console. [Call-progress tone is silenced.]

To Answer a Timed Reminder Call:

1. Timed Reminder is identified by the distinctive audible signal.
2. Pass the light pen slowly over the face plate groove for the HOLD, BUSY, or RING lamps to find the active loop lamp. [On-off tone is heard when the light pen is over the flashing indicator lamp.]
3. Press the associated loop button. [Timed Reminder tone stops.]

Considerations

Selecting the Attendant Display

While the Attendant Display—ICI (Incoming Call Identification) provides up to 30 separate messages, the distinctive ringing available to the Visually Impaired Attendant feature is limited to the first five of these. Three of these are fixed by the system, and two can be selected by the customer. All other signals default to the sixth, common ringing mode. Care must be taken in administering the Attendant Display feature to provide maximum effectiveness to the distinctive ringing signals and avoid confusion.

Light Pen Sensitivity

The light pen may not detect a flashing indicator lamp if the pen is moved too quickly across the faceplate of the console. Therfore, the attendant should move the pen at a relatively slow rate.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features are affected by the operation of this feature.

Malicious Call Trace

A visually impaired attendant should not attempt to trace a malicious call. The controlling attendant must be able to see and record the ICI messages. Also, the controlling attendant must be able to read and communicate the information from the cross-reference of trunks.

Restricting Feature Use

The CODE RING ON-OFF switch enables or disables coded ringing.

The AUD OFF button disables both the Timed Reminder tone and the calls waiting tone.

Hardware Requirements

The Visually Impaired Attendant Service feature requires the following additional or special hardware for a traditional or universal module.

- A 990A light sensor
- A 2A translator
- A 6C grooved faceplate guide.

Feature Administration

The Visually Impaired Attendant Service feature is hardware activated. No separate assignment or administration is required.

Wide Area Telecommunications Service Access

Description

The WATS (Wide Area Telecommunications Service) Access feature can provide users with cost-effective access into the WATS network. Outgoing call service to a predetermined geographical area or areas is provided on a reduced-cost basis compared to ordinary toll service. A similar but separate service is provided for incoming calls by 800 Service Access. This feature (800 Service) has also been known as INWATS. Incoming 800 Service calls can be directed to the attendant queue, to an ACD, EUCD, or UCD/DDC split, to a specific System 85 or DEFINITY Generic 2 extension, or be provided use of the Remote Access feature.

Calls using WATS Access can be originated by local terminal users or Remote Access users provided restrictions are not applied, or they can be set up using attendant assistance.

Feature History and Development

This feature was first available for System 85 with Release 1 as part of Public Network Access. It was reintroduced as WATS Access in Release 2, Version 3. Otherwise, this feature has remained unchanged since its introduction.

User Operations

The following are the user operating procedures for this feature.

DOD Calls From a Local Terminal:

1. Go off-hook. [Dial tone]
2. Dial the WATS trunk-group access code. [Second dial tone]
3. Dial the final destination number.

Attendant-Handled Calls

If ADTGS (Attendant Direct Trunk Group Selection) is available:

Press the appropriate WATS trunk-group button.

If ADTGS is not available:

1. Press the **[START]** button. [Dial tone]
2. Dial the WATS trunk-group access code.

Depending on calling instructions:

3. Press the **[HOLD]** button to remain connected to the call and finish dialing the final destination number (or set up a conference call if desired),

   or

   Press the **[RELEASE]** button to drop from the call and allow the calling party to finish dialing.

Considerations

Tariff Concepts

- Offers bulk-rate discounts for long-distance calls to specified service areas known as WATS bands.
- WATS Bands 1 through 6 are defined for each state.
- Cost per hour of usage decreases as a total usage per access line increases a tapered tariff. Discounts are available for evening and night usage.
- Total monthly cost is a function of the WATS Band(s) selected, number of access lines, total call usage, and time of day calls were placed.
- The WATS tariff provides for a minimum of ***two*** lines placing WATS calls.

Advanced 800 Service Compatibility

The enhanced capabilities of Advanced 800 Service can be applied to incoming 800 Service calls routed to a System 85 or DEFINITY Generic 2. These capabilities provide greater call-routing flexibility and control for subscribers to Advanced 800 Service.

Some of the Advanced 800 Service offerings include Single Number Service, Area Code Routing, Time Manager, Day Manager, Time and Day Manager, Call Allocator, Call Prompter, Routing Control Service, Command Routing, and Courtesy Response. Contact your AT&T Communications representative for more information about these Advanced 800 service offerings.

Hard and Soft Processor Swaps

Stable calls over WATS trunk groups endure a hard processor swap. However, calls cannot be placed over WATS trunk groups during a hard processor swap.

Interactions With Other Features

The following System 85 and DEFINITY Generic 2 features affect or are affected by the WATS Access feature.

DOD (Direct Outward Dialing)

Using outgoing (or 2-way) WATS trunks, the DOD feature is used by System 85 or DEFINITY Generic 2 users to call stations in the WATS Access service area over WATS Access trunks.

AAR (Automatic Alternate Routing)/ARS (Alternate Route Selection)

The WATS Access feature can be set up to work in several different ways. These include direct dialing to the WATS trunk group, Attendant Control of Trunk Group Access, or on System 85 and Generic 2.1 switches, WATS Access can be through an AAR or ARS routing pattern.

The "User Operating Procedures" are generally applicable; however, if access to WATS service is provided through AAR or ARS routing patterns, the AAR or ARS access code is dialed. With AAR or ARS routing, the user is not aware of whether a WATS trunk is being accessed or some other facility (such as, Private Network or FX Access facility).

Look-Ahead Interflow

From a sending (or tandeming) switch, calls can interflow over the public network using ISDN—PRI trunk groups assigned as WATS Trunk Type 27. When this is done, these interflow calls will succeed if every public-network switch between the sending switch and the destination telephone number is ISDN—PRI-capable.

At a sending switch, incoming 800 Service calls can terminate to a VDN with a vector assigned that contains commands for Look-Ahead Interflow. When this is done, these incoming 800 Service calls will interflow normally (that is, according to the commands in the sending and eceiving vectors).

Multiple LDN (Listed Directory Number)

The Multiple LDN feature can be used to direct incoming 800 Service Access calls to the attendant queue.

Personal Central Office Line

Using incoming (or 2-way) WATS trunks, the Personal Central Office Line feature can be used by a caller in the 800 Service area to reach a specific station on the System 85 or DEFINITY Generic 2 switch.

Remote Access

The Remote Access feature can be used to provide access to the WATS feature, and Remote Access services can be provided to incoming calls on 800 Service trunks.

WCR (World Class Routing)

The WCR feature replaces the AAR and ARS features on DEFINITY Generic 2.2 switches. The WCR feature is fully compatible with the WATS Access feature and can be used to route calls over WATS (including 800 Service) trunks. When WATS trunks are included in WCR routing patterns, the appropriate WCR network dial access code is used for WATS Access rather than a trunk dial access code.

Restricting Feature Use

The Restrictions—Attendant Control of Trunk Group Access feature allows the attendant to restrict DOD (Direct Outwad Dialing) by terminal users to WATS Access trunk groups.

The Attendant Control of Voice Terminals, the Voice Terminals Restrictions, and the Miscellaneous Trunk Restrictions features can limit access to WATS by specific terminals or terminal with a specific class of service.

If WATS Access is obtained via the AAR or ARS features, the FRL (Facilities Restriction Level) and Authorization Codes feature can also be used to restrict the accessibility of WATS Access trunks.

Hardware Requirements

The following specific hardware is required to support the WATS Access feature.

For Traditional Modules:

- SN230 trunk circuit pack (four circuits per circuit pack).

For Universal Modules:

- TN747B trunk circuuit pack (eight circuits per circuit pack).

Feature Administration

Assignment of the WATS Access feature is on a per-system and per-trunk group basis.

On System 85 switches, this feature is assigned using the MAAP (Maintenance and Administration Panel). The customer can partially administer this feature using the SMT (System Management Terminal) and the TCM (Terminal Change Management) feature.

On DEFINITY Generic 2 switches, this feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

ADMINISTRATION PROCEDURES — WATS ACCESS			
PROCEDURE	**WORD**	**PURPOSE**	**SMT**
010	3	Assigns restrictions including miscellaneous trunk restriction group, FRL, Toll Restriction, and ARS Toll Restriction to a extension class of service.	Yes
100	1	Assigns trunk-group dial access code, route advance sequence, and trunk type. The applicable trunk-type encodes include: 26 1-way automatic incoming attendant-completing 27 1-way outgoing DOD 28 1-way out DOD with party test	No
101	1	Assigns the characteristics of a WATS Trunk Group.	No
102	1	Assigns Miscellaneous Trunk Restrictions (if desired).	Yes
103	1	Assigns network features associated with a trunk group including FRL, TCM, and conditional routing TCM.	Yes*
115	1	Assigns incoming termination of a trunk group (800 Service).	No
150	1	Assigns trunk features including trunk-group number and recoded announcement characteristics to an equipment location (physical trunk).	No
202	1	Assigns trunk group to Attendant Direct Trunk Group Selection buttons and trunk-group warning levels.	No
204	1	Assigns the ICI (Incoming Call Identification) for the Attendant Console (for 800 Service calls).	No
279	1	Assigns the type of service that WATS calls using ISDN facilities receive.	N/A
* Display only procedure for the SMT.			

The following is the applicable TCM path name used with the AP 16.

TCM SCREEN — WATS ACCESS	
PATH NAME	**PURPOSE**
terminal-change class-of-service	Assigns restrictions including miscellaneous trunk restrictions, FRL, Toll Restriction, and ARS Toll Restriction to a extension class of service.

Notes:

World Class Routing

Description

The WCR (World Class Routing) feature provides flexible call routing for customer network calls (except AUTOVON and Main/Satellite/Tributary) through a single feature. WCR combines the routing capabilities of the previous network routing features AAR (Automatic Alternate Routing) and ARS (Automatic Route Selection) and relaxes most of the constraints set on these earlier features.

The World Class Routing feature performs digit analysis and routing in a general way, without hard coded assumptions about the numbering plan used. Most networking capabilities are now shared by up to seven routing networks. These routing networks are distinguished by the access code used and the numbering plan for the associated physical network. A call from one network can overflow to any other network by converting the digits received to the form used by the target network.

Feature History and Development

The earlier network routing features (ARS and AAR) were based on the North American Numbering Plan and the customer's private network Uniform Numbering Plan. These numbering plans assume certain restrictions in the use of numbers and patterns.

The rapid expansion of telecommunications networks in North America, coupled with the increased use of international telecommunications, premise to make these numbering plans and their inherent assumptions invalid early in the 1990s. The World Class Routing feature replaces both the AAR and ARS features and remedies the shortcomings developing these network routing features.

The WCR feature is first introduced with the DEFINITY Communications System, Generic 2.2, Issue 1.0.

Functional Configuration

The WCR feature is based on the functional configuration shown in Figure 134-1. The first module shown (Internal Digit Analysis) is not part of WCR as such. Rather, it is part of the switch internal dialing plan and is used to identify WCR network dial access codes and to route network calls to the WCR feature for processing. The four other operational modules shown comprise the WCR feature. Each of these WCR modules is described separately.

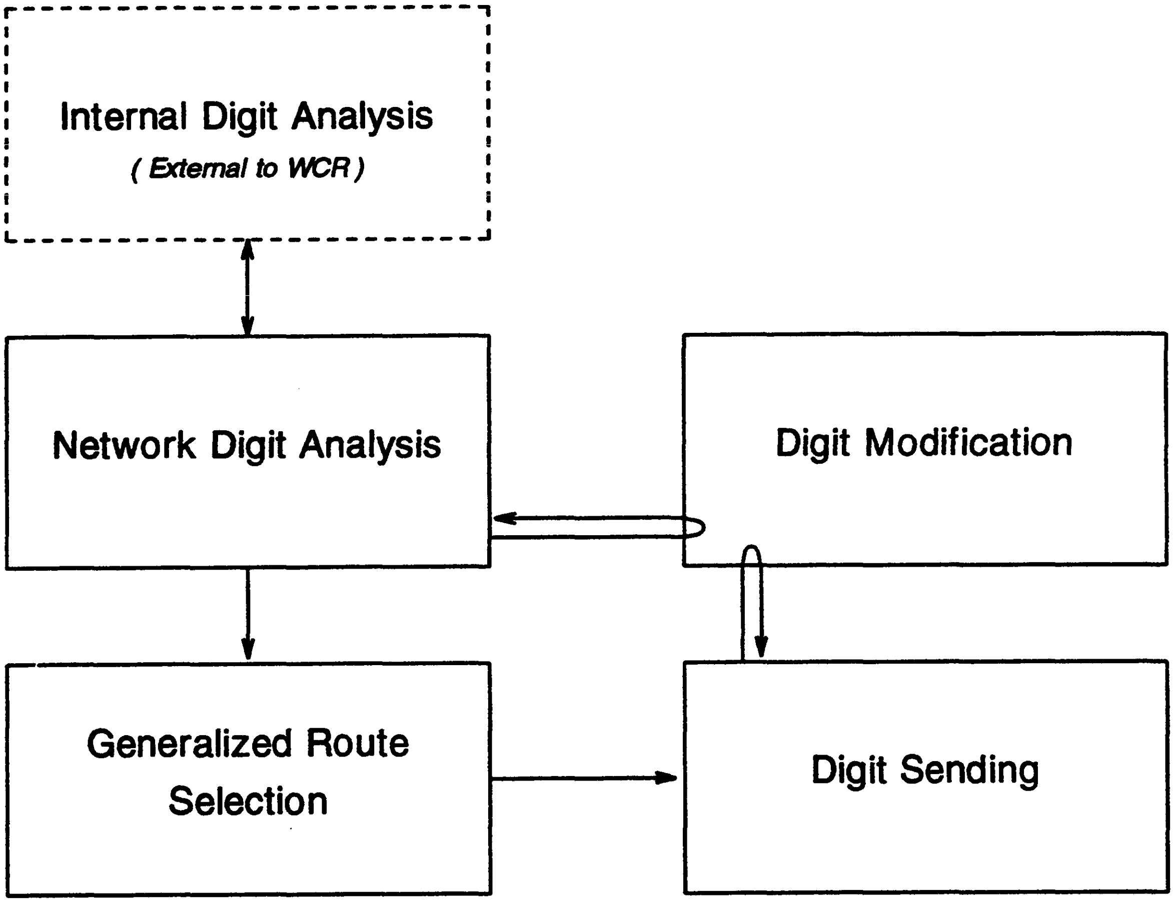


Figure 134-1. Routing Functional Configuration

Digit Analysis

The digit analysis operations are the first functional modules to deal with the dialed digit string. There are two forms of digit analysis.

Internal Digit Analysis

Internal digit analysis is part of the switch internal dialing plan rather than the World Class Routing feature. Internal digit analysis handles dialed numbers that match the local or internal numbering plan of the switch. Internal digit analysis deals with local extension numbers, extension number steering when applicable, trunk access codes, and feature access codes. Internal digit analysis does not handle routing network calls except when the UDP (Uniform Dial Plan) or the ENP (Extension Number Portability) feature is involved These cases are discussed later under Considerations and Interactions with Other Features.

Network Digit Analysis

World Class Routing network digit analysis deals with digit strings that follow a network dial access code and match digit string patterns in the network being used. The final result of network digit analysis is a VNI (Virtual Nodepoint Identifier), which is used to select a route to the call's final destination. Seven digit analysis tables are available to accommodate the needs of routing network numbering plans.

Digit Modification

The digit modification module is used as needed by either the digit analysis module or the digit sending module. Digit modification converts a set of digits (received from one of the other modules) into a digit stream that conforms to the expectations of the next operation. For example, network digit analysis may send a public network number to digit modification for conversion into a private network number to be routed over private network facilities. The digit sending module may send a private network number to digit modification for conversion to a public network number so that it can be sent to the local central office as a subnetwork trunking operation from the private network.

Generalized Route Selection

The generalized route selection module accepts the VNI from digit analysis and uses that VNI, along with other switch and call related parameters (such as time-of-day plan, FRL [Facility Restriction Level], and Bearer Capability) to select the trunk group to be used for all completion.

Digit Sending

The digit sending module controls the way digits are sent over the trunk group selected by generalized route selection module. Digits can be sent as dial pulse or touch-tone signals or as ISDN messages. Certain digit strings (such as dial access codes and interexchange carrier access codes) may or may not be sent, depending on the trunk group selected. The digits sent can also be grouped with timed pauses between groups when required by the distant switch. When applicable, digit sending also creates a call detail record for a given call.

Network Digit Analysis

Network digit analysis handles digit strings that follow a WCR network DAC (Dial Access Code). Network digit analysis selects the VNI (Virtual Nodepoint Identifier) for the call based on the administered characteristics of the digit string and network parameters. A VNI is an administered value, assigned to a call during digit analysis, that identifies preliminary routing for the call. The VNI is discussed in more detail later.

Digit analysis operations are shown graphically in Figure 134-2.

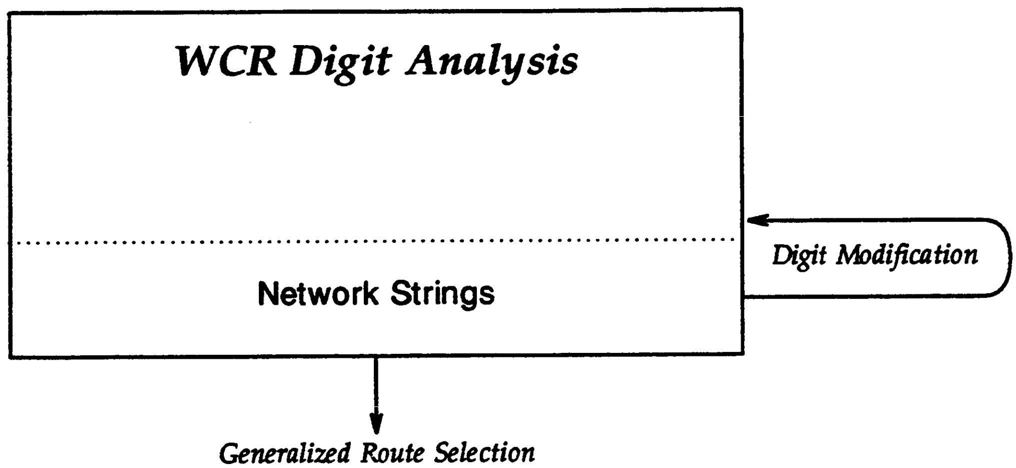


Figure 134-2. Digit Analysis

Networking Structure

The WCR feature provides up to eight software arrangements (or numbering plan structures) called networks (numbered 0 through 7). Networks 1 through 7 are referred to as a ***routing network*** because they use administrable routing patterns. Any of the routing networks (networks 1 through 7) can be administered to have the capabilities of both of the earlier networking features (AAR and ARS).

Network 0

Network 0 provides an interface between the World Class Routing feature and the internal dial plan. Network 0 is not the internal dial plan and is not considered a routing network. Unlike the other WCR networks, network 0 does not have a DAC, and network 0 routing is not generally administerable in the same sense as the other seven networks.

Network 0 is used primarily to specify digit modification on outgoing calls for the ENP feature and for the UDP. When other networks "restart" in network 0, the network 0 translations are not used: instead, the call is sent directly to the internal numbering plan for analysis. Network 0 does not automatically come with the WCR feature but is available at an additional charge (for activating Standard Network in Procedure 276, Word 1).

Network 1

Network 1 is available for basic public network access to provide connectivity to the local central office and basic toll network services. This is roughly equivalent to the ARS feature although without the constraints and limitations that applied to ARS. Only network 1 has separate toll and non-toll dial access codes and only network 1 comes as part of the basic switch feature set (without additional charges).

Networks 2 through 7

Networks 2 through 7 can be used for call routing over private or public networking facilities and alternate interexchange carrier facilities. These networks replace earlier private networking features such as AAR. Each has one dial access code.

Networks 2 through 7 do not automatically come with the WCR feature but are available at an additional charge (by activation of Standard Network in Procedure 176, Word 1).

Network DACs (Dial Access Codes)

Each routing network (1 through 7) is identified by a DAC of from 1- to 4-digits. Network 1 actually has two DACs, one that is used for toll calls and one that is used for non-toll calls. For networks 2 through 7, the toll or non-toll characteristic of the DAC (toll permission) is based on the class of service of the originating extension, since only one DAC is available. However, attendant and incoming trunk calls are always treated as toll allowed.

For locally originated calls, the DAC used determines which network the dialed numbers are evaluated against. For incoming calls on selected trunk types, trunk group prefixing (an administered characteristic), can be used to provide the network DAC.

Network Parameters

Network parameters are the specific rules set up for each routing network in Procedure 312, Word 1. These include:

- Dial Tone Suppression

  Specifies whether or not dial tone is returned when a network is accessed (either by DAC or by prefixing on an incoming trunk call). Normally, dial tone is returned. Dial tone suppression can also be assigned in Procedure 103, Word 1, field 15. When administered in Procedure 312, it applies to all calls in the network (station, attendant, or trunk originated). When administered in Procedure 103, it applies only to incoming calls on a specific trunk group.

- CDR (Call Detail Recording) account code requirements

  Use of a CDR account code may or may not be required. For example, if an account code entry is required for a particular network, and an account code is not dialed, that network cannot be used to route that call. It may be possible to route the call to the same destination over a different network that does not require an account code.

- Toll Prefix requirements

  Specifies whether or not a toll prefix must be entered by the caller when making a toll call. Normally the digit "1" is used for this purpose (in the North American Numbering Plan). Dialing "0" or "01" can also satisfy this requirement if these prefixes are administered as *operator assistance* and *international access* string types.

Digit Strings

The key to digit analysis, and indeed World Class Routing itself, is the understanding and use of the concept of digit strings. For the purposes of the World Class Routing feature, a digit string is any group of numbers (one or more) that uniquely identifies an element of a dialed number (such as an Interexchange Carrier identification number) or an address (area code, office code, and extension). Digit strings are identified in digit analysis and used to determine what to do with the call. For example, an IXC (Interexchange Carrier) access code is used to identify the long distance carrier that should be used to route the call.

String Identifiers

Digit analysis uses string identified to break down the dialed number into its component parts. String identifiers are the elements of a digit string that uniquely identify the string or part of the string, and are administered in Procedure 314, Words 1 and 2.

A string identifier can be from 1 to 18 digits long and specifies a number of attributes and characteristics such as string length and string type. It also specifies actions such as route the call (resolve the call to a VNI) or restart digit analysis (with or without digit modification).

The most common type of string identifiers are address strings. Address strings are the primary source of information used to determine routing. String identifiers are compared to the dialed digits to determine routing. Therefore, any network call a user is allowed to make must be included in the string identifier tables. There is often more than one way to accomplish this task.

There are certain kinds of digit strings that need to be specifically identified. Reasons for this include:

- They may be stored in their own special fields in the call detail record (for example, account codes and interexchange carrier identification codes)
- They may result in special coding in ISDN messages (operator assistance, international access, and interexchange carrier access codes)
- They may need to be remembered for later toll analysis or digit sending operations (toll prefix operator assistance, international access, and interexchange carrier access codes).

The string types currently defined are:

- Account Codes
- Interexchange Carrier Access Codes
- International Calling Prefixes
- Operator Assistance Codes
- Toll Prefixes
- Address Strings.

String identifier types are given a hierarchical relationship based on the constraints of the North American Numbering Plan. Rules that specify which string type may follow which other string type are given in Table 134-A. These rules apply to all networks.

TABLE 134-A. Hierarchy of String Types

Leading String Types	Permissible Following String Types				
	IXC Code	Toll Prefix	International Access	Operator Assistance	Address
Account Code*	Yes	Yes	Yes	Yes	Yes
IXC Code	No	Yes	Yes	Yes	Yes
Toll Prefix	No	No	No	No	Yes
International Access	No	No	No	No	Yes
Operator Assistance	No	No	No	No	Yes
Address**	No	No	No	No	Yes

* If an account code is used, it must be the first string type dialed (immediately following the network DAC). Account codes can't follow any string type, including other account codes.
** Address string types can follow any string type, including another address string.

Exception Strings

Exception string tables are provided to allow for special handling of classes of digit strings such as "555" or "976" office codes for all area codes. Exception strings require matching on a set of digits that is internal (not the leading digits) to the digit string. For example, if you want to deny routing for all "976" office codes, the string identifier "***976" is administered as an exception address string of length 10, and routed to VNI 0.

The " * " character in the string identifier stands for "any digit" and in this application is called a "wild card digit." When a user dials 214-976-1234, standard network translations match on the area code 214 for routing; however, before passing control to the generalized route selection module, the exception tables are checked. When a match is found in the exception network tables, this selection overrides any found in the standard network translations. A wild card digit (*) cannot be used in the standard network translations.

Variable Length Strings

In some cases, the length of a digit string can vary between a lower and upper bound. For example, you may want to route all international calls to the same VNI. However, international numbers can range from 7- to 18-digits in length. In this case, you would define (in Procedure 314, Word 1) an address string identifier "011" with a length of seven digits (in field 9) to route (resolve) to the desired VNI. Then, in Procedure 314, Word 2, you would set the maximum length field (field 5) to 18. These two fields (field 9 in Word 1 and field 5 in Word 2) set the minimum and maximum string length. During digit analysis for this type of digit string, normal interdigital timing (10 seconds) is used. To speed up call routing the user can enter the end of dialing character (**#**) for shorter dialing strings to indicate the end of dialing. Exception strings cannot be variable length strings.

VNI (Virtual Nodepoint Identifier)

The VNI is an index number that represents the destination of a call. A VNI is conceptually similar to the node number or routing designator used with the earlier networking features (AAR and ARS). The VNI is an administered value and is not part of the dialed number (address) of the call.

A VNI can be any number between 0 and 1023. VNI 0 is a special nonrouting VNI used to route calls to intercept. Assignment of a VNI to a call (during digit analysis) is administered using Procedure 314, Word 1, Field 12. The VNI assigned to the call is used by call processing software in the generalized route selection module as part of the routing pattern selection process. For routing purposes, a VNI is associated with a call category in Procedure 317, Word 2 to obtain a pattern number. If a pattern number is not assigned to a VNI (in Procedure 317, Word 2), switch software will use the VNI itself as a pattern number in attempting to complete the call *for call category 0.* For other call categories there must be an associated pattern number or the call will route to intercept.

Administering a Network Numbering Plan

For each network being used, the numbering plan for that network must be fully specified. This is done by defining the entire set of string identifiers for each network using Procedure 314, Words 1 and 2. Since most entries will be for address strings, the basic concept of describing a numbering plan is to:

1. Define the digit strings to be identified
2. Specify what to do with each digit string once identified
3. Populate the exception network tables as needed.

Typically, you would administer an address string beginning with an area code (for example, 213) to be 10-digits long and route to VNI 22. If all other area codes from 210 through 219 are to route to another VNI, you would define an address string beginning with " 21 " to be 10-digits long and route to VNI 20. Both of these translations can reside in the same network. Calls to area code 213 will route to VNI 22, and calls to all other area codes beginning with 21 will route to VNI 20. Further, if a specific office code within area 213 requires special routing it can simply be added to the string identifier tables for the network. For example, "213422" with a length of 10-digits, routes to VNI 40. When a user dials the number 213-422-1234:

- When the user dials "21," call processing software sees three possible choices for routing.
- When the third digit "3" is dialed, two possible routing choices remain.
- If the next three digits dialed are "422," only one routing option remains. For that matter, if the next three digits are not "422" only one routing option is also available, or if the fourth digit is not "4," the number of options is reduced to one.

  The longest matching string identifier is finally chosen.

There are occasions when two digit strings are distinguishable only by the number of digits being dialed. For example, 818 may be both an area code (length 10) and office code (length 7). In this case, the interdigit timing used to collect the 8th digit can be reduced to 4-seconds.

> **NOTE:** The interdigit timing interval for variable length strings is administered in Procedure 285, Word 1, field 3. The possibilities range from 2-seconds to 20-seconds, in 2-second increments. If not administered, the default is 10-seconds. This administration does not affect the standard interdigit timing (10-seconds) used for digit collection.

If an 8th digit is dialed, the area code VNI is chosen and if an 8th digit is not dialed, the office code VNI is selected. The caller can enter the end of dialing character (#) to avoid waiting for the interdigit time out.

Further guidelines for specifying the network dialing plan and a full description of the options available are given in separate documents (Administration Procedures, 555-105-506, Administration of Features and Hardware, 555-105-507, and World Class Routing Implementation Notes, 555-105-531).

A dialed digit string could include a combination of almost any of the above elements. For example, a dialed digit string might be in the form:

9001 (Dial Tone)

1 213 422 1234

This dialed number breaks down as follows:

- DAC (Dial Access Code)

  The leading group of digits "9001" is the network dial access code and is evaluated as part of the internal dialing plan of the switch. The network DAC identifies the network against which the remaining digits are evaluated. Each network has its own set of string identifiers, which define the numbering plan for that network.

- Toll Prefix

  The next digit "1" is the toll prefix This may or may not be required depending on how the switch is administered (that is, the switch administrator can set this as being required or not required for toll calls in Procedure 312, Word 1, field 4).

- Address String

  The next group of digits "213" is the area code. This is part of the address and (because it is an area code) implies that this call will use public network circuits. The area code, combined with the local exchange office code (422) and the extension number (1234), constitutes the address for the call.

In defining a string identifier, you should use the minimum number of digits needed. For example, to determine routing requirements for an address string, an area code or area code plus office code will be normally be sufficient.

Network Numbering Plan Capacity

All WCR networks share a limited pool of string identifier resources (table entries). The more complex the administration for one network, the fewer table entries remain for use in other networks. The total number of entries possible depends on the number of digits in each string identifier and the degree to which string identifiers look like other string identifiers. It would be wasteful to administer every string identifier as a 6-digit area code and office code combination. As a guide to the capacity available, the following set of string identifiers can be accommodated:

10 Operator and Service Codes

100 Interexhange Carrier Codes

256 Area Codes

64 6-digit Expansions of Area Codes

800 Local dialing (Home NPA) Office Codes

999 Private Network Location Codes

128 18-digit International Numbers.

Digit Modification

Calls reach digit modification only if they are referred there by one of the other call processing modules. Digit modification supports two World Class Routing functions network crossover, and subnetwork trunking. The operation of the Digit Modification module is shown graphically in Figure 134-3.

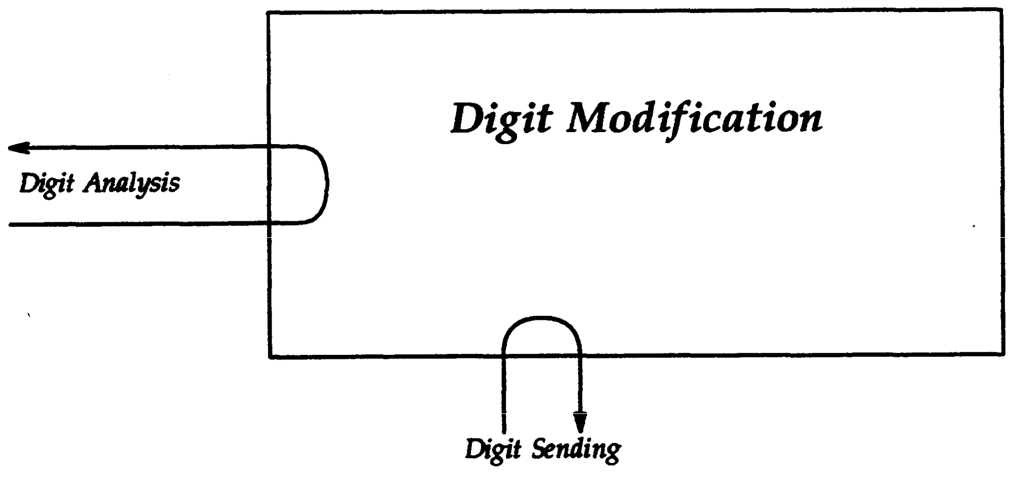


Figure 134-3. Digit Modification

Referral Sources

The digit modification module receives digit strings for conversion either from the digit analysis module or from the digit sending module.

Referrals from Digit Analysis

Digit strings are referred to digit modification from digit analysis when ***network crossover*** is required. With the ***standard network*** option selected in Procedure 276, Word 1 (this is an added cost option), any network call can crossover to any other network (including crossing back to the original network). Crossover between networks must be administered. That is, crossover does not occur automatically. The maximum number of crossovers for a single call is limited to seven. This is to guard against inadvertent looping through a set of networks.

When network crossover is required, the target network will probably need a different numbering plan structure. For example, in the case of a crossover from the public network to a private network, the area code will be deleted (and the IXC access code if used) and the local exchange code will probably change to an appropriate private network location code.

Network crossover is accomplished through a RESTART action coded in Procedure 314, Word 1, field 11 (encode 1). When a restart is invoked, field 12 specifies the digit modification index number to be used to locate instructions for digit modification. In this case, the digit modification performed is similar to the 10- to 7-digit conversion that was used with the earlier ARS feature; however, WCR digit modification is significantly more powerful. With the WCR feature, this operation is sometimes referred to as M to N conversion where both M and N stand for any number of digits from 0 to 31. See the Digit Modification Instructions section for further details on this operation.

After digit modification, crossover to another network is specified in Procedure 314, Word 1, field 13. A restart can occur in the same network (for example to delete the home area code). Restarting in the same network is not a crossover. Or, you may want to cross over to network 0 to route the call using the internal numbering plan (for example when deleting a home location code in the private network). A crossover to network 0 is a crossover; however, when this occurs, the call exits World Class Routing.

Referrals from Digit Sending

Digit strings are referred to digit modification from digit sending when subnetwork trunking is required. Unlike the network crossover function, the subnetwork trunking capability comes with the basic WCR feature (the standard network option in Procedure 276, Word 1 is not required).

Digit modification is used when the routing preference selected for a call (by the generalized route selection module) requires a different address structure, from that passed to the routing pattern. When this happens, the address string must be modified to meet the requirements of the selected preference For example, the generalized route selection module chooses a routing pattern for a call where the first preference trunk groups are private network trunk groups that use a 7-digit address. The same routing pattern could use public network (toll) routes in later preferences to reach the same end point. These public network routes would require 10-digit address strings. If for some reason, no trunks are available in the private network preferences, the public network routes would be selected and subnetwork trunking would be used to convert the address digits to the 10-digit format required by the public network route.

Digit modification, at this point in call processing, is limited to the address string and ***will not*** affect route selection. Remember that route selection has already been made (by the generalized selection module). When digit modification is required, a digit modification index (for the digit sending module) is specified in Procedure 318, Word 1, field 7.

Digit Modification Index

Digit modification instructions are stored in digit modification tables. The digit modification index is the table entry number used to locate these instructions. Digit modification indexes are numbers from 0 to 4095 and are defined in Procedure 320, Word 1. A digit modification index number is used in two cases:

- ***Network Crossover***

  When network crossover is required, a digit modification index is accessed from the digit analysis module by a restart instruction set in Procedure 314, Word 1, Field 11. The digit modification index to be used is specified in field 12, and the network that the digit string is to be converted for is specified in field 13.

- ***Modification of Digits Sent***

  When the digits sent need to be changed from the address structure passed to the routing pattern, a digit modification index is accessed from the digit sending module by an instruction set in Procedure 318, Word 1, field 7.

Digit Modification Instructions

The digit modification instructions accessed by a digit modification index number are in the form of a specific number of digits to delete (from 0 to 31) and a specific list of digits to insert. The list of digits to insert is also in the range of from 0 to 31 digits. A digit modification instruction tells the software to modify the digit string as follows:

- Either delete NO DIGITS (Digits to delete = 0; this is the default)

  or

- Delete a specific number of digits (from 1 to 31) from the front of the digit string being modified.

  then

- Insert NO DIGITS (the list of digits to insert is empty; this is the default)

  or

- Insert a specific set of digits (from 1 to 31 digits) in place of those that were (or were not) deleted in the first part of the instructions.

As you can see from the above options, digit modification instructions allow you to delete digits, or insert digits, or both delete and insert digits. In all three cases, you are operating on the front of the digit string. That is, if you delete three digits it will always be the first three digits of the digit string. If you insert three digits, they are inserted in front of any digits that remain in the digit string after the deletion operation.

Also, you are limited to a maximum of 31 digits in each operation. That is, you can delete up to 31 digits and you can insert up to 31 digits. However, the number you end up with cannot exceed 68 digits. For example, if you start out with a 10 digit number and don't delete any digits, you can insert up to 31 digits for a total of 41 digits. Note that the 41 digit total here, is the result of the maximum digits to insert limit (31) rather than the maximum total digits (68).

GRS (Generalized Route Selection)

The purpose of the generalized route selection module is to find an accessible route for a call. Generalized route selection makes every effort to accomplish this purpose. The routing options followed (in order) are:

- Route the call over an accessible, available trunk group. This may be done (or attempted) before all digits have been collected.
- Collect sufficient digits to determine if it is a toll-free call.
- Request an authorization code to increase the FRL and gain access to a route if necessary.
- Queue the call on an accessible trunk group if no trunks are available.
- Route the call to the attendant if the ACTGA (Attendant Control of Trunk Group Access) feature is active on an accessible trunk group.
- Provide intercept treatment or reorder tone to the caller.

The generalized route selection module receives the call from network digit analysis with the VNI selected. GRS then performs three basic operations: call category definition, routing pattern selection, and preference selection. The generalized route selection operations are depicted graphically in Figure 134-4.

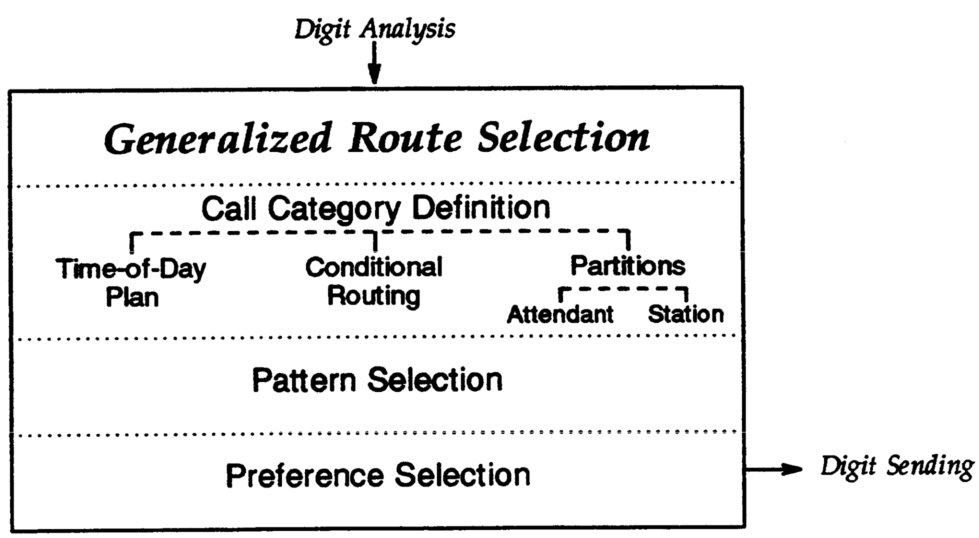


Figure 134-4. Generalized Route Selection

The GRS operations occur in the order shown in Figure 134-4. Each operation provides an input to the succeeding operation.

Call Category Definition

A call category allows the call category parameters (time-of-day plan, conditional routing count, and switch partitioning) to be used for call routing. If call categories are not used, the VIN is used as the pattern number for routing the calls.

Up to 256 call categories (numbered from 0 to 255) can be defined using Procedure 317, Word 1. There are 21,861 possible combinations of the different elements that make up a call category definition. At first, this looks like a severely limiting factor in call category definition. However, the same call category (number) can be given more than one definition. That is, more than one set of call category elements can map to the same call category number. The call category is used, along with the VNI, to select a routing pattern for the call.

Call categories are defined in terms of three factors: the time-of-day plan, conditional routing count (also known as satellite hop control), and tenant partition (if applicable). Each of these factors is discussed separately in the following paragraphs.

Time-of-Day Plans

There can be from one to seven WCR time-of-day plans. These plans are numbered from 1 to 7. One of these plans (and only one plan) is active at any given time of day on any day of the week.

The time-of-day and day-of-week functions will usually operate automatically as clocked functions translated in Procedure 316. When in the automatic mode, the switch changes the time-of-day plan up to six times a day each day of the week.

However, there is also a manual mode which overrides the automatic clocking. The manual mode allows the attendant or administrator to select a specific time-of-day plan. The manual mode can be selected in one of three ways:

- Explicitly Administered Manual Mode

  Manual mode can be expressly set by switch administration using Procedure 286, Word 1, field 14. This blocks automatic clocking and the time-of-day plan must be specifically selected (either through system administration or from the attendant console) each time it is to be changed.

- Clocked Manual Override

  Clocked manual override is administered to take effect during a specific period in Procedure 287, Word 1. This has the effect of overriding the automatic clocking function but not turning it off. That is, clocked manual override is in effect between the start and end times set in Procedure 287, Word 1. The automatic clocking function is then reinstated when the clocked manual override period ends. This is a one time operation. That is, when clocked manual override is administered, it takes effect the next time the assigned start day and time occurs. It remains in effect until the end day and time occurs, and will not take effect again unless re-administered.

- Dial Access from the Attendant Console

  A Dial Access Code (plan change DAC) can be dialed from the attendant console and a plan number entered. The time-of-day plan that corresponds to the plan number entered takes effect immediately. That plan remains in effect until changed by the attendant again dialing the DAC and another plan number. To return to the automatic mode, the attendant must again dial the plan change DAC followed by the number "0."

Manual override is useful for special holidays and other events that do not occur on a regular day of the week basis or for one-time events such as a company picnic or anniversary.

As a typical example, one plan would be used for normal business hours, another plan for off-hours on normal business days, and a third plan could be used for weekends and holidays when the business is closed, while yet another plan could be set up for weekends or holidays when the business is open.

A starting day and time is assigned to a plan number in Procedure 316, Word 1. This is the start or change time used when automatic clocking is in effect. The same plan can be assigned a start time for each day of the week, and in fact can be given different start times on the same day. There can be up to six transitions (automatic plan changes) on any given day.

Plan numbers are associated with a call category number in Procedure 317, Word 1. The current plan in effect is associated with a call in call processing. This is then used, along with the call's conditional routing count, and when applicable, the callers partition to determine the call category of the call.

Conditional Routing

Conditional routing (also known as satellite hop control) counts the number of sensitive (for example satellite) links a call has used to reach the local switch (hence the name satellite hop control). The conditional routing count value ranges from 0 to 2. This results in the following three possible values:

- Conditional Routing Count 0—Satellite count = 0
- Conditional Routing Count 1—Satellite count = 1
- Conditional Routing Count 2—Satellite count = 2 or more.

A call originating on the local switch is assigned a conditional routing count 0. For incoming (tandem) calls, when two (or more) satellite links have already been used, a conditional routing count of 2 is assigned. When the conditional routing count value reaches 2, the use of additional satellite links may introduce signaling delays and signal degradation. When the conditional routing count value reaches 2, the World Class Routing feature can be programmed to select a pattern that will not use additional satellite links.

The conditional routing count associated with a call is sent between switches as a second TCM (Traveling Class Mark). This second TCM follows the Facilities Restriction Level

TCM when used. A trunk group is marked for conditional routing in Procedure 103, Word 1, field 12. An incoming trunk call that does not have a second TCM is given a count of 1 if the trunk group is marked for conditional routing. If the outgoing trunk selected is marked for conditional routing and if the conditional routing count (second TCM) is to be sent, the World Class Routing feature increments the count prior to sending the second TCM.

The conditional routing count is used, along with the current time-of-day plan and the tenant partition status of the call origination point (station or attendant) to determine the call category of a call. Conditional routing counts are associated with a call category in Procedure 317, Word 1. The Call category is then associated with routing patterns and VNIs in Word 2.

Partitions

On a partitioned switch (a switch where the Tenant Services feature is active), it may be necessary to control access to specific patterns and trunk groups (preferences) based on the partition where a call originates. Trunk groups may be "bought and paid for" by a specific tenant who may not want other tenants using their trunk groups. This is particularly important for private network trunk groups and premium service trunk groups such as FX (Foreign Exchange) and WATS (Wide Area Telecommunications Service) trunks. Therefore, on a partitioned switch, the partition of origin is a significant consideration in selecting routing patterns and preferences for calls.

Attendant Partitions

There can be up to 41 attendant partitions (numbered from 0 to 40) on a partitioned switch. Partition 0 is the universal partition and has general access to all facilities and other partitions. Each of the other attendant partitions represents a subdivision of the switch or a tenant group. Calls originated from an attendant console assigned to a specific attendant partition are assigned to a call category associated with that attendant partition (defined in Procedure 317, Word 1). If an attendant is assisting a station with call completion, the station partition is used when determining the call category. If Tenant Services is not active, partitioning is not used in determining call category.

Station Partitions

There can be up to 1000 station partitions (numbered 0 to 999) on a partitioned switch Extension partition 0 is the universal extension partition and calls to or from an extension assigned to extension partition 0 are treated as though the switch were not partitioned. Other extension partitions are subdivisions of the switch and consist of one or more extensions. Calls originating from an extension assigned to a specific extension partition (other than extension partition 0) are assigned a call category associated with their extension partition (defined in Procedure 317, Word 1). If the tenant services feature is not active, all station originated calls are evaluated as though they had originated from extension partition 0.

Incoming trunk calls are not associated with a particular partition. Incoming trunk calls are treated like calls originated in extension partition 0.

Pattern Selection

Pattern selection is based on the VNI provided by the network digit analysis module and the call category determined by the preceding GRS operation. There can be as many as 1023 routing patterns (numbered from 1 to 1023).

Routing patterns are associated with VNIs and call categories in Procedure 317, Word 2. When the VNI selected for a call (during network digit analysis) and the call category assigned to that call (during the call category definition operation) match a combination of VNI and call category assigned to a routing pattern (in Procedure 317, Word 2), that routing pattern is used to route that call.

Preference Selection

Each routing pattern consists of from 1 to 16 preferences (trunk groups) arranged in order of preference. Preferences (and their trunk groups) are assigned to patterns using Procedure 318, Word 1. Often, the order of preference is a least cost first sequence; however, with World Class Routing this is not necessarily the case. Quality of transmission, may be a consideration, or in a network intended largely for data transmissions, digital trunk groups may be considered preferable to analog trunk groups.

Generalized route selection examines the preferences in the selected routing pattern. This process is similar to the preference selection process used by the earlier AAR and ARS features. Preferences are searched in order, for an idle trunk that meets the needs of the call. In addition to the idle state of a trunk, other factors are considered during the preference selection process.

Partitions

In addition to being used to choose a call category, the originators partition (in a partitioned switch) is checked against those allowed for the trunk group. Trunk groups can be assigned to extension partitions in Procecdure 270, Word 5.

Several tenants can use the same pattern but select only those trunk groups (preferences) that are assigned to them. This capability is a carryover from the earlier ARS implementation where, due to the limited number of patterns, both pattern selection via call category and preference selection, had to be used to service all 999 partitions.

FRL (Facility Restriction Level)

Each call processed by the World Class Routing feature has an FRL (Facilities Restriction Level) assigned. For locally initiated calls the FRL used initially is the ***default FRL*** assigned to the originating extension class of service in Procedure 010, Word 3. A default FRL is assigned to incoming trunk groups in Procedure 103, Word 1.

For incoming trunk calls, if it is not possible to complete a call using the default FRL, the original caller's FRL may be included either as a TCM following the address digits, or in the call setup message for ISDN calls. If the origional caller's FRL is higher than the default FRL for the trunk, the original caller's FRL is used in an attempt to choose a route.

Unauthorized Call Control FRL (Facility Restriction Level)

As one of the final steps in ***network digit analysis,*** the Unauthorized Call Control FRL assigned to each the dialed digit string is examined. The unauthorized call control FRL is administered in the Action Attribute field (field 13) of Procedure 314, Word 1. The unauthorized call control FRL is compared to the FRL of the call. If the FRL of the call is equal to or higher than the unauthorized call control FRL for the digit string, the call is allowed to proceed.

Routing Preference FRLs

Each routing preference within each pattern, is assigned its own FRL in Procedure 318, Word 1, Field 4. The switch compares the FRL assigned to the call with the FRL of each preference (in order). A call can use a preference only if the FRL of the call is equal to or greater than the FRL of the preference. If the call can access a preference, the switch checks that preference for an available trunk. If every trunk is busy, the switch checks the next preference.

The same trunk group can reside in more than one pattern and have a different FRL in each preference. In this way, the preference's FRL can more closely correspond to the needs of the pattern and preference.

Figure 134-5 shows an example of such an arrangement. This example uses a 7-digit Uniform Numbering Plan

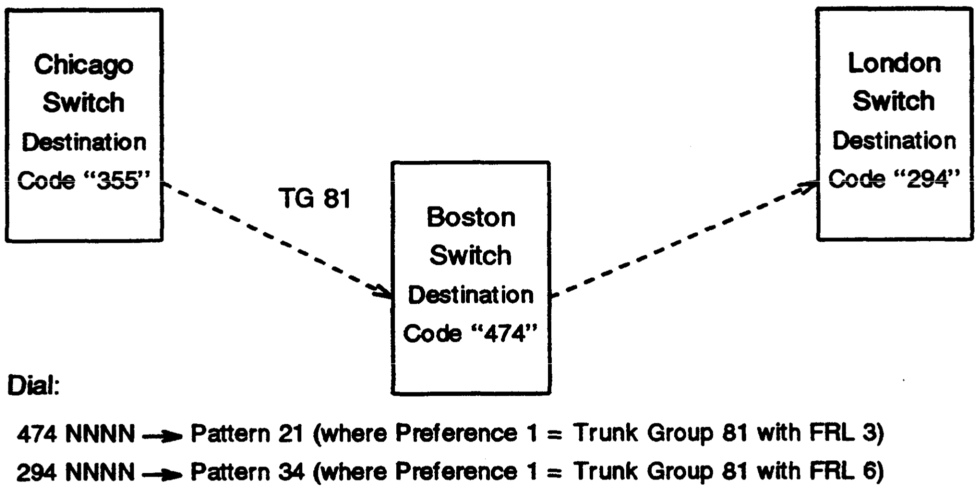


Figure 134-5. Multiple FRLs Assigned by Preference

In this example, the destination code "474" is a tandem switch in Boston, and the destination code "294" is a main switch in London. Trunk group 81 connects the switch in Chicago to the switch in Boston and appears in both Pattern 21 and Pattern 34 on the Chicago switch As a preference in Pattern 21, it routes calls between the Chicago and Boston switches. As a preference in Pattern 34, it carries calls between Chicago and London that are routed by way of Boston. Destination code "474" points to Pattern 21, and destination code "294" points to Pattern 34. Trunk group 81, serving in two patterns, has an FRL of "3" in Pattern 21 and an FRL of "6" in Pattern 34.

The caller in Chicago dialing "294 NNNN" needs a higher FRL to call London than to call Boston even though both calls route over the same trunk group.

Changing the FRL

Authorization Codes

If all accessible preferences are busy, or if the number dialed is denied due to the unauthorized call control FRL, and a higher FRL would allow access to additional preferances, the switch can prompt the caller (with recall dial tone) for an authorization code. The Authorization Code feature (discussed separately) provides a means of overriding the default FRL assigned to a specific voice terminal and replacing it with an FRL assigned on an individual (caller) basis.

If the dialed authorization code has an FRL that is higher than the default (previous) FRL, the switch replaces the default FRL with the authorization code's FRL. Using this new FRL, the switch makes another attempt to find an available trunk If the new FRL is still too low, the switch tries to queue the call on the first accessible preference. The call's FRL (either the default FRL or the authorization code FRL) must allow access to a preference before the call can be queued. The trunk group's queue length (number of calls allowed in queue) must also allow access to queuing. If queued, the switch checks the preference every 2 seconds for an idle trunk. If no trunk becomes idle before the time-in-queue limit is exceeded, the switch makes a "last try" on all accessible preferences in the pattern.

Prompting for Authorization Codes is not done on tandem calls if the incoming trunk group is optioned to receive TCMs.

FRL Raising

Just before the "last try," when the time-in-queue limit elapses for a call, the switch can raise the call's FRL to help provide an allowable trunk facility for the call. FRL Raising is assigned on a per-system basis in Procedure 330, Word 1.

FRL Raising first compares the timed-out call's current FRL with the assigned Threshold FRL (Field 3). If the call's current FRL is greater than or equal to the Threshold FRL, this call is qualified for FRL Raising. At this time, the switch substitutes the assigned Raised FRL (Field 4) for the timed-out call's current FRL if the Raised FRL will be higher than the current FRL.

In addition to possibly raising the FRL on the last try, the call is changed to "toll allowed" to increase the possibility of completion. Note that this does not mean a non-toll caller can place a toll call. A non-tolI route must be a possible pattern choice for the call to have been queued in the first place.

Alternate FRLs

Alternate FRLs is an attendant function of the Facilities Restriction Level feature. By using this function, the attendant can change the default FRL values based on a change scheme administered using Procedure 286, Word 1. The alternate FRL function changes the default FRLs assigned to stations and switch facilities. Alternate FRLs can be higher, lower, or the same as the basic default FRLs. The alternate FRL function is most commonly used to modify FRL assignments during times when a business is closed or on reduced operations.

DCS Preferred

When a call is made to an extension number using the UDP (Uniform Dial Plan) or the ENP (Extension Number Portability) feature, and DCS is active on the switch, a preference containing DCS trunks is chosen if possible. If there are no DCS trunks available but a non-DCS trunk is available, a non-DCS trunk will then be chosen. DCS and non-DCS trunks may be intermixed in the same pattern. If a non-DCS trunk is chosen, the call will complete but feature transparency is lost.

ISDN Required/Preferred

When a station originates a WCR call, the station class of service is checked to see if ISDN is required or preferred. If *SDN is required* only ISDN trunks are selected for call routing. If *ISDN is preferred,* an attempt is made to find an available ISDN trunk to route the call. If no ISDN trunks are available, non-ISDN trunk (if available) are used to route the call. If the extension class of service specifies *use any trunk* (no ISDN preference or requirement), the first accessible trunk (either ISDN or non-ISDN) is used to route the call. Bearer capability is an element of the ISDN (Integrated Services Dial Network) standard supported by the CCITT. On the DEFINITY Generic 2 switch, bearer capability is implemented through the BCCOS (Bearer Capability Class of Service) feature. While BCCOS is based on ISDN standards, it is not limited to ISDN calls or facilities. Bearer capability elements are of primary interest in data calling however, bearer capability applies to voice calls as well as to data calls.

Each calling facility and call support element (extension, line, and trunk) is assigned a BCCOS. This BCCOS identifies the type of call the facility can initiate or support and (for the call type) the requirements (such as Modem Pooling) needed to support the call.

For World Class Routing, bearer capability is an important element in preference selection for network call routing. Identification of the call type and resource requirements is based on the best available information, obtained as follows:

- ***Call Setup Messages***

  For ISDN calls, call control information contained in the ISDN call setup message associated with each specific call is the primary source of information on protocol and call handling facility requirements. Call control setup messages (originating from ISDN facilities) contain IEs (Information Elements) that indicate the type of call (for example, voice, Mode 0 data), protocol used, data rate, and other information needed to identify required resources.

- ***Optional Query***

  Data modules (both ISDN and DCP) have the ability to respond to requests for additional information from the switch. For information that is needed but not available in a specific call setup message this optional query ability is used.

- ***Default Values***

  The last resort for determining resources needed for a specific call is the customer administered ***default BCCOS*** assigned to trunk groups and extensions. This BCCOS provides requirements and characteristics for specific ISDN facilities. The default BCCOSs are associated with the facility (trunk or extension) and not a specific call.

Switch actions based on BCCOS are specified in administration (Procedure 014, Word 1, Fields 4 through 13) for each preference. These switch actions determine how a call with a specific BCCOS will be handled by each preference in each routing pattern. Three specific switch actions are used:

- Circuit switch the call
- Insert a Modem Pooling conversion resource
- Block the call.

With Bearer Capability, the search algorithm operates essentially as follows:

1. ***Preferred Option***

   The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (for example, Mode 2 data, 1200 bps, restricted channel, etc.). If a match is found and a trunk is available, other factors permitting the action taken is to "circuit-switch the call."

2. ***Acceptable Option***

   While looking for a preference that calls for circuit switching the call, the search algorithm also checks for a preference that calls for ***insert a Modem Pooling conversion resource.*** If a preference is encountered that calls for the action ***insert a Modem Pooling conversion resource,*** that preference is recorded for future reference if needed. This option applies only to data calls.

3. ***Exercising the Alternative***

   If the search for the ***preferred option*** is not successful (no usable and available trunk is found), the algorithm tries to connect the call to an ***acceptable option*** trunk if one is also available.

4. ***Unacceptable Option***

   Blocking the call is an unacceptable option. All other alternatives for routing the call (Authorization Code, FRL raising, Queuing, etc.) must first be exhausted. No attempt wilI be made to connect a call to a trunk when the switch action specified is ***block the call.***

A more detailed description is provided under the Bearer Capability feature.

Toll Analysis

Each preference assumes (as a default) that the related trunk group is toll-free. If this is not the case (as with many public network trunk groups), a toll-free table is associated with the preference in Procedure 318, Word 1, field 6. Once assigned a toll-free table, the default assumption about assigned trunk groups changes from toll-free to toll. The toll-free table is then administered in Procedure 319, Word 1 to contain those routing string identifier numbers that are to be treated as toll-free.

There are 63 toll-free tables. While there is a finite number of entries that can be made in these tables, this number of entries is shared between all tables. That is, entries can be distributed as necessary among the toll tables used. All entries could be in one table or distributed between several tables. Them will be sufficient entries available to satisfy toll-free table requirements. A special toll-table entry of 0 means that all calls using the associated trunk group are toll calls.

If users dial the non-toll dial access code for network 1, or if their class of service does not allow toll calls, the preference selections are checked to verify that they will not result in toll charges. If the preference will result in toll charges, that preference is skipped.

Warning Tone

A routing preference may also be administered so that a short burst of tone (warning tone) is given when that preference is selected. This warning tone is traditionally used to alert users to the fact that toll charges are about to be incurred. However, with World Class Routing, the options available in Procedure 318, Word 1, field 5 allow for three possibilities:

- Tone is given on all calls (regardless of toll status)
- Tone is given on no calls (regardless of toll status)
- Tone is given on toll calls only, as determined by the associated toll-free table.

Symmetrical Routing

A symmetrical network is one in which two or more switches can route to each other in a way that, without special provisions, a call could be routed back to a switch that has already routed the call. With alternate routing paths available, this could result in an infinite loop or a call "chasing its tail." One method of protecting against this problem is to apply a symmetrical routing depth limit to affected trunk groups. A trunk group can be marked as belonging to a symmetrical network arrangement in Procedure 103, Word 1, field 4.

When trunks are marked for symmetrical routing, a symmetrical routing depth limit is assigned in field 5 of Procedure 285, Word 1. When a symmetrical routing depth limit is specified, it indicates the preference beyond which an incoming trunk call, marked as symetrical, cannot choose an outgoing trunk group marked as symmetrical for routing. For example, if the symmetrical routing depth limit is set to "3," the trunk group cannot be chosen for a tandem call if the trunk group appears in preferences 4 through 16 in the routing pattern chosen for that particular call. In this case, the preference is skipped. Locally originated calls may always route over a trunk group marked as symmetrical if other criteria (FRL, BCCOS, etc.) are satisfied.

Trunk Reservation Limit

If administered in Procedure 103, Word 1, field 8, a specified number of trunks in the trunk group are reserved for first preference routing. For example, if 12 trunks are reserved for first preference routing and there are fewer than 12 trunks available when a call finds this trunk group using a preference in the range of 2 through 16, the preference cannot be used for that call. This assumes that the trunk group does appear as a first choice preference in at least one routing pattern. If not, the reserved number of trunks could never be used.

FEAC (Forced Entry of Account Codes)

Trunk groups assigned to preference within WCR routing patterns, can be administered to require the use of account codes (see the CDR feature). If a call is originated by a station and an account code is not dialed, any preferences that require account codes are skipped during the route selection process. (See also the Considerations section.)

Attendant Control of Trunk Group Access

If a preference is assigned a trunk group that is under Attendant Control of Trunk Group Access, that preference will be skipped unless there is no alternative route available. After all other methods are attempted to route the call, the call is routed to the attendant for completion.

Digit Sending

The digit sending module receives digit strings from the generalized route selection module with a ***routing pattern and preference*** selected. Digit sending determines if the call can be sent out without further action, or if additional actions are required. Additional actions that may be required include ***digit modification, digit formatting and ISDN messaging,*** and depend on the characteristics of the preference selected.

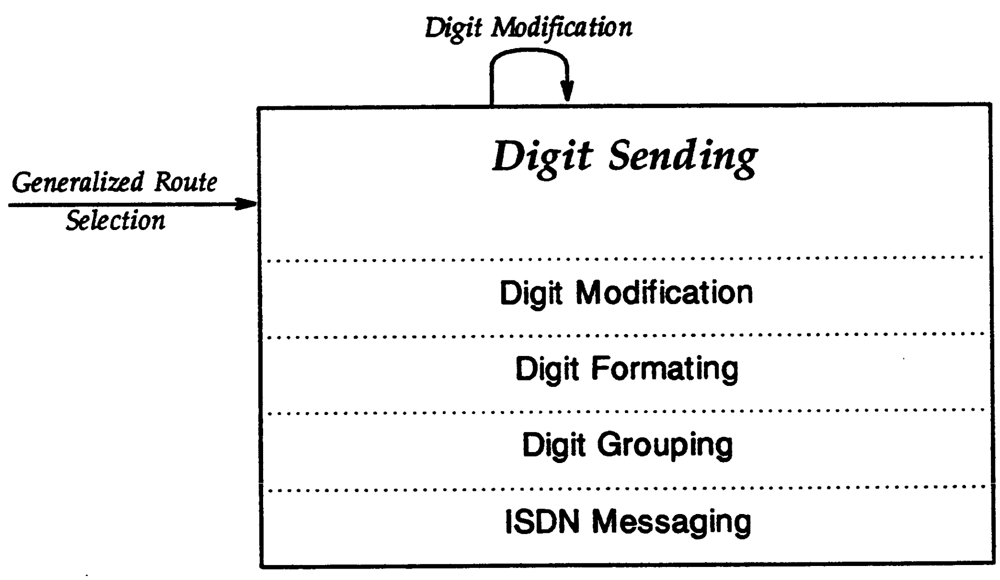


Figure 134-6. Digit Sending

Digit Modification

The digit sending module is one of the sources of input for the digit modification module. Only address strings (with a possible operator assistance or international calling prefix) are subject to digit modification at this point in call processing. As the VNI, routing pattern and preference have already been selected, modifications made here will not effect routing. Rather, the modifications that are made at this point are caused by the routing selection. That is, if a preference has been selected that requires a different address string than the address string originally used to route the call, this is the type of digit modification that is done at this point.

For example, a call may have been routed using a 7-digit private network address string (location code and extension). This address form might be correct for the first several preferences in the routing pattern selected. However, if no accessible and available trunks are found in these preferences it may be necessary to use a lower preference, such as a public network toll route, that requires a different address form. This is the *subnetwork trunking* function and is similar to that used with the earlier AAR feature. The selected preference might require that an area code and office code be substituted for the location code. Similarly, a 7-digit public network number may result in routing to a pattern that contains a Foreign Exchange trunk group to an office in a different area code. In this case, digit modification is used to add the switch's home area code to the number dialed. This is not done automatically as it was in the earlier ARS feature.

If the operator assistance code or international calling prefix is dialed, these generally need to be deleted when an ISDN—PRI preference is chosen. These codes are represented in a different way in ISDN messaging and are not included as part of the called number IE (Information Element).

Digit modification referral from the digit sending module uses a digit modification index number in the same way as the digit analysis module. The digit modification index is defined in the same way (Procedure 320, Word 1) as for digit analysis; however, for the digit sending module the referral is assigned in Procedure 318, Word 1, field 7.

Digit Formatting

Depending on the preference selected for a call, digit formatting may or may not be required before sending the digit string. Digit formatting consists of assembling component elements (for example, IXC code, toll prefix, area code, office code, etc.) of the digit string in the form expected by the distant switch. An end of dialing character (#) can be sent based on digit formatting translations alone (Procedure 321, Word 1, field 16), whether or not it was part of the dialed number.

Digit Grouping

When routing to some older or less sophisticated switching equipment, it may be necessary to break digits up into groups and insert pauses between the groups, before sending the digits to the distant switch. This is to allow the distant switch to operate on the digits as they are received.

If digit grouping is required, it is administered by defining a digit sending index in Procedure 321, Word 2. The digit sending index number is then associated with a specific pattern, preference, and trunk group in Procedure 318, Word 1, field 8. The special digit group size of "99" is used to mean "send all remaining digits." There are 511 digit sending indexes. In most cases, digit grouping will not be required.

Digit Sending Modes

Over Traditional Module Trunks

For trunks on traditional modules, the mode of sending (touch-tone vs. dial pulse) can differ for each digit group (fields 4, 7, 10, and 12 of Procedure 321, Word 2).

Over Universal Module Trunks

For trunks on universal module changing modes during digit sending is not supported. That is, for trunks in universal modules, all digits are sent using the mode specified for the first digit group (field 4).

Default Mode

If a digit sending index is not specified, the default sending mode for the trunk group is used (as administered in Procedure 101, Word 1).

ISDN Messaging

The requirement for ISDN messaging depends on the specific preference selected for the call. ISDN message building and exchange is required when the selected preference is an ISDN—PRI (Primary Rate Interface) trunk group. ISDN—PRI trunks use a form of message oriented signaling where call control and other call related information is passed between switches in predefined message packets.

The ISDN messaging function is the process of constructing ISDN message packets and exchanging them with the distant switch. Much of this message construction is taken care of automatically by the ISDN "interworking function." However, for ISDN trunk groups an ISDN sending index must be defined. ISDN sending indexes are defined in Procedure 322, Word 1 and associated with a pattern, preference, and trunk group in Procedure 318, Word 1, field 9. A maximum of 1023 ISDN sending indexes are available. Translations in Procedure 322 provide instructions to the ISDN message creation software on how to populate the following IEs:

- The Called Number IE with the *Type of Address* and the *Numbering Plan ID*
- The NSF (Network Specific Facilities) IE with the type of service desired
- The NSF, TNS (Transit Network Selection), or Called Number IEs with the carrier identification code for the interexchange carrier to be used
- The TCM IE with FRL and conditional routing count values.

For more detailed information on ISDN messages and ISDN in general see the ISDN—PRI feature and Appendix G.

World Class Routing Process Flow

The following diagrams (Figures 134-7 through 134-21) depict the logic used in WCR call processing. These flow diagrams show the different considerations applied at different steps in WCR call processing. While these flow diagrams do not reflect the internal working details of WCR call processing, they should provide enough information to assist the reader in developing a general understanding of and appreciation for the call processing steps involved.

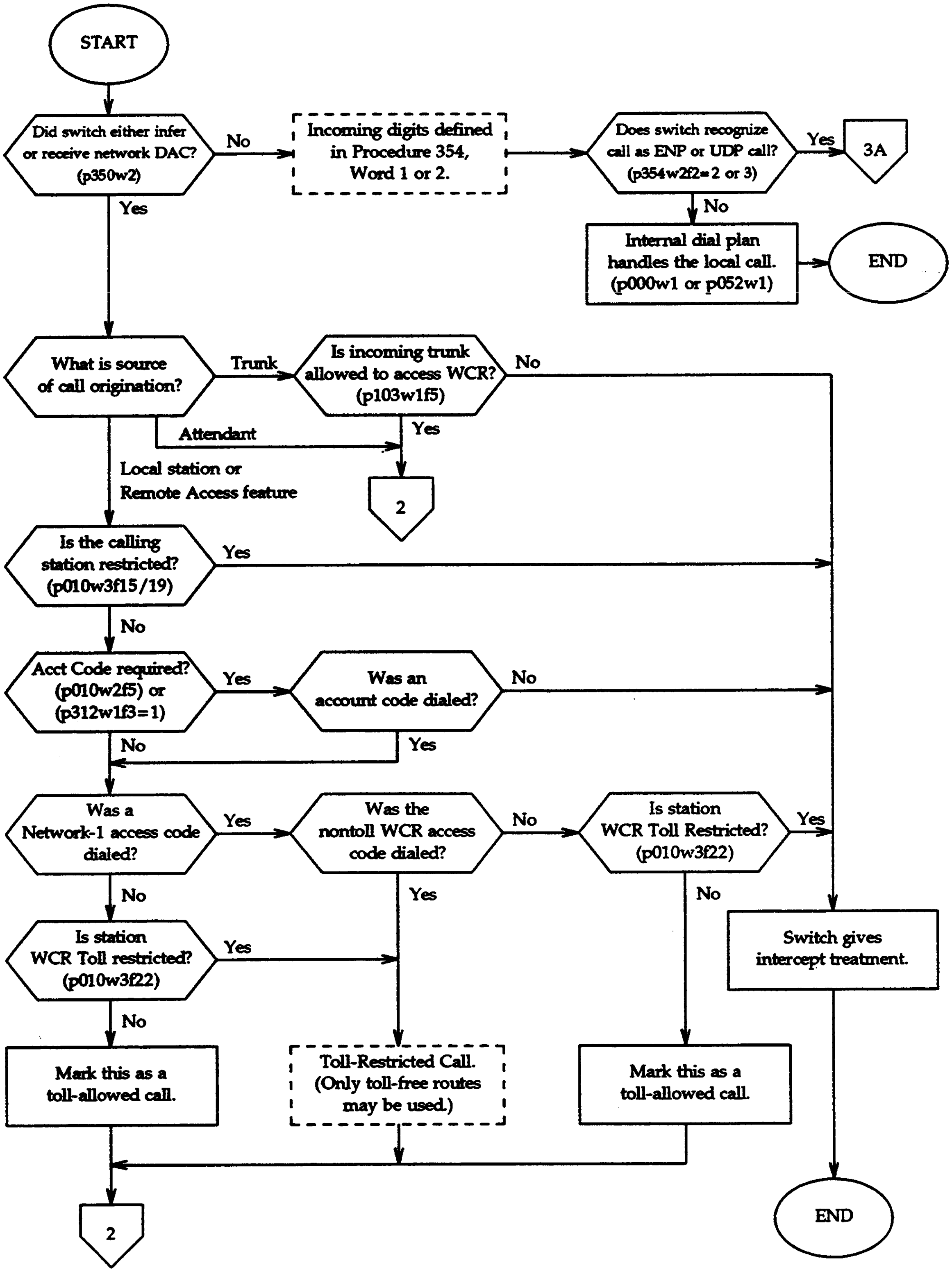


Figure 134-7. Logic Diagrams — Access to World Class Routing

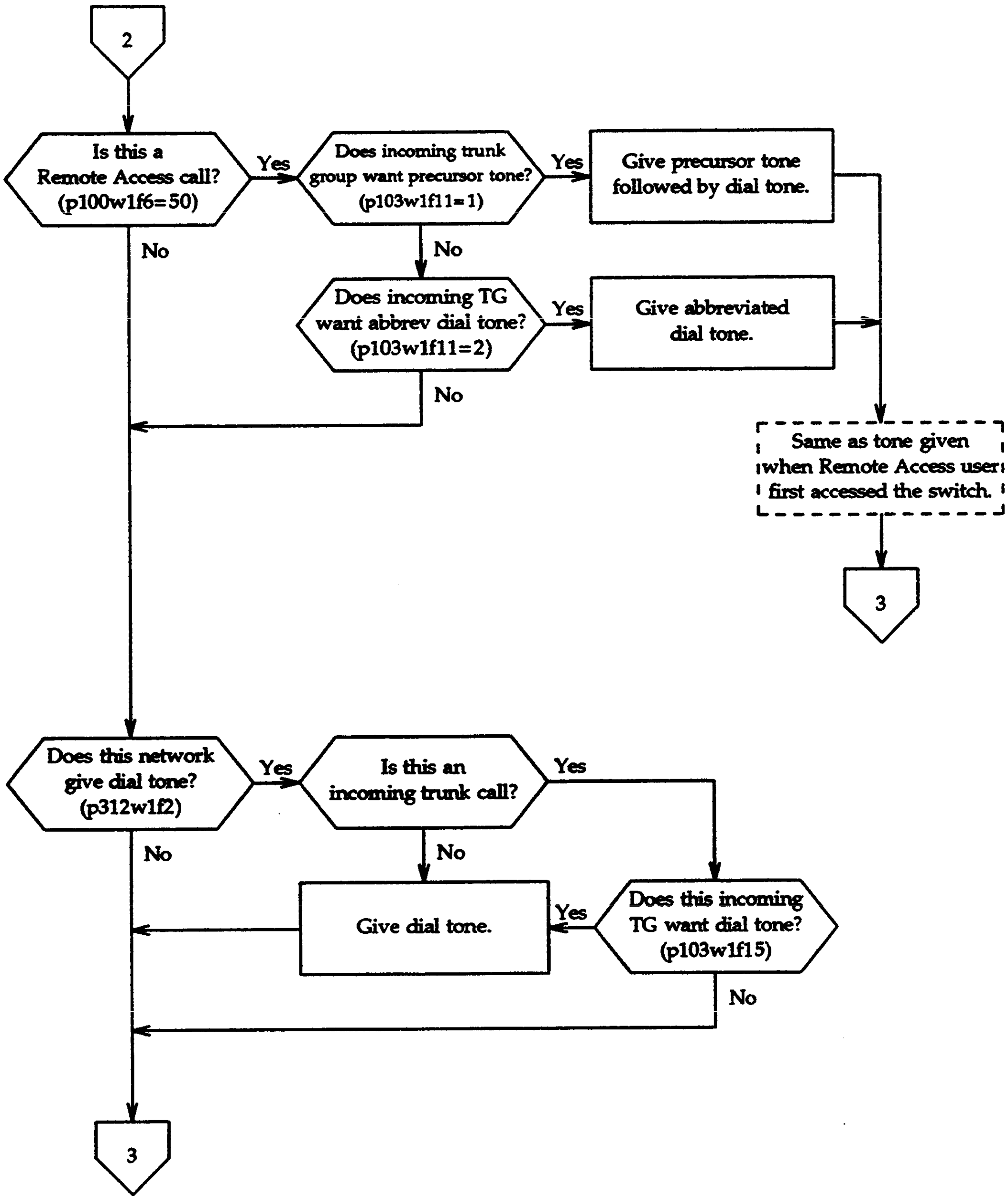


Figure 134-8. Logic Diagrams — Add Network Tones

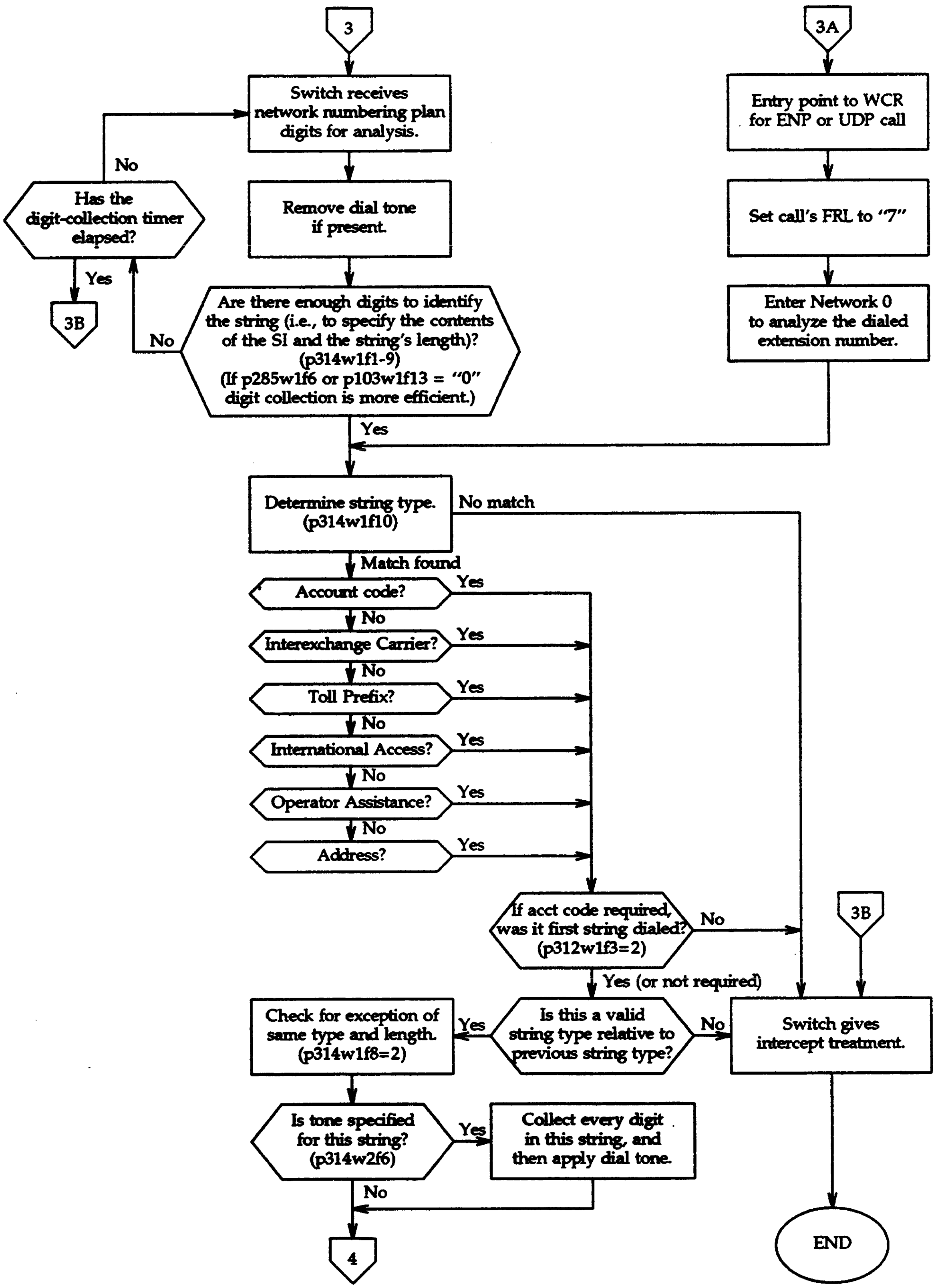


Figure 134-9. Logic Diagrams — Network Digit Analysis-String Identification

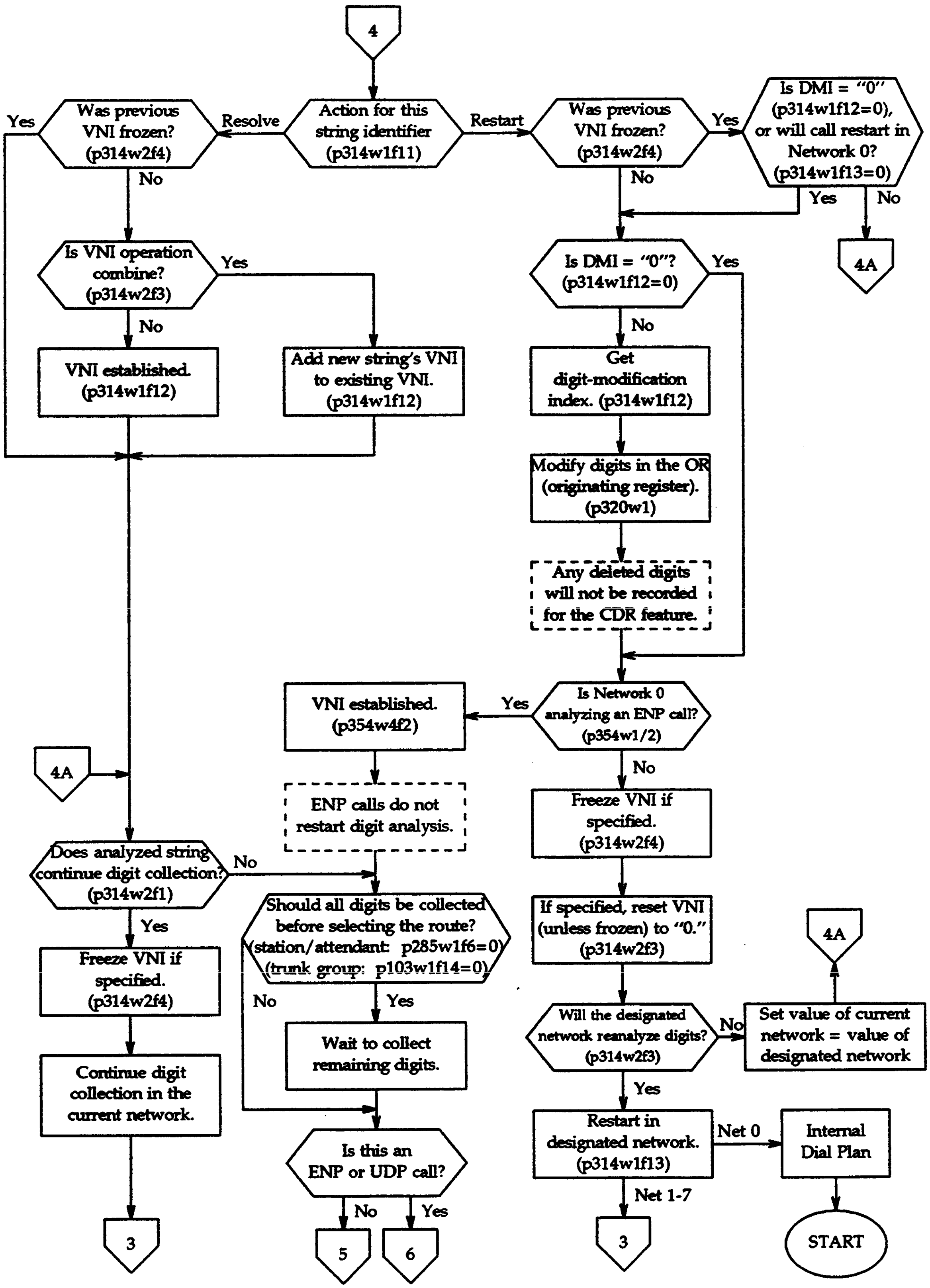


Figure 134-10. Logic Diagrams — Network Digit Analysis-Determining the VNI

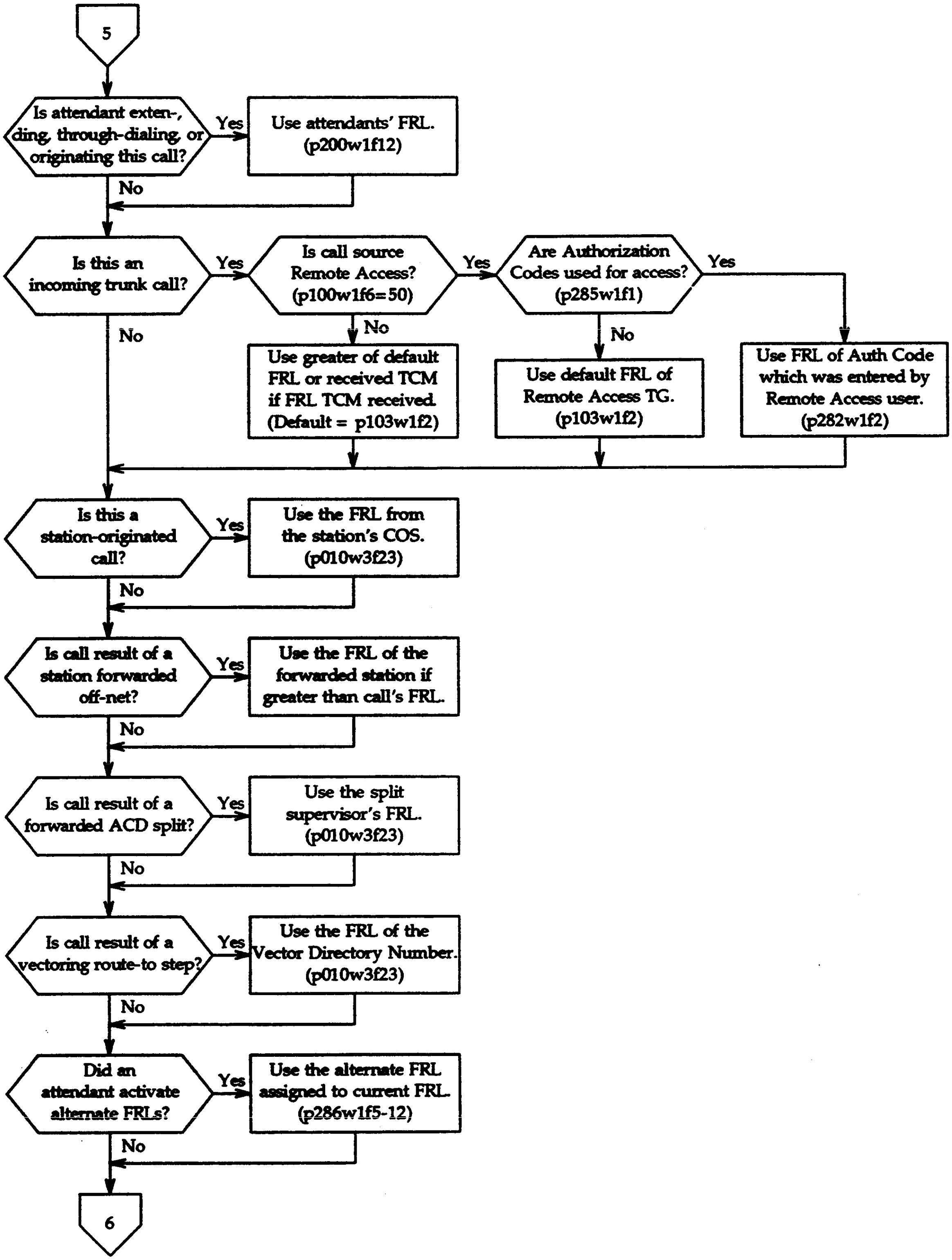


Figure 134-11. Logic Diagrams — Network Digit Analysis-Determine Calls FRL

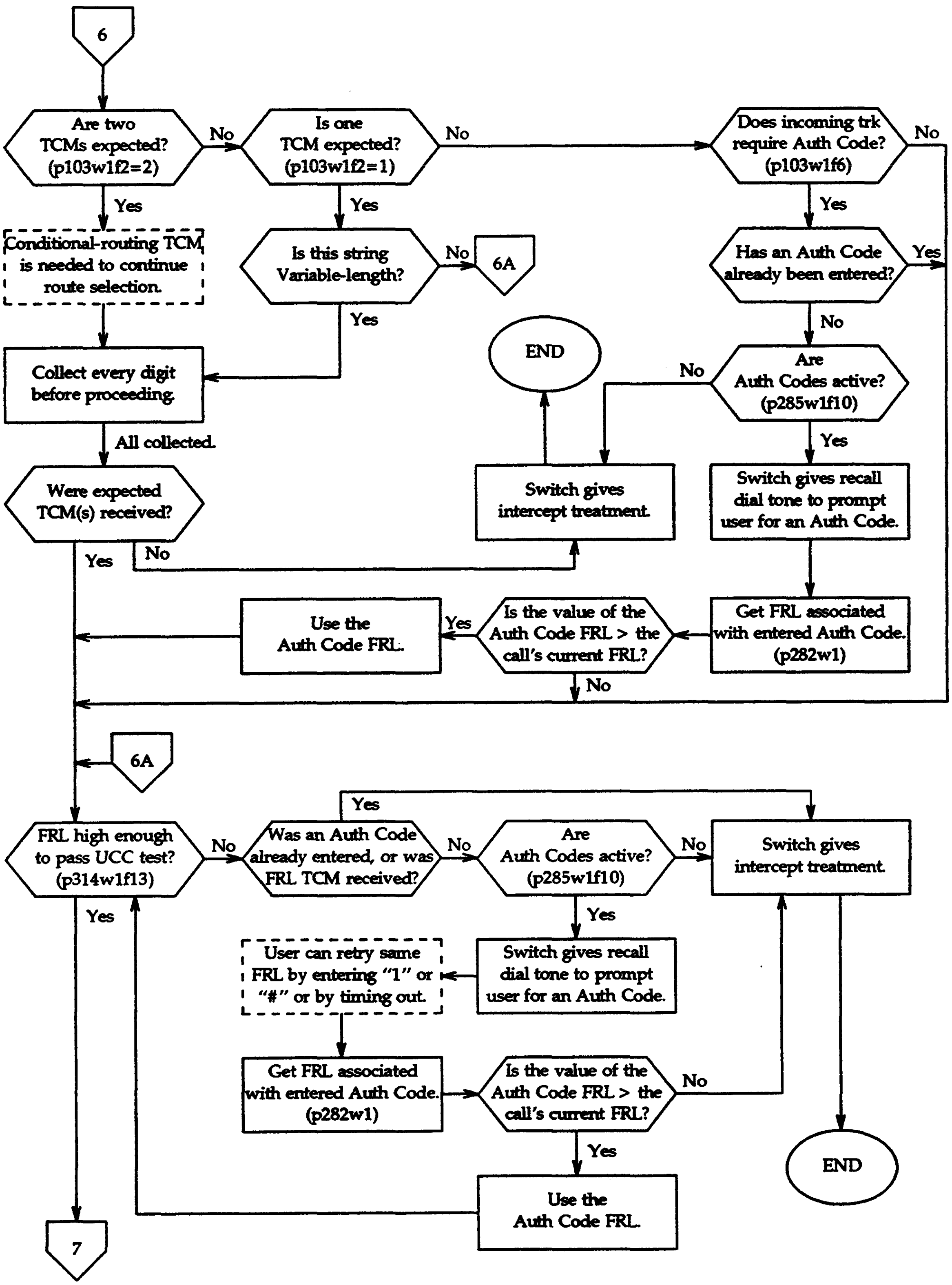


Figure 134-12. Logic Diagrams — Check Permissions

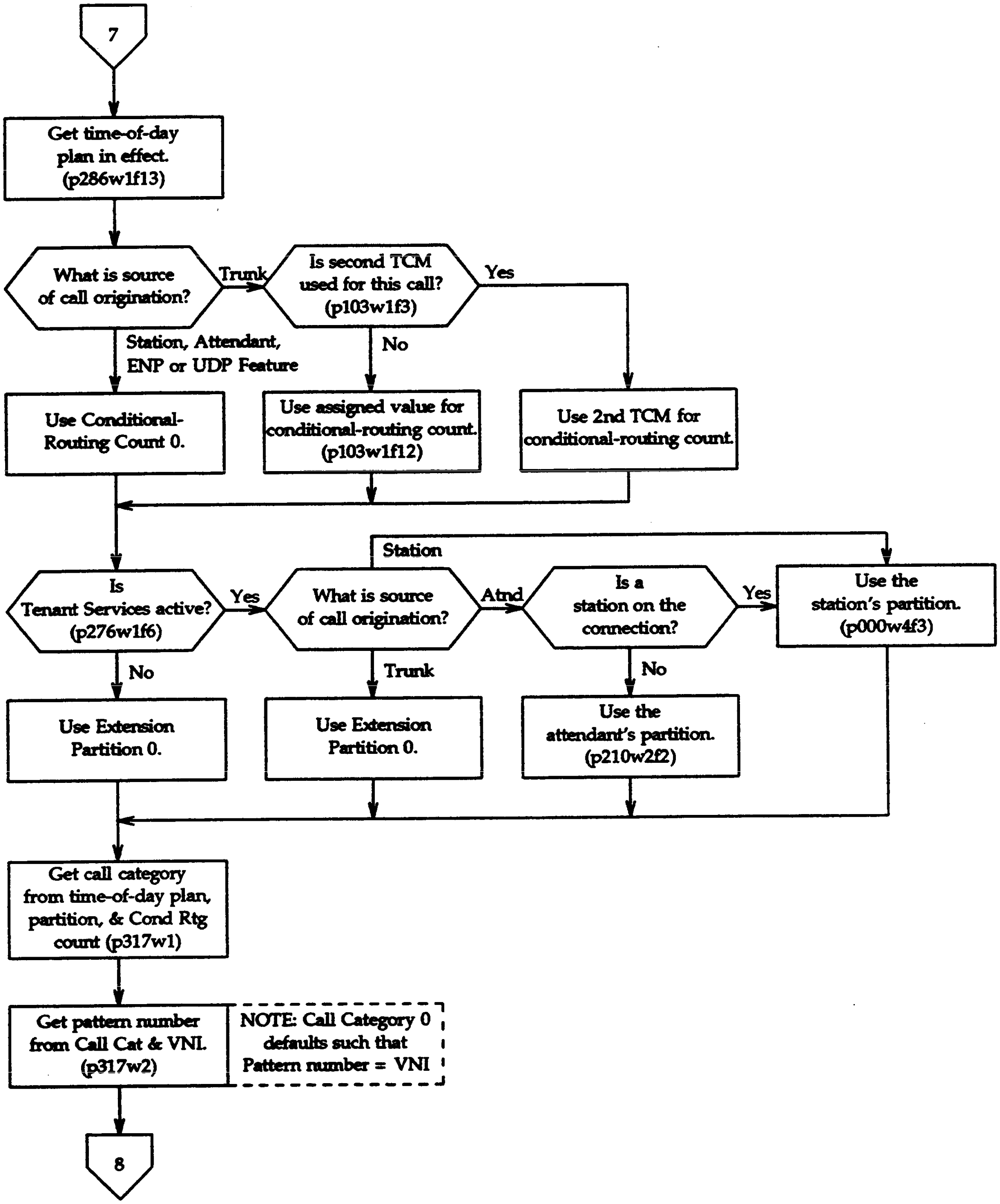


Figure 134-13. Logic Diagrams — Pattern Selection

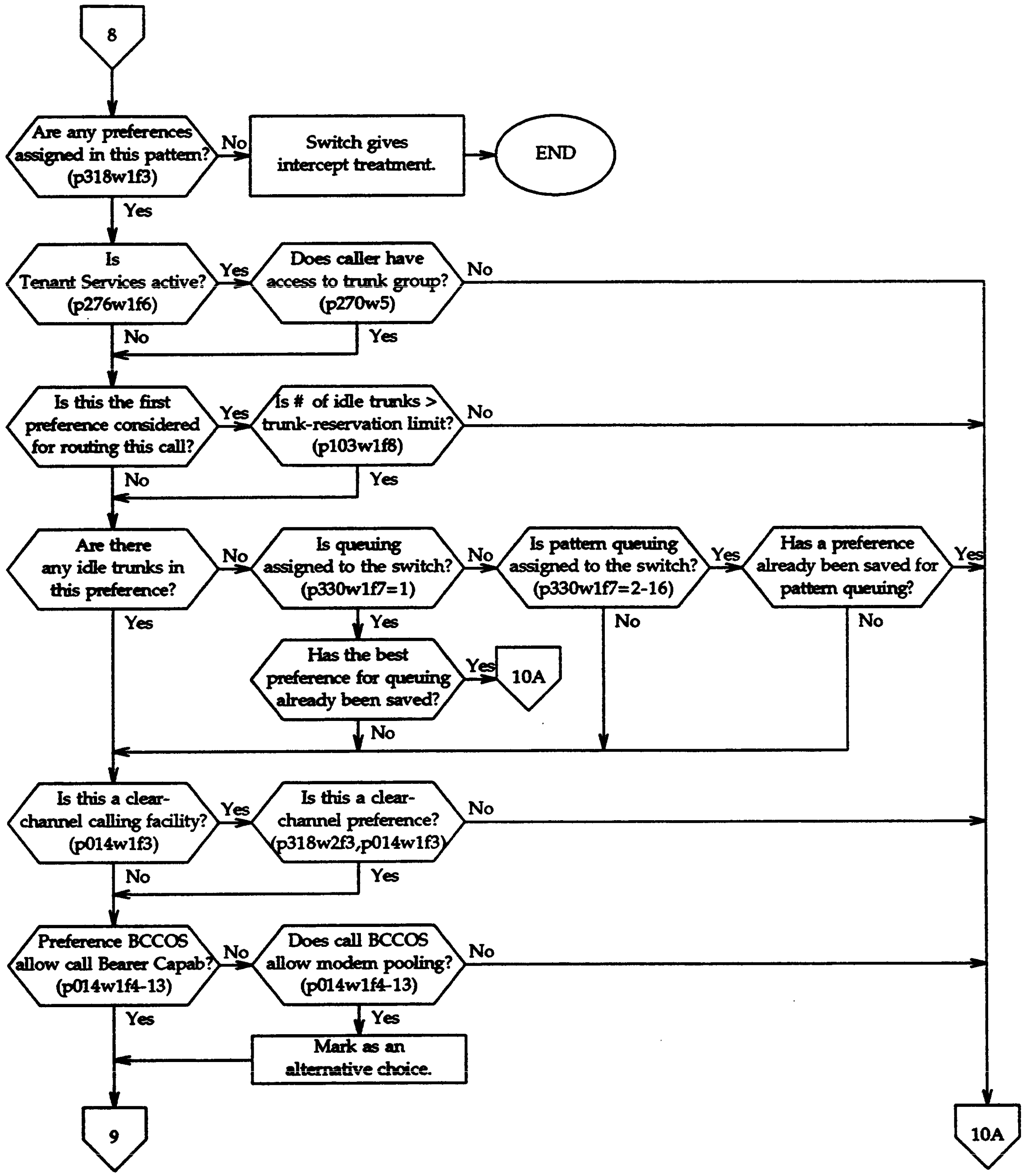


Figure 134-14. Logic Diagrams — Preference Selection (Part 1 of 3)

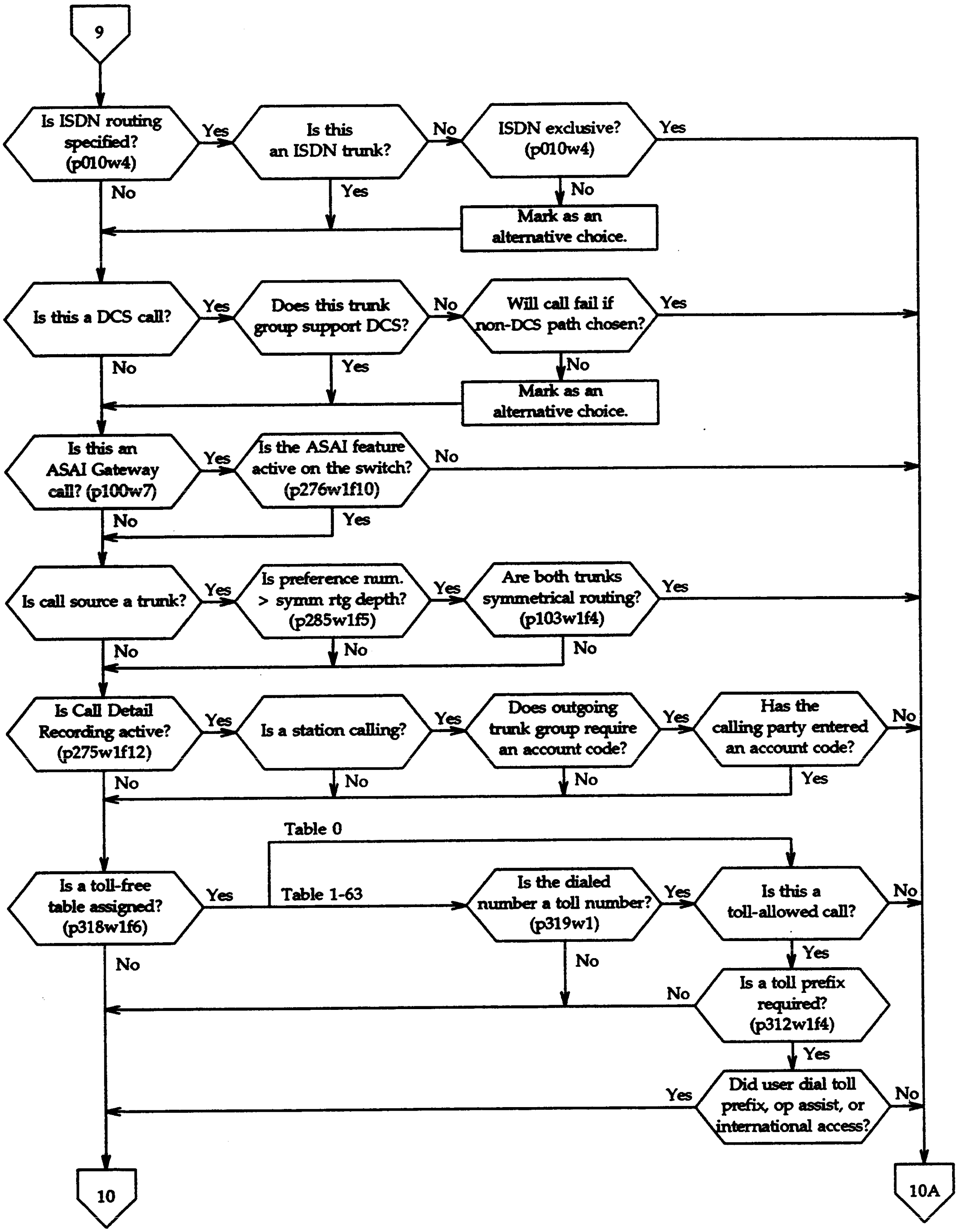


Figure 134-14. Logic Diagrams — Preference Selection (Part 2 of 3)

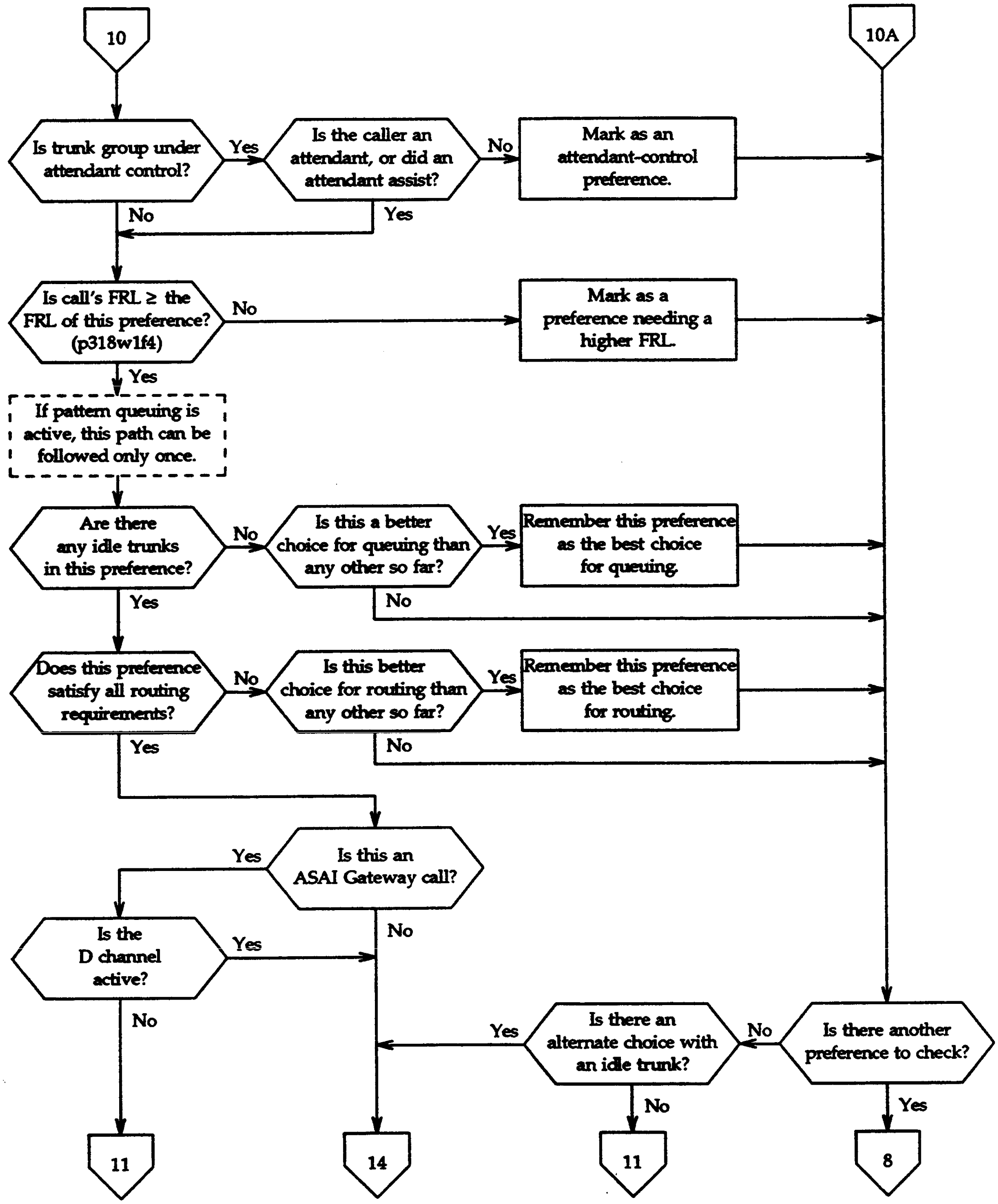


Figure 134-14. Logic Diagrams — Preference Selection (Part 3 of 3)

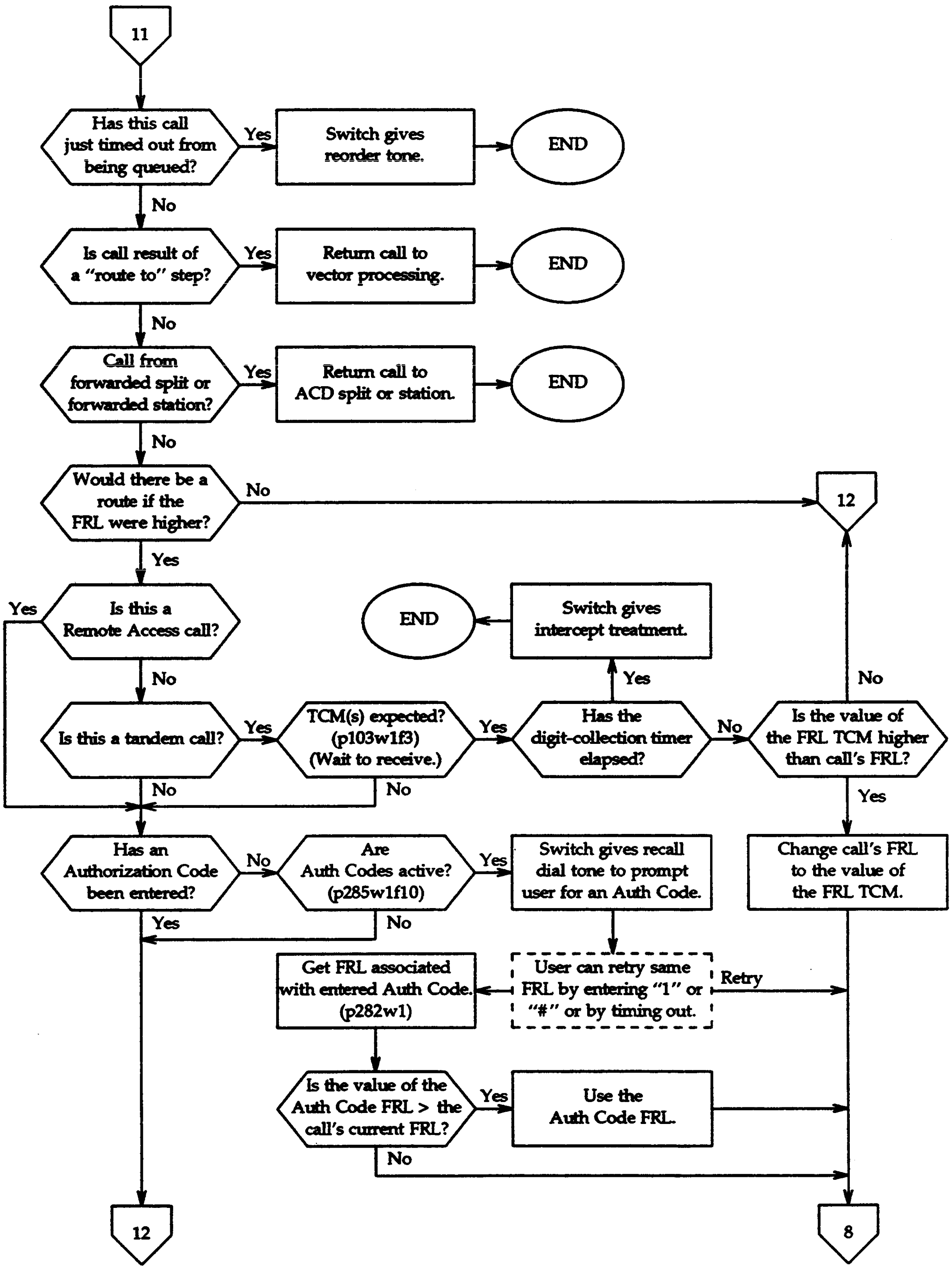


Figure 134-15. Logic Diagrams — No Available Circuit

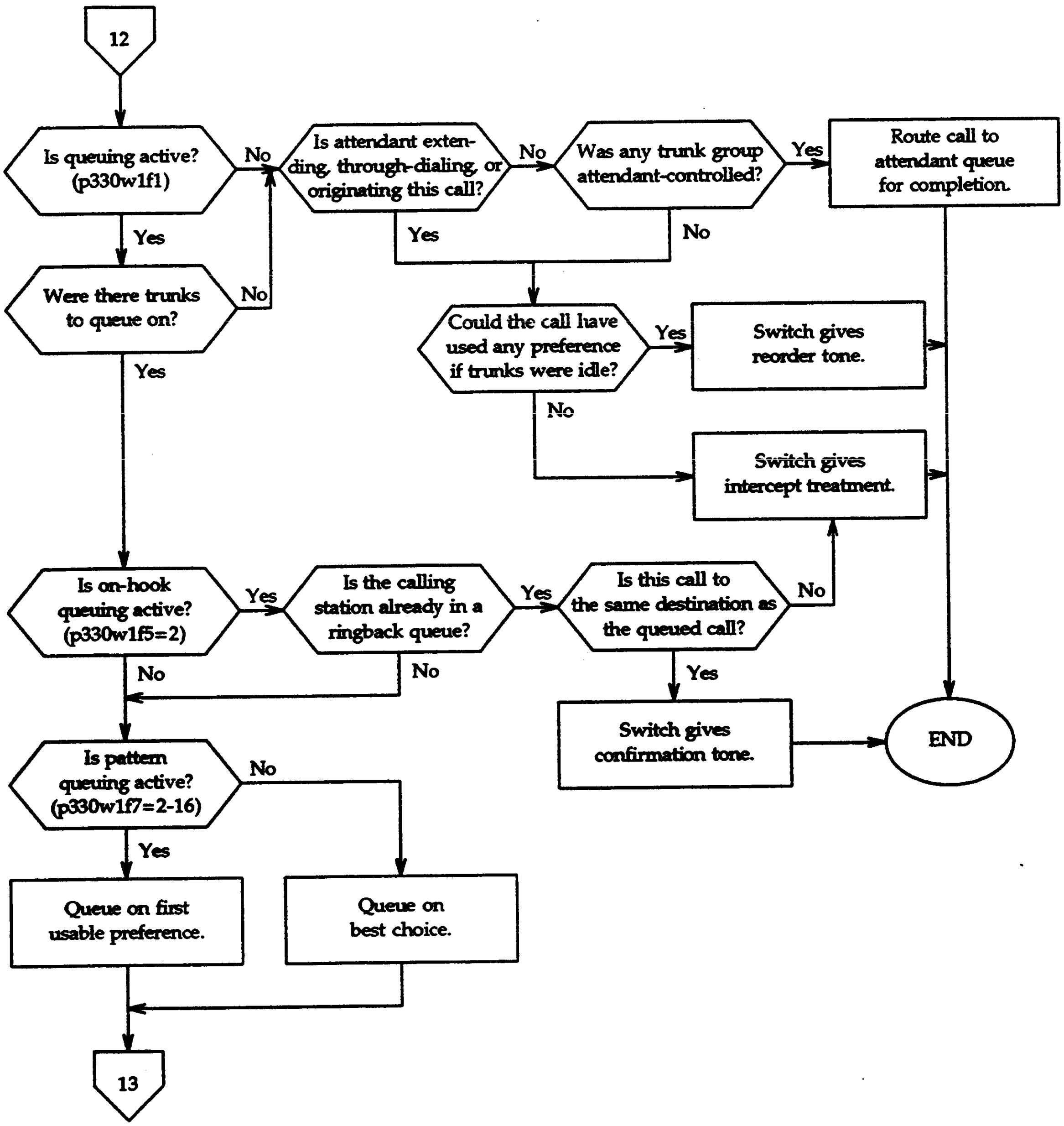


Figure 134-16. Logic Diagrams — Queuing (Part 1 of 2)

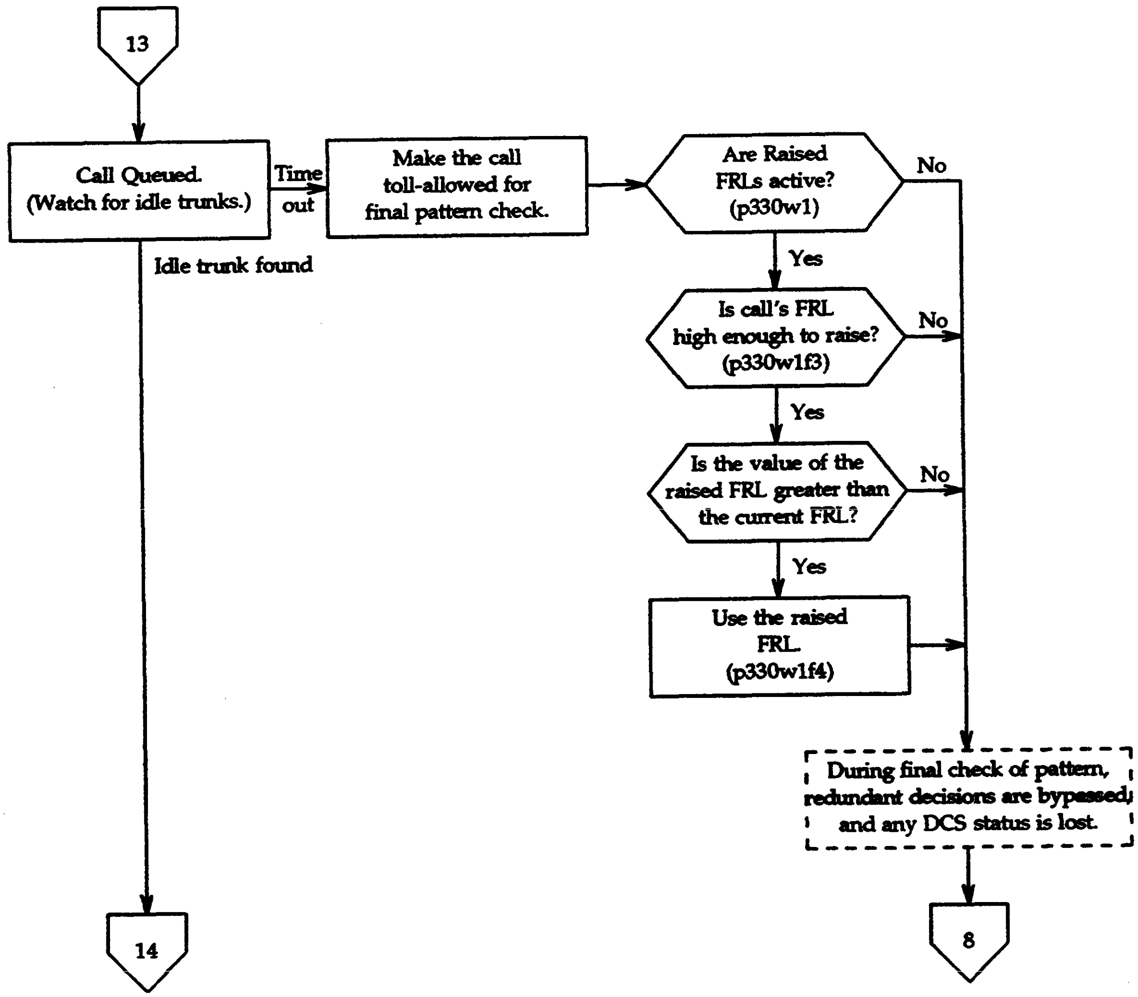


Figure 134-17. Logic Diagrams — Queuing (Part 2 of 2)

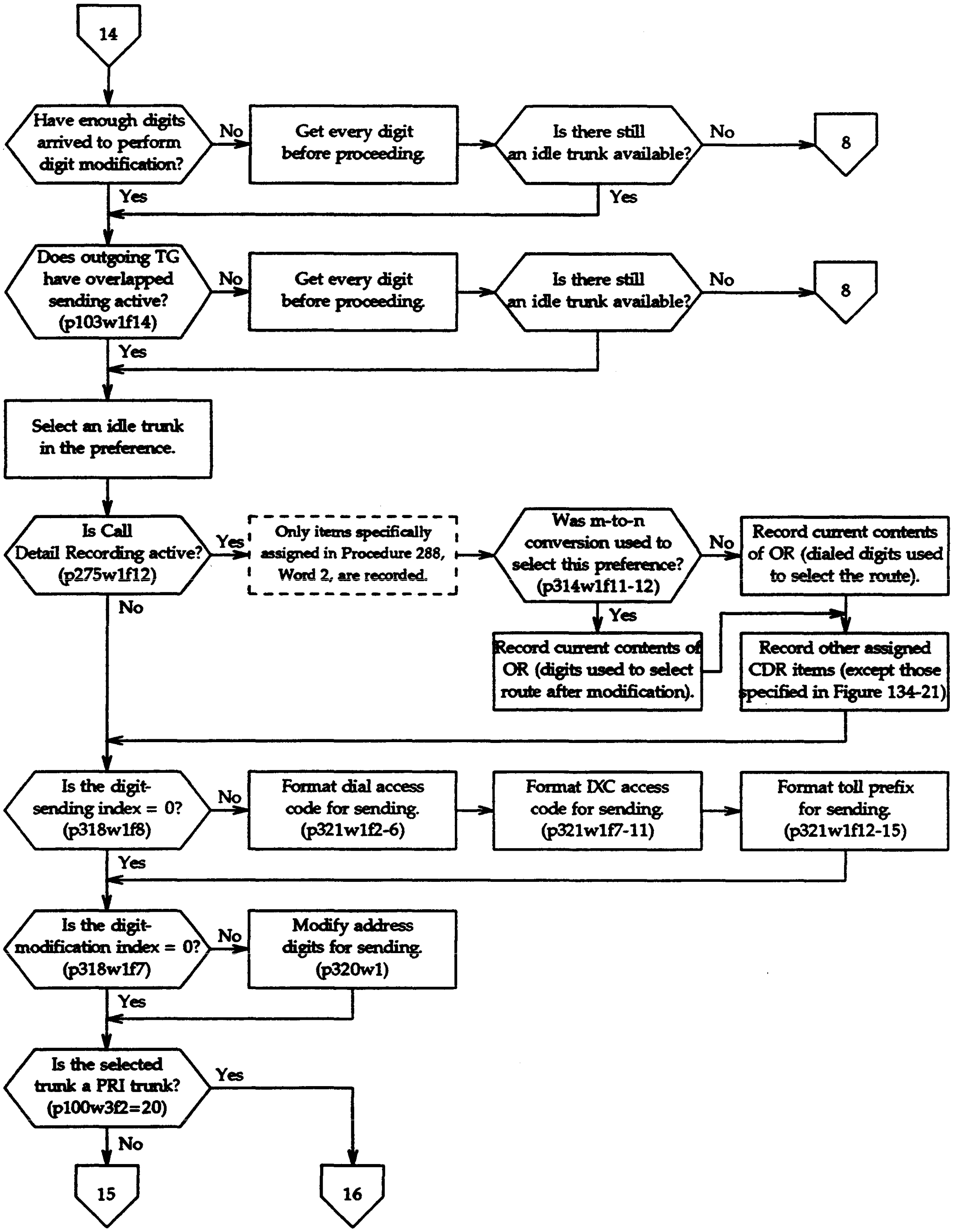


Figure 134-18. Logic Diagrams — Digit Formatting and Modification

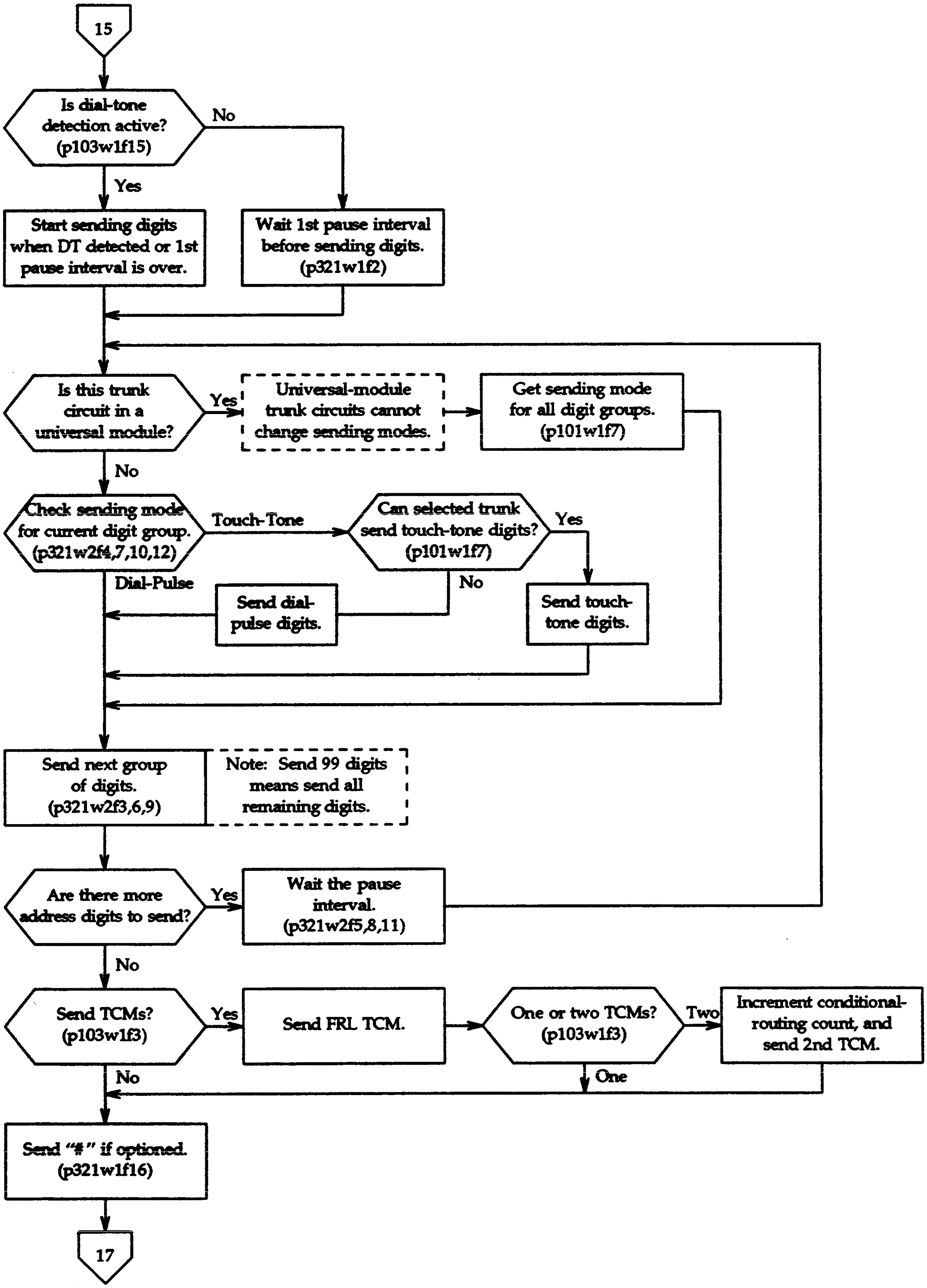


Figure 134-19. Logic Diagrams — Digit Sending — Non ISDN—PRI calls

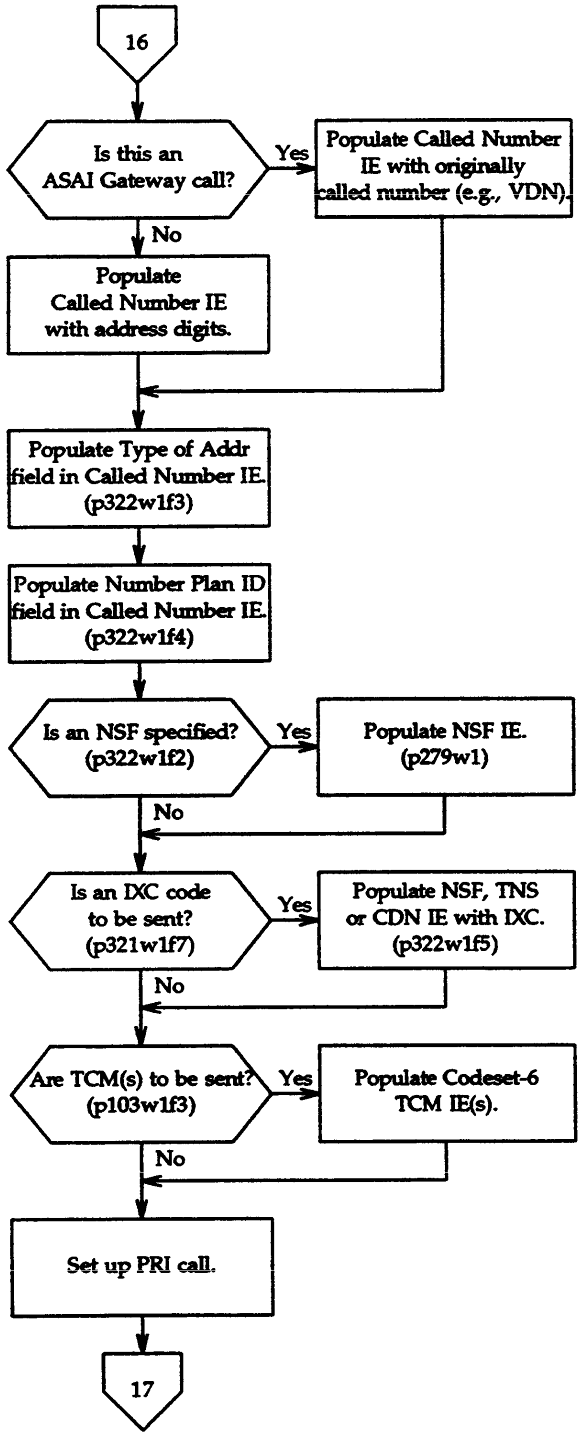


Figure 134-20. Logic Diagrams — Digit Sending — ISDN—PRI Calls

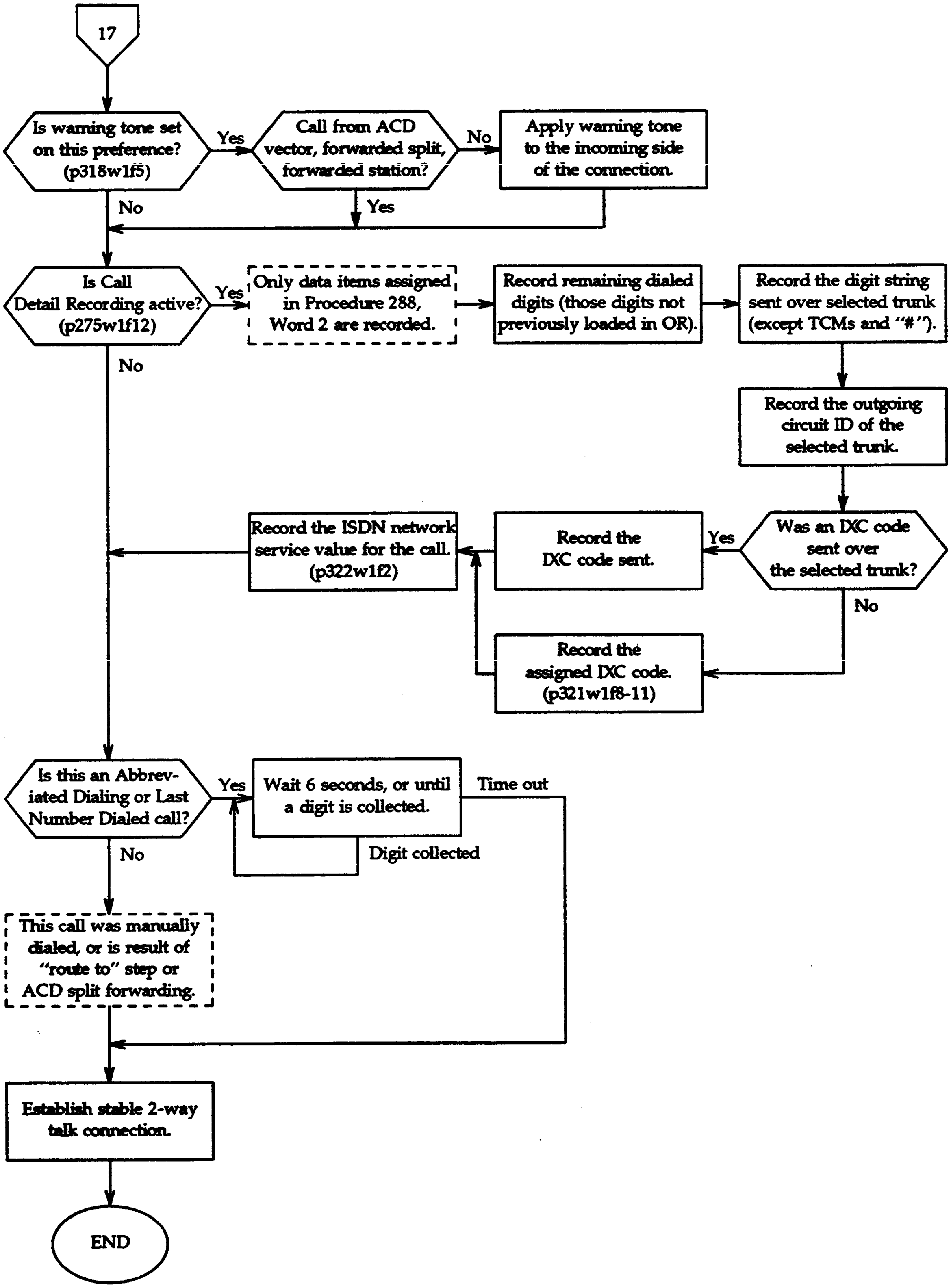


Figure 134-21. Logic Diagrams — Establishing Stable Connection

User Operations

The WCR feature is controlled completely by software. Basic user operations consist of dialing the appropriate feature access code and destination number. Where other features apply (such as the Authorization Code feature) these user operations are given under the applicable feature description. The following operations apply specifically to the World Class Routing feature.

Attendant Operations*

To manually change the time-of-day plan:

From the attendant console:

1. Enter the Plan Change DAC. [Dial tone is heard; the display shows the current time-of-day plan in effect.]
2. Enter the plan number for the time-of-day plan to be started. [Confirmation tone is heard; the display shows the new time-of-day plan.]

To return the system to automatic time-of-day plan changes from the attendant console:

1. Enter the Plan Change DAC. [Dial tone is heard; the display shows the current timed-day plan in effect.]
2. Press **[0]** on the touch-tone dialing pad. [Confirmation tone is heard; the display goes blank.]

   (The switch returns to automatic mode and implements the next automatic time-of-day plan change when that change is scheduled to take place.)

Considerations

Account Codes in Network Administration

An account code prefix is a digit assigned to a network numbering plan (via a network string identifier) that indicates that an account code is being dialed rather than an address. To preserve compatibility with the earlier AAR feature, an account code prefix can be assigned the tone option (set in Procedure 314, Word 2).

FEAC (Forced Entry of Account Codes)

Account codes can be given string identifiers in the digit analysis module. This insures not only that an account code is entered, but also that account code prefixes used are valid for the network being accessed. With WCR account code prefixes are not necessarily limited to the first digit.

For DEFINITY Generic 2.2 switches, FEAC is administered in Procedure 312, Word 1, field 3. This administration provides three options:

* These operations can be performed only from an attendant console.

- Encode 0 = account code not required.
- Encode 1 = account code is required and must be entered using the CDR DAC, prior to dialing the WCR network DAC.
- Encode 2 = account code is required and may be entered either before or after dialing the WCR network DAC.

Of the above options, encodes 0 and 2 can be used with WCR when account code prefixes are given string identifiers. With encode 0, FEAC is not active and has no bearing on account code checking in the network digit analysis module. With encode 2, FEAC is active and an account code must be entered either before or after the network DAC. If entered before the network DAC, the account code is not checked against any string identifiers administered for the network.

AUTOVON Access

The AUTOVON (Automatic Voice Network) is a private network arrangement belonging to the US Government. AUTOVON access, to and from the DEFINITY Generic 2, is implemented and controlled through the Precedence Calling feature *(see* Feature Interactions). Precedence calls are not routed by the WCR feature (Precedence Calling has its own separate routing patterns). However, the Precedence Calling feature must prefix one of the network 1 access codes to reach the AUTOVON routing software. The *standard network* switch software option is no longer required for AUTOVON access.

Digit Manipulation

The digit modification module allows digits to be manipulated in either of two ways, within specific limits, as follows

- Digit Deletion
  a. Digits are deleted from the front of the digit string being manipulated
  b. The number of digits deleted can be 0
  c. Digit deletion is limited to the total number of digits in the string, but not more than 31 digits in a single operation.
- Digit Insertion
  a. Digits to be inserted are always inserted after digit deletion
  b. Digits are always inserted in front of any digits that remain after the digit deletion operation
  c. Digit to insert can be 0 (field 3 of Procedure 320, Word 1 is set to 0)
  d. A maximum of 31 digits can be inserted in a single operation
  e. The maximum size of a digit string (digits remaining after deletion plus digits inserted) is 68.

Overlapped Sending

Overlapped sending is sometimes used on tandemed calls. It is the process of beginning to transmit digits on the outgoing trunk before all digits have been received from the incoming trunk. When networks consisted largely of dial pulse trunks, considerable delay could be experienced if every switch in the network waited until all digits were received before tandeming calls on to the next switch.

As more trunks began using "touch-tone" sending, this delay became less significant, and with ISDN—PRI messaging, it becomes irrelevant.

For the WCR feature, the default is not to use overlapped sending. If it is important to begin sending digits as soon as possible, two conditions must be administered:

- Digit collection optimization must be disabled. This is done for local stations and the attendant by using encode 1 in Procedure 285, Word 1, field 6. For incoming trunk groups, this is done with encode 1 in Procedure 103, Word 1, field 13.
- Digit sending optimization must be disabled. This is done for outgoing trunk groups by using encode 1 in Procedure 103, Word 1, field 14.

When **both** conditions occur, overlapped sending will result.

NOTE: Extensive use of overlapped sending can adversely affect processor occupancy.

Standard Network Option

The basic switch software provides only the two dial access codes required for toll and non-toll calls on network 1. At additional cost, the standad network option provides the following additional World Class Routing functions:

- Dial Access to networks 2 through 7
- Crossover between networks through digit analysis
- Administration of numbering plans in network 0 and networks 2 through 7
- Send/Receive TCMs on designated trunk groups.

Trunk Group Prefixing

Trunk group prefixing is the function of inserting from one to four prefix digits at the beginning of an incoming number on a specific trunk group. Trunk group prefixing is assigned in Procedure 101, Word 3. The administered prefix is added at the beginning of the incoming digit strings on the assigned trunk group. The prefix corresponds to the appropriate network dial access code that would have been used if the call had originated locally. Prefix digits are sometimes referred to as ***inferred*** digits.

Prefixes can be assigned to incoming calls on the following trunk types:

CCSA/APLT Trunks	Trunk Types 12 through 15
DID Trunks	Trunk Types 30 and 31
Tie Trunks	Trunk Types 32 through 47
Main/Satellite Trunks	Trunk Types 70 through 78
ISDN Dynamic	Trunk Type 120

Non-ISDN trunk groups can be assigned one prefix of from 1 to 4 digits. For ISDN trunk groups (any trunk groups using signaling type 20), up to 8 different prefixes can be assigned, based on the ***type of address*** (Procedure 101, Word 3, field 2) indicated in the Called Number IE of the ISDN *Call Setup Message.* Using the "type of address," up to eight different network numbering plans can share the same trunking facilities without numbering plan conflicts. Note that ISDN trunk groups are not limited to trunk type 120.

(See the ***ISDN—PRI*** feature and ***Appendix G*** for information on ISDN message oriented signaling, and Table F-B, ***DEFINITY Generic 2, Trunk Types and Signaling Characteristics*** in Appendix F for the trunk types that can use signaling type 20.)

Trunk Types Assignable in Preferences

The following trunk types can be assigned to WCR routing preferences:

CCSA/APLT Trunks	Trunk Types 12 through 15
Regular CO Trunks	Trunk Types 16 thtough 20
FX Trunks	Trunk Types 21 through 25
WATS Trunks	Trunk Types 26 through 29
DID Trunks	Trunk Types 30 and 31
Tie Trunks	Trunk Types 32 through 47
Main/Satellite Trunks	Trunk Types 70 through 78
ISDN Dynamic	Trunk Type 120

Unauthorized Call Control

Unauthorized call control can be provided in either (or both) of two ways:

Unauthorized Call Control FRL

The unauthorized call control FRL is assigned to a string identifier and applied to a called number during digit analysis. This is a limited form of unauthorized call control in that it limits access to a called number rather than absolutely blocking access to the number (string identifier). The unauthorized call control FRL works in the same way as the FRL assigned to a routing preference. A call's FRL (either the default FRL or one obtained through use of an authorization code) must be equal to or greater than the unauthorized call control FRL to have access to the number. If a call's FRL is 7 (the highest possible FRL), that call has access to all numbers.

String Identifier Administration

Another way of providing unauthorized call control in World Class Routing is through the administration of a string identifier. Digit strings that digit analysis cannot identify (have not been administered) or are specifiy administered as not allowed (cannot route) provide a positive and specific form of unauthorized call control. For example, dialing 900 numbers can be blocked by administering the 900 digit string identifier in Procedure 314, Word 1 as an address string that resolves to an ***empty or unroutable VNI.*** This can be done by resolving the string to VNI 0 (a non-routing VNI). Alternatively, you can resolve the VNI to a null routing pattern (in Procedure 317, Word 2 assign the VNI to a routing pattern with all dashes in field 3).

UDP (Uniform Dial Plan)

The uniform dialing plan (previously called RNX routing), allows ranges of extension numbers to be converted into private network dial plan numbers for routing to the switch on which those extensions reside. With UDP, users may dial other users on other switches by using their extension numbers and do not need to know on which switch the dialed extension resides. This is particularly important in a DCS (Distributed

Communications System) environment and where a large organization is served by several switches.

The extension number range is indicated as assigned to UDP in Procedure 354, Word 2. The call is sent directly to WCR network 0 with an FRL of 7, for further analysis and routing. In network 0, the extension number ranges are entered as string identifiers which restart in the private network after the digit modification specified has been performed.

On the terminating switch, the home location code is recognized as a string identifier in the WCR private network digit analysis and this string is specified as restarting in network 0 after appropriate digit modification to restore the extension number. Restarting in network 0 does not use network 0 translations for analysis. It is a special indicator that sends the call to the switch internal dial plan for analysis.

Access to both network 0 translations and the private network requires that the Standard Network option be selected in Procedure 276, Word 1, field 1.

Universal Trunk Sequence

When the trunk seized uses the universal trunk sequence, the switch will react differently depending on the far end response. If a ready signal (wink) is not received from the far end switch, digits are outpulsed as soon as they are ready. If a ready signal is received from the far end switch before digit outpulsing begins, the local switch will wait for 5-seconds before outpulsing digits.

Interactions with Other Features

The following System 85 and Generic 2 features affect or are affected by the operation of this feature.

Abbreviated Dialing

The Abbreviated Dialing feature works with World Class Routing in the same way it did with the previous networking features AAR and ARS. The network dialing digit sequence can be stored in an abbreviated dialing storage location and then used through the appropriate abbreviated dialing operating procedure to place any World Class Routing network call.

ASAl (Adjunct/Switch Application Interface)

The ASAI feature works with World Class Routing as it did with the earlier networking features, AAR and ARS. (In Generic 2.1, the ASAI feature was called ITGI [Integrated Telemarketing Gateway Interface].) Incoming telemarketing or call-center calls are generally routed to a call vector using its Vector Directory Number (VDN). VND route-to steps then generally redirect the call to a private network number dedicated to the ASAI interface. Since World Class Routing can analyze public network numbers to the units digit, the VND route-to step can send the call to a specific public network number administered in Network 1, and thus to the desired ASAI routing pattern. In this way, the Standard Network option (Procedure 276, Word 1) is not required.

APLT (Advanced Private Line Termination)

The APLT feature works with World Class Routing as it did with the earlier networking features, AAR and ARS. Incoming APLT calls can route to the WCR feature for subsequent routing. Outgoing calls can use APLT trunks as a routing preference. If a dial pulse station is assisted by the attendant in placing a call that routes to an APLT trunk, and if the APLT network requires that an authorization code be entered, the attendant must enter the authorization code for the caller. The (dial pulse) station user cannot enter authorization code digits after the attendant releases from the call.

Attendant Control of Trunk Group Access

The Attendant Control of Trunk Group Access feature interaction with World Class Routing is similar to the interaction with the earlier networking features, AAR and ARS. With WCR, placing a trunk group under attendant access control, essentially removes that trunk group from availability to the World Class Routing feature. If the trunk group is one of several in an available routing pattern this simply limits the number of potential choices available to World Class Routing. If the controlled trunk group is the only trunk group in the routing pattern, or if all other accessible trunk groups are unavailable, the call will be routed to the attendant. This routing to the attendant takes place only after all other attempts to route the call have been unsuccessful.

Authorization Code

The Authorization Code feature works with the WCR feature in essentially the same way as it does with the previous networking features (AAR and ARS). If the default FRL of a call is not high enough to allow call processing to access an available trunk, the switch can prompt the caller to enter an authorization code. If the FRL assigned to the authorization code is higher than the default FRL originally assigned to the call, the new FRL is used in another attempt to connect the call.

If the caller does not have an authorization code or for some reason doesn't want to use it, dialing a "1" or a "#" will cause the switch to skip that step and avoid the 10-second time out that would otherwise occur if no authorization code is dialed. Even if an authorization code is not entered, the switch will make one more attempt to route the call.

NOTE: Authorization codes cannot begin with the digit "1" or the "#" character.

AAR (Automatic Alternate Routing)

The AAR feature is replaced by the World Class Routing feature in DEFINITY Generic 2.2. World Class Routing administration options can provide all the functionality that was formerly available through the AAR feature.

ACD (Automatic Call Distribution)

While ACD itself is not directly involved with World Class Routing the ***interflow*** function of the ACD feature uses WCR to route calls to distant switches. When this happens, ACD interflow is fully compatible with the World Class Routing feature. The FRL used for an interflow call is that of the split supervisor. These calls will not queue, will not be prompted for an authorization code, will not be given warning tone, and are toll allowed.

Automatic Callback

Automatic Callback works with the World Class Routing feature in a DCS environment. Once Automatic Callback is in effect for a call between DCS nodes, the WCR feature verifies that an accessible trunk is available before attempting to recall the originator.

ARS (Automatic Route Selection)

The ARS feature is replaced by the World Class Routing feature in DEFINITY Generic 2.2. World Class Routing administration options can provide all the functionality that was formerly available through the ARS feature.

Bearer Capability

Bearer Capability is a significant factor in the generalized route selection module processing for the WCR feature. Preference selection is based partially on the BCCOS (Bearer Capability Class of Service). Two methods are available for routing the call:

1. The search algorithm first looks for a preference that matches the call requirements in the call setup message or BCCOS (for example, Mode 2 data, 1200 bps, restricted channel). If a match is found and a trunk is available (other factors such as FRL permitting), the action taken is to circuit-switch the call.
2. If a match is not found, the algorithm attempts to connect the call to a preference for which the action to take is not block the call. With currently available options, this would be a preference where the action is to insert a modem pooling conversion resource.

Bridged Call

The ARS Toll Restriction is assigned to a class of service in Procedure 010, Word 3. The class of service is then assigned to an extension in Procedure 000, Word 1. When ARS Toll Restriction is assigned to a ***shared extension,*** the restriction applies to every image of the extension.

CDR (Call Detail Recording)

The CDR feature and the WCR feature are compatible with each other; however, some functions work differently than with the earlier networking features (AAR and ARS).

FEAC (Forced Entry of Account Codes)

The FEAC function was originally designed for use with the earlier networking feature, ARS. With ARS, the account code is entered before the network dial access code, and only the length of the account code entered is checked. With the WCR feature, string identifiers can be administered for account code prefixes in the digit analysis module. Also with WCR, account code prefixes are not necessarily limited to the first digit. This provides the ability to verify not only the account code length, but also the specific account code prefix digits entered.

With DEFINITY Generic 2.2, FEAC is supported on a network basis and is assigned using Procedure 312, Word 1. Three options are perovided:

- Encode 0 = account code not required.
- Encode 1 = account code is required and must be entered using the CDR DAC, prior to dialing the WCR network DAC.
- Encode 2 = account code is required and is entered either before or after dialing the WCR network DAC.

Of the above options, encodes 0 and 2 can be used when the WCR feature verifies account code prefixes. With encode 0, FEAC is not active and has no bearing on operations in the WCR digit analysis module. With encode 2, FEAC is active and an account code is entered either before or after the WCR network DAC. If the account code is entered before the WCR network DAC, it is not checked against string identifiers administered for the network.

See also the feature interaction with the IXC (Interexchange Carrier) Access feature.

Call Forwarding

The Call Forwarding features (Follow Me, Busy and Don't Answer, and Don't Answer) interact with the WCR feature in much the same way as they did with the earlier networking features, AAR and ARS.

- Incoming Network Calls

  Incoming network calls to an extension with Call Forwarding active will forward just like local calls.

- Call Forwarding Off Net

  The Call Forwarding—Busy and Don't Answer and the Call Forwarding—Don't Answer features cannot be used to forward calls off net.

  Call Forwarding—Follow Me can be used to forward calls off net. Calls forwarded to an off net number use the World Class Routing feature for route selection and processing. The user should verify that they can call the desired (forwarded to) number before forwarding their calls there. For this purpose of forwarding calls off net, the following rules apply:

  Forwarded-to Address Length

  Calls can be forwarded by a station to off-net addresses with up to 31 digits. However, if a call is forwarded by an attendant within a DCS (Distributed Communications System) for a station on a distant node, the forwarded to address is limited to 7 digits.

  Within a DCS (Distributed Communication System)

  If the attendant is to be able to set up Call Forwarding for stations on a remote switch, special constraints must be applied to the WCR implementation. Call Forwarding within a DCS (where DCIU messaging is involved), must be limited to networks 1 and 2. When the DCS includes switches prior to Generic 2.2, network 1 must be defined as the public network (equivalent to the ARS feature) and network 2 must be defined as a private network (equivalent to the AAR feature).

- No Trunks Available

  If there are no network routes accessible and available to forward the call off-net, the call does not forward. In this case, the call rings the originally called station, or follows the assigned coverage path if coverage criteria are met.

- Time-of-Day Plan

  A call may not forward off net if a time-of-day plan change results in routing that is denied to the call. Routing permissions are based on the originally called extension's attributes and restrictions.

- Permissions

  For FRL purposes, the higher of the calling or called party's FRL is used.

- Forwarding to Toll Addresses

  Calls forwarded off-net may use toll routes, depending on administered options. If a call forwarded off-net uses a toll route, CDR (Call Detail Recording) will show the forwarding (originally called) extension as the calling party.

Call Vectoring

The Call Vectoring feature is fully compatible with the WCR feature. Call Vectoring can cause a call to route to a WCR number. When a WCR call is routed by Call Vectoring, the FRL of the VDN is used as the default FRL. These calls will not queue, will not be prompted for an authorization code, and will not be given warning tone.

DCS (Distributed Communications System)

The DCS feature is fully compatible with the WCR feature. When DCS is in effect, either WCR or the Main/Satellite feature must be used for call routing.

The DCS feature provides feature transparency for a limited set of switch features. When used with the WCR feature, both messaging and call routing are based on either ENP Extension Number Portability) or UDP (Uniform Dial Plan) routing. DCS calls are dialed using an extension number. Calls dialed using a network DAC (standard WCR dialing procedures) are not provided with DCS transparency.

For DCS calls, WCR route selection gives preference to DCS trunks. If DCS trunks are not available, non-DCS trunks are selected and transparency is lost. DCS trunks do not need to be located in an earlier preference for this selection process.

ENP (Extension Number Portability)

Extension Number Portability is fully compatible with the World Class Routing feature. Extension Number Portability (sometimes referred to as node number routing) allows individual users to move to a different switch and take their extension number with them. The ENP feature requires careful combination of translations on all switches involved in the ***portability subnetwork.*** All switches must provide proper routing for each ported extension. In Generic 2 and System 85, there is no limit to the number of extensions that can be ported. Every extension can be ported as needed.

A single extension can be designated as "ported" in Procedure 354, Word 2. A range of extensions is specified in Procedure 354, Word 1. Each switch in the portability subnetwork is given a node number and it is this node number that is used in Procedure 354, to specify where an extension resides. When an extension number is dialed, the call is routed to WCR network 0 with an FRL of 7 for further analysis. In network 0, extension number ranges are entered as string identifiers which ***restart*** in the appropriate routing network with digit modification to convert the extension number to a private network number. With ENP, the network identified for restart is ignored because routing is not based on the network numbering plan. Special translations (in Procedure 354, Word 4) define the VNI to be used to port an extension to a node number.

In a portability subnetwork, the home location codes are shared by all switches rather than defining a specific switch. In a 5-digit extension numbering plan there is a home location code for each first digit (or first digit pair). When a ported call reaches another switch in the portability subnetwork, the home location code is recognized as a string identifier in the WCR private network digit analysis module and this string is specified as restarting in network 0 after appropriate digit modification to restore the extension number. Restarting in network 0 does not use network 0 translations for digit analysis. Rather it sends the call to the switch internal dial plan for analysis. That analysis may result in the process described in the previous paragraph to tandem the call along to the next node, or in routing the call to the local extension, whichever is appropriate.

Access to network 0 translations and to the private network(s) requires that the Standard Network option be selected in Procedure 276, Word 1, field 1.

FRL (Facility Restriction Level)

The World Class Routing feature uses FRL to select accessible trunk groups for a call. This is handled by the generalized route selection module. The FRL feature provides one means of controlling access to World Class Routing networks (on a per extension class of service or Authorization code basis). The functioning of the FRL feature with World Class Routing is described in detail earler in this chapter under the Generalized Route Selection section.

FX (Foreign Exchange) Access

The Foreign Exchange Access feature is fully compatible with the World Class Routing feature. The FX feature provides access to central offices in remote areas where an orginization experiences high calling activity (either incoming, outgoing, or both). For these situations, the FX feature can reduce toll charges.

FX trunks can be included in WCR routing patterns and use their own toll-free tables to administer the non-toll office codes that can be reached from the FX central office. When FX trunks are accessed through the WCR feature, it is not necessary for callers to dial the trunk dial access code for these trunks.

Information Systems Network Interface

ISN data stations can use the WCR feature (via the circuit switch) when placing calls over an external network. When this is done, the Modem Pooling feature may also be required If analog trunk groups are selected for the outgoing call).

ISDN—PRI (Primary Rate Interface)

ISDN—PRI trunk groups can be used as WCR routing preferences. When this is done, the type of address and numbering plan ID for each ISDN preference must be specified to match what is expected by the serving ISDN office. This specification is made (along with the Network Service Value) for WCR preferences in Procedure 322, Word 1.

If the operator assistance code or international calling prefix is dialed, these need to be deleted from the dialed number when a PRI trunk group is selected for routing. The codes are represented elsewhere in ISDN messaging and should not be a part of the ISDN Called Number IE.

Intercept Treatment

Intercept Treatment works with the WCR feature in much the same way as it did with the previous networking features, AAR and ARS. Intercept treatment, including programmable intercept, is provided for:

- Attempts to dial digits that don't match a string identifier in the network used
- Routing to VNI 0 or to a pattern that has no preferences assigned
- Attempting to place a toll call when the caller is toll denied due to class of service or the toll-free network 1 access code was used
- Failing to dial an account code when an account code is required.

IXC (Interexchange Carrier) Access

With World Class Routing, IXC selection can be automatic (as it was with the ARS feature), or an IXC Access code, in the form 10XXX or 101XXXX, can be dialed to specify the carrier to be used. Dialing an IXC code was not possible with the earlier network routing features.

Depending on the preference chosen for routing: the dialed IXC, an administered IXC, or no IXC could be sent to the next switch for call routing.

The CIC (Carrier Identification Code), the digits following the 10- or 101- prefix, is recorded in the CDR according to the following rules:

- If an IXC code is sent, the coresponding CIC is recorded.
- If an IXC code is not sent, the CIC value administered for the preference (in fields 8 through 11 of Procedure 321, Word 1) is recorded.

Last Number Dialed

The Last Number Dialed feature is compatible with the World Class Routing feature with the following constraint. The Last Number Dialed feature is limited to 20 manually dialed digits. Even though WCR can handle digit strings of up to 68 digits, calls placed using the LND feature are still limited to 20 dialed digits.

LWC (Leave Word Calling)

The Leave Word Calling feature interacts with World Class Routing through the DCS feature. Within a DCS, a station on one node can use Leave Word Calling to a station on another node via the WCR feature as long as there is a DCIU link available and at least 1 trunk connection available.

Look-Ahead Interflow

0n DEFINITY Generic 2.2 switches, the WCR feature is required to route Look-Ahead Interflow calls. In order to provide private network routing for interflow calls within WCR, the "Standard Networking" field (Procedure 276, Word 1) must be assigned.

Dialing Plan

When the WCR feature is used to route Look-Ahead Interflow calls, the digit contents of a vector-group list item for a "route to" step must conform to the dial plan of the network to be used. Besides conforming to the dial plan for the network, the vector-group list items for Look-Ahead Interflow "route to" steps must be prefixed by the appropriate network DAC. For public network routing, a "1" prefix digit for toll calls or an international access code may also be required.

When System 85s, DEFINITY Generic 2 switches, and DIMENSION FP 8, Issue 3 switches are part of a private network, the dial plan format for the network can have one of four forms:

- RNX (3-Digit Location Code) + XXXX (4-Digit Extension Number)
- RNX (2-Digit Location Code) + XXXX (4-Digit Extension Number)
- RNX (3-Digit Location Code) + XXX (3-Digit Extension Number)
- RNX (2-Digit Location Code) + XXX (3-Digit Extension Number)

When Look-Ahead Interflow calls are to be routed over public network facilities, the digit contents of a vector-group list item for a "route to" step must conform to the public network rules for DDD (Direct Distance Dialing). The DDD formats for the public network can have one of three forms:

- NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- NPA (3-Digit Area Code) + NXX (3-Digit Office Code) + XXXX (4-Digit Extension Number)
- International Telephone Number

Besides conforming to the dialing plan for the network, a pattern must be translated for the location code specified within the destination digits of a "route to" step. When this is not done, the "route to" step is treated as having an invalid destination. If the "route to" step is the final effective step in the sending (or tandeming) vector, vector processing treats the step as a "stop" step. Otherwise, vector processing continues with the next sequential step in the vector.

Queuing

At sending (or tandeming) switch, Queuing and Pattern Queuing do not apply to Look-Ahead Interflow calls. Instead, if every preference is busy, the Look-Ahead Interflow software will either retry routing the call at 2-second intervals (if final effective step) or continue vector processing with the next sequential vector step.

FRL Raising

Since Queuing does not apply to Look-Ahead Interflow calls, FRL Raising (which is invoked after the queue times out) also does not apply to Look-Ahead Interflow calls.

Bearer Capability Classification

For voice calls, the Bearer Capability Class of Service (BCCOS) is not a significant consideration. This is because voice calls are usually compatible with any carrier facility. However, the WCR feature does check the BCCOS of calls that are diverted outside the switch by a "route to" step. Therefore, when applicable, the BCC of the outgoing (best-choice) preference must be compatible with the BCC in the local calling party's COS or the BCC assigned to the incoming trunk group.

Trunk Reservation Limits

The Trunk Reservation Limit (assigned in Procedure 103, Word 1) does not prevent Look-Ahead Interflow calls from accessing the first preference. Rather, assigning a Trunk Reservation Limit to the trunk group has the effect of reserving trunks in the preference to ensure the routing of Look-Ahead Interflow calls.

Conditional Routing

The Look-Ahead Interflow feature is compatible with Conditional Routing. For Look-Ahead Interflow calls, software increments the satellite hop count whenever a "route to" step diverts a call over a routing preference with a satellite link. Also, software sends the current value of the satellite hop count as the second TCM for Look-Ahead Interflow calls.

Digit Modification

The Look-Ahead Interflow feature is compatible with digit modification trunking. For Look-Ahead Interflow calls that are routed over trunks that require conversion, the digit modification module can change the digit formats for vector-group list items so that the next switch receives the expected digits.

Time-of-Day Plan

The routing pattern selection for "route to" steps conforms to the currently active time-of-day plan. (This is the case whether the currently active time-of-day plan was invoked by an automatic plan change, clocked manual override, or manual override.)

Whenever a time-of-day plan is active where a pattern's best-choice preference results in the selection of a non-PRI trunk group, "route to" steps (if successful in diverting calls) will route the calls on a non-Look-Ahead basis.

Unauthorized Call Control

The routing of Look-Ahead Interflow "route to" steps can be blocked by unauthorized call control. Whenever a vector-group list item for the Look-Ahead Interflow feature contains a digit string that is marked for call control, vector processing at the sending switch either treats the "route to" step as a "stop" step (if the final effective step) or continues with the next sequential step in the vector.

When the digits of a "route to" destination are undefined in WCR or translate to the intercept pattern (VNI 0), the "route to" step is considered to have an invalid destination.

Toll Restriction

Toll Restriction does not limit the routing of Look-Ahead Interflow "route to" steps to an answering destination. If toll restriction is assigned to a VDN's class of service, this assignment is ignored

Tandem Processing

As part of the Look-Ahead Interflow SETUP message, an intervening (tandem) switch is always requested to route the interflow call on ISDN-Prefered basis. The tandeming switch attempts to select ISDN routes first during its route-selection process. If the tandeming switch cannot find an available ISDN route, a non-ISDN route is selected and an "Interworking" CALL PROGRESS message is returned to the sending switch. When routing is modified to a non-ISDN route, the call continues to route on a non-Look-Ahead basis.

Main/Satellite/Tributary

The Main / Satellite / Tributary feature provides a way of routing calls to other switches when an extension number is dialed that resides on the other switch. Although routing over Main / Satellite trunks is normally accomplished from the internal dial plan (using extension number steering), Main / Satellite trunks can be used in WCR network routing patterns.

When Main / Satellite trunks are used for WCR network routing digit modification must be accomplished from the digit sending module (subnetwork trunking) to configure the digits sent according to the expectations of the receiving switch. The full extension number is not always sent. Any digit sending parameters administered for the trunk group in Procedure 104, Word 2, are ignored.

Modem Pooling

The Modem Pooling feature is compatible with the World Class Routing feature. For data calls, the WCR feature will insert a Modem Pooling conversion resource in the call path as needed. Need is determined during preference selection in the generalized route selection module, based on the Bearer Capability of the selected preference (Procedure 318, Word 2). This provides an acceptable (but not preferred) route for data calls.

Personal Central Office Line

A Personal Central Office Line is a direct trunk connection to a serving CO or FX. As such it bypasses routing software on the local switch. The WCR feature cannot be used on and has no effect on calls using a Personal Central Office Line.

Precedence Calling

Incoming and outgoing calls are screened by the WCR feature. When they are identified as Precedence Calling calls, they are passed to the appropriate AUTOVON routing patterns (Procedure 305). Precedence Calling calls are not routed by the WCR feature.

NOTE: The Precedence Calling feature is usually associated with AUTOVON Access *(see* Considerations earlier in this chapter). However, Precedence Calling is not necessarily limited to use with the AUTOVON. When the Precedence Calling is used over a private network other than AUTOVON, these calls are still routed using AUTOVON routing patterns set up in Procedure 305 rather than by the WCR feature.

Incoming precedence calls must use a WCR network DAC (through trunk prefixing) to reach the AUTOVON routing software. For the purposes of Precedence Calling only, a network 1 DAC is recommended since it is available without requiring that the Standard Network option (Procedure 276, Word 1, field 1) be selected.

Queuing

The Queuing feature works with World Class Routing in much the same way as it did with the earlier networking features, AAR and ARS. If all accessible routes for a particular call are busy, the call may queue on the best choice trunk group (preference) in the routing pattern. Queuing of *international calls* is allowed with the WCR feature (this was not allowed with the earlier networking features).

The Pattern Queuing option allows any number of preferences in the pattern to be checked for an idle trunk during the entire queuing process. The switch may check the first accessible preference only (ignoring Pattern Queuing), or the first two preferences, and so forth. If Pattern Queuing indicates that three of ten preferences are to be checked, the switch still checks all ten during the "last try." When a call is placed in queue, it still queues on the first accessible choice preference (factors such as FRL and BCCOS permitting) and is restricted by that trunk group's queuing parameters (such as queue length and time-in-queue).

CAUTION: *Care must be exercised when setting the number of preferences to be included in Pattern Queuing. Increasing the number of preferences to be checked means an increase in processing time. If this added processing time does not produce a significant increase in calls served, queues could begin to fill to capacity and occupancy is increased with no offsetting benefit.*

Remote Access

The Remote Access feature works with the World Class Routing feature in the same way that it did with the earlier networking features. Network calling permissions available to remote access callers is entirely dependent on switch administration. Extension class of service 31 is used for remote access calls. Calling permission assigned to class of service 31 (combined with the assigned FRL) determine network accessibility for remote access callers.

If an authorization code is used for access to the Remote Access feature, that same authorization code will be used to determine FRL for the call when attempting to access a network routing pattern and preference. The caller will not be prompted to enter an authorization code again.

Restriction—Code Restriction

The Code Restriction feature is used to allow dialing to specific area and office codes when a CO or FX trunk dial access code is used. The Code Restriction feature has no effect on calls routed via the World Class Routing feature.

Code Restriction will not support ***interchangeable area codes*** when they are introduced. The WCR feature must be used to route calls after interchangeable area codes are used.

Restriction—Miscellaneous Trunk Restrictions

The Miscellaneous Trunk Restrictions feature has no effect on calls routed via the World Class Routing feature.

Restriction—Toll Restriction feature

The Toll Restriction feature restricts access to toll calls using a trunk dial access code. Toll restriction has no effect on calls placed via the WCR feature.

The Toll Restriction feature will not recognize ***interchangeable area codes*** as toll calls. The WCR feature must be used for toll call control when interchangeable area codes are used.

Route Advance

The Route Advance feature specifies alternative trunk groups to be used when the primary trunk group has no idle trunks. Route Advance is used only when the trunk group is accessed via a trunk group dial access code. Route Advance has no effect on calls routed via the WCR feature.

Tenant Services

The Tenant Services feature is fully compatible with the World Class Routing feature. With World Class Routing, both public and private network routing preferences can be partitioned for outgoing calls.

Incoming calls are always treated as belonging to tenant partition 0, even if the same trunk group is assigned to one or more different partitions for outgoing calls. If several tenants use private networking, it is possible to provide proper routing for incoming calls by using different networks for different tenants, and on incoming ISDN calls, use prefixing to assign the appropriate WCR network DAC. Alternatively, the calling switch can send the proper DAC for the network being used by the tenant.

Partitioning for the Tenant Services feature is a significant factor in determining the call category within the generalized route selection module of the World Class Routing feature. See the discussion on Partitions under Call Category Definition, earlier in this chapter for more detailed information on this interaction.

Touch-Tone Calling Senderized Operation

The World Class Routing feature relies on digit sending as a separate function from digit collection and analysis. For touch-tone trunks, sufficient senders are required to provide dial tone detection and transmission of touch-tone digits in traditional modules. For universal modules, the trunk circuits have the sending capability built in. If traditional modules are included in the switch configuration, at least one touch-tone sender is required even if all trunks are dial pulse trunks.

WATS (Wide Area Telecommunications Service)

The WATS feature is fully compatible with the WCR feature. The WATS feature provides a bulk toll calling capability for a specific geographical area at reduced cost. This service may be provided by the local operating company or by one or more interexchange carriers. WATS trunk groups can be included in WCR network routing patterns as needed.

Restricting Feature Use

Access to World Class Routing networks can be restricted by the following features:

- Attendant Control of Trunk Group Access feature
- Authorization Code feature
- FRL (Facility Restriction Level) feature

  The Alternate FRL function of the FRL feature is activated (or deactivated) by the attendant.

- Restrictions—Attendant Control of Voice Terminals feature

  Attendant controlled voice terminal retrictions that deny access to World Class Routing networks are the following:

  — Controlled Outward Restriction

  — Controlled Total Restriction.

- Restriction—Voice Terminal Restrictions feature.

  Fixed voice terminal restrictions (class of service) that deny terminal users access to World Class Routing networks include:

  — Origination Restriction

  — Outward Restriction

  — Terminal-to-Terminal Only Calling.

Hardware Requirements

The WCR feature requires the following specific hardware items.

For Traditional Modules:

- SN251 Touch-Tone Dialing Register/Receiver Circuit Packs (four circuits per SN251)
- SN252 Touch-Tone Calling Sender Circuit Packs (four circuits per SN252)

For Universal Modules:

- TN148C Tone Detector Circuit Packs (2 senders and 4 receivers per TN748C).

Regardless of the Module Type:

- TN492B Real-Time Clock Circuit.

  Supports automatic (or clocked) time-of-day plan changes.

Feature Administration

Assignment of the WCR feature is on a per system basis. Specific permissions including toll access are assigned on an extension class of service basis.

The WCR feature is assigned using the DEFINITY Manager II.

This feature can also be administered using the Manager IV.

The following are the applicable administration procedures.

Administration Procedures — World Class Routing

Procedure	Word	Purpose
000	3	Assigns the default BCCOS to an extension number.
010	3	Assigns station class of service restrictions including WCR Toll access for network 1 and FRL.
075	2	Displays routing pattern searches, plans, patterns and indexing schemes including bearer capability and digit sending indexes.
075	3	Displays network digit modification string searches
100	1	Assigns trunk group characteristics including DAC, trunk type, and public network access/egress.
100	2	Assigns the default BCCOS to a trunk group.
101	1	Administers trunk group characteristics such as toll restriction.
101	3	Administers trunk group prefixing for network routing.
103	1	Administers network trunk group translations, including minimum FRL, authorization code requirements, and conditional routing and second TCM.
275	1	Assigns System class of service features and characteristics including CDR and trunk-to-trunk connections.
275	3	Assigns system class of service characteristics including toll call requirements and node numbers.
276	1	Assigns feature group class of service including Standard Network. Field 1 must be set to 1 for network 0 and networks 2 through 7 to be accessible.
285	1	Assigns network parameters to the system class of service, including network dialing plan digits, authorization code requirements, sending method and interdigit timing interval.
286	1	Defines alternate FRL assignments. Also sets the time-of-day plan and control mode in effect.
287	1	Sets times for clocked manual override for the time-of-day plan. Displays current time-of-day plan and control mode.
288	2	CDR variable format
312	1	Administers network characteristics for networks 1 through 7 such as account code requirement, dial tone suppression, and toll prefix dialing requirements.
314	1	Administers network dial plan definitions used in network digit analysis. Word 1 assigns string identifiers.
314	2	Assigns analysis operations to the string identifier assigned in Word 1.
314	3	Display only. Provides information on the number of unallocated software nodes (not physical network nodes).

(Continued)

Administration Procedures — World Class Routing *(Continued)*		
Procedure	**Word**	**Purpose**
315	1-2	Display only. Provides results of digit analysis on entered digits for network analysis and debugging.
316	1	Sets start time for individual time-of-day plans.
317	1	Assigns characteristics to call categories.
317	2	Assigns a pattern number to a call category and VNI combination.
318	1	Assigns a trunk group and preference number to a routing pattern. Also assigns network routing characteristics such as FRL, warning tone, toll-free table, digit modification index digit sending index, and ISDN sending index.
318	2	Assigns the BCCOS to a pattern and preference.
319	1	Assigns digit string identifiers to a toll-free table index.
320	1	Defines a digit modification index including number of digits to delete and the digits to insert.
321	1-2	Assigns digit sending instruction to a digit sending index, including pause insertion.
322	1	ISDN sending index parameters including NSF values and IXC messaging options.
350	2	Assigns manual change time-of-day plan and network dial access codes. specific encodes are: ● Network 1 (Toll Free) = encode 32 ● Network 1 (Toll Route)= encode 33 ● Time-of-day Play change= encode 60 ● Network 2 access = encode 61 ● Network 3 access = encode 108 ● Network 4 access = encode 109 ● Network 5 access = encode 110 ● Network 6 access = encode 111 ● Network 7 access = encode 112
354	1	Assigns blocks of extension numbers to node numbers for Extension Number Portability.
354	2	Associates an extension number with a node number or DAC. This procedure is used to change the node number assignment for an extension number that has been ported to a new node.
354	3	NPA-XX assingment for ISDN message creation.
354	4	Maps node numbers from Procedure 354, Word 2 to a VNI.

Notes:

Appendix A: Configuration Limits

DEFINITY Communications System, Generic 2 and SYSTEM 85, Release 2

Configuration Limits by Version

The following tables provide a quick reference to the limits of Generic 2 and System 85, Release 2, configurations by Version. These limits are based on allocated or available memory table space. Each application has a maximum limit, and the system has a limit that further constrains the sum of all applications.

Maximum Line Records:	32,703 (in Release 2, Version 3)
Sum of Line Applications:	62,000
Difference:	29,297 (More Applications than Line Records).

That is, while a specific application may have a limit of its own, say 10,000 Multiappearance Station Sets, the system may not permit that limit to be reached. This is because the sum of related applications (Line Limits) must be less than the total of their combined maximums (in the described case, 29,497 fewer total applications).

Line Limits (Note)

Application	System 85, Release 2				DEFINITY	
	V 1	V 2	V 3	V 4	G2.1	G2.2
Analog Station sets (2500, 7100 Series)	7,000	8,000	32,200	32,000	32,000	32,000
Multiappearance Station Sets (7200H Series, 7300H Series, 7400D Series, 7500D Series)	5,000	5,000	10,000	10,000	10,000	10,000
Voice Data Stations (7400D Series with data module or cartridge and 7500D Series with ADM-T)	2,500	4,020	8,000	8,000	8,000	8,000
Display Stations (7400D or 7500 Series with display, 515 BCT and 510D)	900	2,000	5,000	10,000	10,000	10,000
Line Side Data Modules (DTDM, PDM, TDM, MDM, 7400A, EIA Ports, etc.) Plus Total Multiappearance Sets	5,000	8,040	16,000	16,000	16,000	16,000
Line Records	15,000	19,145	32,700	32,703	32,703	32,703
Dedicated Switch Connections (voice and data)	—	—	1023	1023	1023	1023

NOTE: The Line Records limit is the upper limit of combinations of other line applications. The sum of other limits exceeds this figure, but the total administered in an installation cannot.

Limits for the multiappearance sets, voice/data sets, display sets, and line-side data modules cannot be realized simultaneously. Moreover, these limits should be reduced if heavy feature usage or a heavy traffic load is anticipated.

Station Feature Limits						
Application	**System 85, Release 2**				**DEFINITY**	
	V1	**V2**	**V3**	**V4**	**G2.1**	**G2.2**
Abbreviated Dialing:						
-Characters Per Button	20	20	20	20	20	20
-Characters Per Call	36	36	36	36	36	60
-Maximum in System List	99	99	9999	9999	9999	9999
-Maximum in a Nonsystem List	30	30	95	95	95	95
-Number of Nonsystem Lists	2047	5118	13,107	52,223	52,224	52,224
-Number of Group Lists	500	1000	9999	9999	9999	9999
-Maximum in All Lists	24,000	65,535	65,535	262,143	262,144	262,144
Mnemonic (Keyboard) Dialing:						
-List Entries	—	—	300	1000	1000	1000
-Mnemonic Characters	—	—	10	10	10	10
-Characters in Number	—	—	20	20	20	20
Default (Terminal) Dialing:						
-Characters in Number	—	—	20	20	20	20
Button Table Words*	64,000	220,000	400,000	400,000	476,670	476,670
Call Pickup Groups	999	999	999	999	999	999
Coverage Groups	3000	4096	4096	4096	4096	4096
Effective Coverage Groups Dual Paths Counted as 1)	3000	3047	3047	3047	3047	3047
Dispiay-Voice Terminal:						
-Display Names	5000	8500	32,767	32,767	32,767	32,767
-Average Characters Per Name	22	22	22	22	22	22
-Maximum Characters Per Name	30	30	30	30	30	30
-Maximum Characters Per Msg (buffer)	40	40	40	40	40	40
Intercom Records:						
-Auto/Manual	300	300	300	300	300	300
-Dial	280	280	280	280	280	280
Last Number Dialed:	3000	6000	6000	6000	6000	6000
-Maximum Digits	—	—	—	20	20	20
Line Appearances and Images:						
-Appearances Per Extension	12	12	12	12	12	12
-Images Per Line Appearance	16	16	16	16	16	16
-Images Per Extension	192	192	192	192	192	192
-Images Per Terminal	30	30	30	30	52	52
Leave Word Calling Messages on the Switch	3000	6000	6000	6000	6000	6000
Extension Classes of Service	63	63	63	63	63	63
Message Waiting Lamps (Auto)	7500	10,500	32,000	32,000	32,000	32,000
Message Waiting Lamps (Auto) Per Extension	3	3	3	3	3	3

* See the next table for Button Table Word requirements.

Button Table Word Requirements	
Application	**V1 to V4 Requirement**
2500, 7101A, 7102A, 7103A, and Encore Station Sets	None
Straight Line Sets	1 Word Each Unit
7203H, 7303S, 7401D, 7403D, and 7410D Station Sets, and 10-Button MET Sets	12 Words Each Unit
7205H, 7305S, 7405D, 7434D, PC/PBX Station Sets, and 20- and 30-Button MET Sets	36 Words Each Unit
7404D Station Sets	8 Words Each Unit
7406D and 7406D With Display Station Sets	32 Words Each Unit
CALLMASTER Station Set	36 Words Each Unit
510D Terminal	36 Words Each Unit
515 BCT	12 Words Each Unit
C201A, C401A, and C401B Call Coverage Modules	20 Words Each Unit
Display Modules	8 Words Each Unit
Data Modules (PDM, TDM, DTDM, MDM, EIA Port)	2 Words Each Unit
Dual Port Data Modules and ADFTC	4 Words Each Unit
One Button Transfer	1 Word Each Unit
F201A and F401A Function Key Modules	24 Words Each Unit
Manual Message Waiting Button	1 Word Each Unit
Stations Signaled by Station Busy	2 Words Each Unit
Stations Signaled by Manual Signaling	1 Word Each Unit

(Continued)

Button Table Word Requirements *(Continued)*	
Application	**G2 Requirement**
2500, 7101A, 7102A, 7103A, and Encore Station Sets	None
Straight Line Sets	1 Word Each Unit
7401D Station Sets	11 Word Each Unit
7203H, 7303S, 7403D, and 7410D Station Sets	12 Words Each Unit
7205H, 7305S, 7405D, 7434D, PC / PBX Station Sets	36 Words Each Unit
7404D Station Sets	8 Words Each Unit
7406D and 7406D With Display Station Sets	30 Words Each Unit
CALLMASTER Station Set	30 Words Each Unit
7505D and 7506D Station Sets (with or without data)	19 Words Each Unit
7507D Station Set (with or without data)	42 Words Each Unit
10-Button MET Set	7 Words Each Unit
20-Button MET Set	17 Words Each Unit
30-Button MET Set	27 Words Each Unit
510D Terminal	21 Words Each Unit
515 BCT	12 Words Each Unit
C201A, C401A, and C401B Call Coverage Modules	20 Words Each Unit
Display Modules	7 Words Each Unit
Data Modules (PDM, TDM, DTDM, MDM, EIA Port)	2 Words Each Unit
Univasal Data Modules (UDM)	4 Words Each Unit
Dual Port Data Modules and ADFTC	4 Words Each Unit
One Button Transfer	1 Word Each Unit
F201A and F401A Function Key Modules	24 Words Each Unit
Manual Message Waiting Button	1 Word Each Unit
Stations Signaled by Station Busy	2 Words Each Unit
Stations Signaled by Manual Signaling	1 Word Each Unit

System Parameters Limits						
Application	**System 85, Release 2**				**DEFINITY**	
	V1	**V2**	**V3**	**V4**	**G2.1**	**G2.2**
ACD (EUCD):						
-Agents	—	512	1024	1024	1024	2048
-Agents Measured (by CMS)	—	—	1023	1023	1023	1023
-Service Observers (Active)	—	—	64	64	64	64
-Splits	—	30	30	60	60	60
-Recorded Announcements	—	30	30	84	84	255
Answer-Back Channels:						
-Call Park and Loudspeaker Paging*	9	9	9	9	9	9
-Code Calling Access	6	6	6	6	6	6
Attendant Features:						
-Conference Bridges	13	13	13	13	13	13
-Console Positions	28	40	40	40	40	40
-Switched Loops Per Console	6	6	6	6	6	6
-Switched Loops	168	240	240	240	240	240
-Remote Console Positions (ORPI)	—	40	40	40	40	40
-Consoles (100s Groups)	100	100	100	100	100	100
-Originating Registers (ORs)	28	40	40	40	40	40
-Voice Terminal Restriction Groups	63	63	63	63	63	63
Call Forwarding—Follow Me:						
-Off-Net Forwarding Relationships	3200	3200	3200	3200	3276	3276
Call Vectoring:						
-Number of Vectors	—	—	—	128	128	511
-Steps per Vector	—	—	—	15	15	15
-Recorded Announcements	—	—	—	84	84	255
Calling Number Display Units	20	20	20	20	6	6
DCIUs						
-DCIU Links	8	8	8	8	8	8
-APs per DCIU	7	7	7	7	7	7
-AUDIX Adjuncts per DCIU	—	4	4	8	8	8
-DCS Links per DCIU	8	8	8	8	8	8
-Logical Channels per Link	64	64	64	64	64	64
Maximum Digits in Dial Access Code	3	3	3	4	4	4
Dial Access Codes (Feature and Trunk)	175	500	1104	1104	1104	1102
Dial Pulse and Touch-Tone ORs	246	246	246	246	246	458
Total ORs	300	300	300	300	300	512

* The Call Park and Loudspeaker Paging Access features share the same nine answer-back channels.

(Continued)

System Parameter Limits *(Continued)*

Application	System 85, Release 2				DEFINITY	
	V1	V2	V3	V4	G2.1	G2.2
DS1:						
-DS1 Circuit Packs Per Switch	255	255	511	511	511	511
-Circuit Packs Per Universal Carrier						
Line Side	—	—	—	—	20	20
Trunk Side	—	—	—	—	10	10
-Line Side DS1 Circuit Packs per DS1 Carrier	—	4	4	4	4	4
-Trunk Side DS1 Circuit Packs per DS1 Carrier	2	2	2	2	2	2
-73-Series Port Cicuits Per DS1 Port Carrier	—	16	16	16	16	16
FADS:						
-CAS Display Units	1	1	1	1	1	1
-UCD Display units	12	—	—	—	—	—
Listed Directory Numbers (DID)	4	4	9	999	999	999
Loudspeaker Paging Zones	18	18	18	18	18	18
Network Modules	18	31	31	31	31	31
Remote Modules	—	15	30	30	30	30
Network Cabinets Per Module						
Traditional Module	4	4	4	4	4	4
Universal Module	—	—	—	—	1	1
Port Carriers Per Module						
Traditional Module	12	12	12	12	12	12
Universal Module	—	—	—	—	3	3
PCC (Processor Communications Circuit) Circuit Packs	—	—	—	3*	3*	3*
Port Circuit Packs:						
-Per Traditional Port Carrier	16	16	16	16	16	16
-Per Universal Port Carrier	—	—	—	—	21†	21†
Recorded Announcements (Non ACD/EUCD)	2	15	15	15	15	15
System Status Indicator Lamps	128	168	168	168	168	168
Tenant Services:						
-Extension Partitions	—	—	—	1000	1000	1000
-Extension Partition Group	—	—	—	500	500	500
-Attendant Partitions	—	—	—	41	41	41
CDR:						
-Number of Data Item Encodes	—	—	—	76	76	76
-Maximum Record Length	18	18	18	24	24	24
-Maximum LSUs	2	2	8	8	8	8
Malicious Call Trace:						
-Maximum Simultaneous Traces	—	—	—	15	15	15

* Only ***one*** PCC port can be used and this port can only be used as a CDR port.
† This includes the service slot which can only accommodate a TN748C Tone Detector.

Trunk Limits (per Switch)						
Application	**System 85, Release 2**				**DEFINITY**	
	V1	**V2**	**V3**	**V4**	**G2.1**	**G2.2**
ANI Boards	2	2	2	2	2	2
Contact Interface Boards	34	34	34	34	34	34
Preselected Call Routing Groups	255	255	255	982	982	982
Preselected Call Routing Trunks Per Trunk Group	99	99	99	99	99	99
Personal Central Office Lines (Trunks)	150	150	150	150	150	150
AIOD Queues	6	6	6	6	6	6
Trunks, Physical (Including Host Access and Modem Pooling)	2250	5000	6000	6000	6000	6000
Trunk Records, Assignable (Outgoing Trunk Queues, Physical Trunks, and Trunk Intercom Records)	2705	7525	10,500	10,500	10,500	10,500
Trunk Records (Total)	3250	7970	11,046	11,046	11,046	11,046
Trunk Groups:						
-Modem Pooling Trunk Groups	175	238	238	982	982	982
-Host Computer Access Trunk Groups	175	238	238	982	982	982
-Total (Including Host Access and Modem Pooling)	255	255	255	999*	999*	999*
-Trunk Group Dial Access Codes	255	255	255	999	999	999
-Trunks Per Trunk Group (Modem Pooling)	99	99	99	99	99	99
-Trunk Per Trunk Group (Host Computer Access)	99	99	99	99	99	99
-Trunks Per Trunk Group (Others**)	99	255	255	255	255	255
RLTs (Release Link Trunks):						
-Inward†	110	110	110	110	110	110
-Outward	16	16	16	16	16	16
-Groups at Main (CAS Branch)	40	40	40	40	40	40
Remote Access Trunks	45	45	45	6000	6000	6000
Restriction Levels (Code Restriction)	4	4	4	4	4	4
Route Advance, Trunk Groups Per Pattern	5	5	5	5	5	5

* The first 17 trunk groups are dedicated to internal service facilities. The number of trunk groups available for customer use is 982.

** Trunk groups 16 and 17 can have a maximum of 458 trunks in Generic 2.2.

† The limit of 110 for RLTs is imposed by the number of System Status Indicator (SSI) that can be used. If RLTs are not monitored by SSIs, the limit is the same for other types of trunks.

Network Parameter Limits						
Application	**System 85, Release 2**				**DEFINITY**	
	V1	**V2**	**V3**	**V4**	**G2.1**	**G2.2**
AAR *(System 85 and Generic 2.1)*:						
-Patterns	255	255	640	640	640	—
-Trunk Groups Per Pattern	4	4	16	16	16	—
-Conditional Routing Call Categories	—	—	3	3	3	—
-Maximum Valid RNXs	780	780	780	780	780	—
-Maximum Number of Routes	10,240	10,240	10,240	10,240	10,240	—
ARS *(System 85 and Generic 2.1)*:						
-Patterns Per Plan	64	64	64	64	64	—
-Plans—Time Dependent	3	3	3	3	3	—
-Trunk Groups Per Pattern	16	16	16	16	16	—
-Call Categories for Tenant Services	—	—	—	64	64	—
-Foreign NPAs (6-Digit Translation)	64	64	160	160	160	—
-Patterns Per 6-Digit Translation	4	4	10	10	10	—
-Maximum Number of Routes	1024	1024	1024	1024	1024	—
Unauth. Call Control/10- to 7-Digit Conversion						
-3-Digit NPAs	None	None	None	None	None	—
-6-Digit NPA-NXX Combinations	500	500	500	500	500	—
-7-Digit NPA-NXX-X Combinations	2048	2048	2048	2048	2048	—
-8-Digit NPA-NXX-XX Combinations	2048	2048	2048	2048	2048	—
-9-Digit NPA-NXX-XXX Combinations	2048	2048	2048	2048	2048	—
-l0-Digit NPA-NXX-XXXX Combinations	2048	2048	2048	2048	2048	—
Generic 2.2 M to N Conversions	—	—	—	—	—	4095
Authorization Codes	9000	9000	90,000	90,000	90,000	90,000
Facilities Restriction Levels	8	8	8	8	8	8
DCS:						
-Maximum Nodes	12	20	20	63	63	63
-Maximum ES Nodes	—	—	—	63	63	63
-Maximum Nodes with Attendants Centralized at One Node	12	20	20	40	40	40
-Maximum Nodes per AUTOVON Interface	12	20	20	40	40	40
Maximum Extension Numbers Per Network (5 Digit Dialing)	—	—	—	100,000	100,000	100,000
Maximum NPA-NXX Designators	99	99	99	99	99	

(Continued)

Network Parameter Limits *(Continued)*						
Application	**System 85, Release 2**				**DEFINITY**	
	V1	**V2**	**V3**	**V4**	**G2.1**	**G2.2**
ISDN:						
-Codesets	—	—	—	8	8	8
-Codepoints Per Codeset	—	—	—	1024	1024	1024
-Mappings Per Codeset	—	—	—	—	256	256
-Codeset Mappings	—	—	—	—	16	16
-Maximum ISDN Call Records	—	—	—	6000	15,000	15,000
-Maximum Calls on D-channel	—	—	—	23	500	500
-Maximum D-channels						
BRI:	—	—	—	—	10,000	10,000
PRI:	—	—	—	512	512	512
-Maximum NFAS D-channel Groups	—	—	—	—	255	255
-Maximum Number for Interface ID	—	—	—	—	32	32
-Bearer Capability Classes of Service	—	—	—	—	256	256
WCR:						
-Maximum Networks	—	—	—	—	—	8*
-Access Code Length	—	—	—	—	—	4
-Maximum Digit string Length	—	—	—	—	—	31
-Call Categories	—	—	—	—	—	256
-Time-of-Day Plans	—	—	—	—	—	8
-Plan Changes Per Day	—	—	—	—	—	6
-Patterns	—	—	—	—	—	1023
-Preferences Per Pattern	—	—	—	—	—	16
-Toll-free Tables	—	—	—	—	—	63
-Toll-free Table Maximum String Length	—	—	—	—	—	7
-Maximum Digits Deleted	—	—	—	—	—	31
-Maximum Digits Inserted	—	—	—	—	—	31

* Network numbers range from 0 through 7. Network 0 is reserved for the internal (local) dialing plan.

Notes:

Appendix B: Call Distributors

Comparison of Call-Distribution Features

The following table provides a quick comparison of the functions provided by the various call-distribution features during Release 2.

Function	DDC R2 V1	UCD R2 V1	EUCD R2 V2	ACD R2 V3	ACD R2 V4	ACD G2.1	ACD G2.2
Maximum Number of Splits (Groups) in Switch	28 (shared)	28 (shared)	30	30	60	60	60
Maximum Number of Agents (Members) in Switch	1120 (shared)	1120 (shared)	512	1024	1024	1024	2048
Maximum Number of Agents in Single Split	40	40	512	1024	1024	1024	1024
Incoming Call Queue for Each Split	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Priority Portion of Incoming Call Queue	No	No	Yes	Yes	Yes	Yes	Yes
Direct Hunting	Yes	No	Yes	Yes	Yes	Yes	Yes
Circular Hunting	No	Yes	Yes	Yes	Yes	Yes	
Most Idle Agent Distribution	No	No	No	Yes	Yes	Yes	Yes
Time Interval Between Scans for an Available Agent (in seconds)	2.0	2.0	0.2	0.2	0.2	0.2	0.2
Agent Override	No	No	Yes	Yes	Yes	Yes	Yes
Service Observing	No	No	No	Yes	Yes	Yes	Yes
30A8 Queue Warning Lamps	Yes	Yes	Yes	Yes	Yes	Yes	Yes
106B Agent Status Lamps	No	No	Yes	Yes	Yes	Yes	Yes
106B Assignments Based on Extension Number	No	No	No	No	No	No	Yes
FADS Measurement of Agent Activity	No	Yes	No	No	No	No	No
CMS Measurement on AP 16 or 3B2/5	No	No	No	AP,3B	3B	3B	3B
Stroke (Event) Count Measurements	No	No	No	Yes	Yes	Yes	Yes
Multiple Agent Work States	No*	No*	Yes	Yes	Yes	Yes	Yes
Automatic Answering and Zip Tone	No	No	Yes	Yes	Yes	Yes	Yes
Unique Announcement From Each Split to Calling Party	No	No	Yes	Yes	Yes	Yes	Yes

* Group busy and member busy are available using an access code.

Function	DDC R2 V1	UCD R2 V1	EUCD R2 V2	ACD R2 V3	ACD R2 V4	ACD G2.1	ACD G2.2
System-Wide Recorded Annct. to Calling Party	Yes*	Yes*	Yes	Yes	Yes	Yes	Yes
Internal Callers Receive Anncts.	No	No	Yes	Yes	Yes	Yes	Yes
Verification of Announcements	No	No	Yes	Yes	Yes	Yes	Yes
Music After Announcement(s)	No	No	Yes	Yes	Yes	Yes	Yes
Multiple Music Sources	No	No	No	Yes†	Yes	Yes	Yes
Intraflow	No	No	Yes	Yes	Yes	Yes	Yes
Interflow	No	No	Yes	Yes	Yes	Yes	Yes
Look-Ahead Interflow	No	No	No	No	Yes‡	Yes	Yes
Queue-of-Origin Announcements	No	No	Yes	Yes	Yes	Yes	Yes
City-of-Origin Announcements	No	No	Yes	Yes	Yes	Yes	Yes
Queue-of-origin Displays	No	No	Yes	Yes	Yes	Yes	Yes
City-of-Origin Displays	No	No	Yes	Yes	Yes	Yes	Yes
DNIS (Dialed Number ID Service)	No	No	Yes	Yes	Yes	Yes	Yes
Abandon Call Search	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Automatic Available Splits	No	No	No	Yes	Yes	Yes	Yes
Queue-Status Display	No	No	No	No	Yes	Yes	Yes
Multiple Call Handling	No	No	No	No	Yes	Yes	Yes
Malicious Call Trace	No	No	No	No	Yes	Yes	Yes
Call Vectoring	No	No	No	No	Yes	Yes	Yes
RLT (Release Link Trunk) Termination to ACD split or VDN	No	No	No	No	No	Yes**	Yes
Lamp Indication for Stroke-Count Buttons	No	No	No	No	No	No	Yes

* This announcement can be periodically repeated
† Beginning with Issue 1.4 of R2 V3.
‡ Beginning with Issue 1.3 of R2 V4.
** Beginning with Issue 3.0 of DEFINITY Generic 2.1.

Appendix C: Administration Facilities

Call-processing software in both the System 85 and DEFINITY Generic 2 switches uses a variety of tables located in system memory to keep track of:

- Port circuit addresses
- Trunk circuit addresses
- Extension numbers
- Feature assignments
- Voice terminal button assignments
- Network configuration
- System configuration
- System-wide options.

Collectively, these tables are called ***translations.*** Switch are shipped with a tape which includes generic software and translations. When the switch is installed, the tape is used to load the main memory. The information used to create these translations comes from the order form received by the factory. A printed record of these translations is provided by the Customer System Document (shipped with each switch).

Since translations are stored in memory, information such as the number of modules the system has, the equipment location of a data module, or the features a voice terminal may access can easily be changed. The process of changing translations is called administration. System 85 uses a series of programs called "Procedures" to change translations. A person wanting to administer the system (change translations) can access the procedure programs by using one of the following system Management vehicles:

System 85 Management

- DEFINITY Manager IV The Manager IV (formerly the CSM [Centralized System Management]) is a forms-based CRT screen user interface. Manager IV maintains databases which contain copies of translations for one or more System 85 switches. This allows an administrator to change translations off-line from the switch. Manager IV is then programmed to contact the System 85 at a later time; when contact is made, Manager IV tells the switch to execute the procedures needed to make the corresponding changes to switch translations.
- MAAP (Maintenance and Administration Panel) — A numeric key pad, a 25-digit display, and a set of flipcharts are used to access Procedures (procedures). (Flipcharts are cards attached to the MAAP that allow the user to interpret the data appearing on the 25-digit display.) Procedures are used to change translations. The MAAP also uses another category of Procedures to perform maintenance tests.

- VMAAP (Visual Maintenance and Administration Panel) — A program that in a UNIX System environment which provides a CRT screen interface that emulates the 25-digit display of the MAAP. In order to interpret the numbers appearing on the screen, a user must have a copy of the flipcharts for the software version being run on the switch. VMAAP terminals access this software through an RS-232 link which connects to the System 85 remote port interface circuit.
- SMT (System Management Terminal) — A unit almost identical to the MAAP which offers access to a subset of the Procedures available through the MAAP.

(The following item does not apply to R2 V4 or Generic 2)

- FM (Facilities Management) — A feature on the AP 16, available to the switch administrator via a BCT. This feature is used for network administration.

(The following item does not apply to R2 V4 or Generic 2)

- TCM (Terminal Change Management) — A feature on the AP 16, available to the switch administrator via a BCT. This feature is used for feature assignment and terminal rearrangement.

(The following item does not apply to R2 V4 or Generic 2)

- RMATS-II (Remote Maintenance, Administration, and Traffic System II) — A service available from an AT&T service center. This service has the same administrative capabilities as the MAAP. It can also be used to run maintenance routines.

All of these system management vehicles work with older versions of System 85. But, the AP 16 system management features and RMATS-II, which were available as system management vehicles for older versions, are not supported in R2 V4.

Manager IV, VMAAP, AP 16, TCM and FM, and RMATS-II all use the TN492 remote port interface circuit to access the switch. For switches prior to R2 V3, only one of these tools can use the remote port at any given time. For R2 V3 and later switches, two simultaneous connections are allowed if one is accessing maintenance procedures and the other is accessing administrative procedures. Two simultaneous administrative sessions or two simultaneous maintenance sessions are not allowed.

DEFINITY Generic 2 Management

The two major differences in administration between DEFINITY Generic 2 and System 85 are the DEFINITY Manager II, and the addition of General Terminal Administration.

DEFINITY Manager II

DEFINITY Generic 2 uses a series of programs called Procedures to change translations. In System 85, devices such as the MAAP (Maintenance and Administration Panel) and the SMT (System Management Terminal) are used to access the Procedure programs.

With DEFINITY Generic 2, the MAAP and SMT are replaced by the DEFINITY Manager II, a PC (Personal Computer) based administration and maintenance tool.

DEFINITY Manager II has three modes of operation: basic, enhanced, and task.

Basic Mode

In this mode the DEFINITY Manager II emulates a MAAP by displaying field numbers and their values on the screen (see Figure C-1). Paper copies of the Procedures (called Flipcharts) are needed to interpret the display. This mode of operation gets all of its Procedure display information from the switch. DEFINITY Manager II can also be used from off-site locations via a dial-up connection to a TN492C remote interface port or the TN563 SCSI adapter.

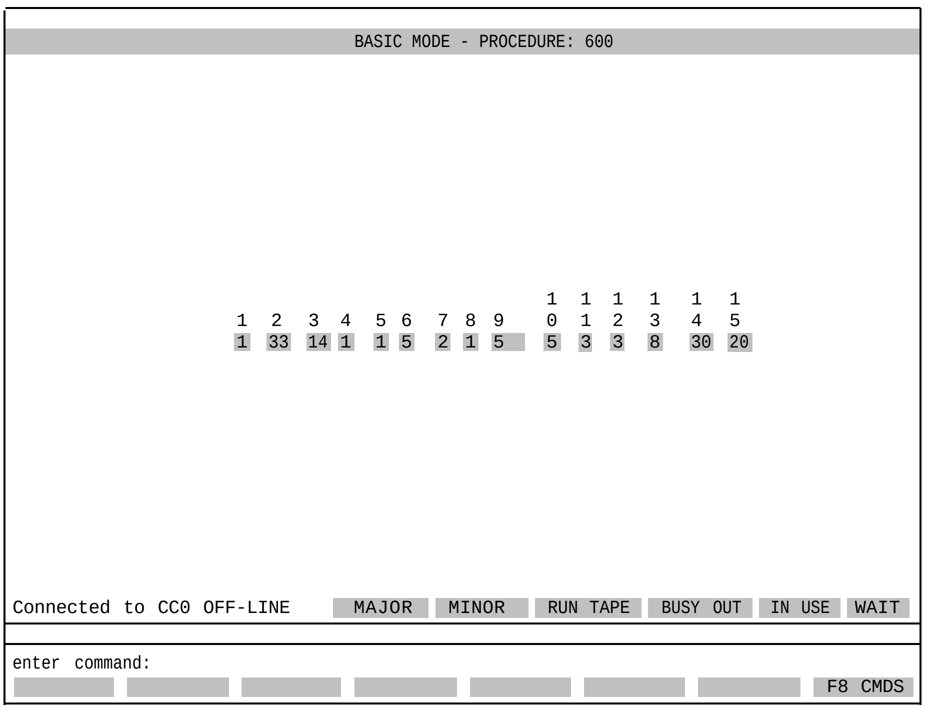


Figure C-1. Sample of Basic Mode Screen

Enhanced Mode

In this mode the DEFINITY Manager II serves as an intelligent MAAP emulator by using a version-dependent database called the SSB (Switch Support Base). The SSB supplies an electronic version of a flipchart whenever a Procedure is accessed (see Figure C-2). (The SSB is not used with basic mode.) Field numbers and names are displayed. Moreover, encode definitions, notes, on-line help, and error information are available to the user. In most cases, this supplementary information goes beyond what has traditionally been offered on flipcharts in System 85. In this mode, DEFINITY Manager II can also be used from an off-site location via a dial-up connection to a TN492C remote interface port or the TN563 SCSI adapter.

```
                    ENHANCED MODE - PROCEDURE: 100, WORD: 1
                            TRUNK GROUP TRANSLATION

 1. Trunk Group: ---

DIAL ACCESS CODE/TRUNK ID CODE
   2. Digit 1: --
   3. Digit 2: -
   4. Digit 3: -
   5. Digit 4: -

 6. Trunk Type: ---
 7.      Dial Access Restriction: -
 8.   Personal CO Line Appearance: -
 9. Public Network Access/Egress: -

DISPLAY ONLY
  10. Signaling Type: ---



Connected to CC0 ON-LINE  MAJOR  MINOR  RUN TAPE  BUSY OUT  IN USE  WAIT

enter command:
                  F3 DATA            F5 HELP  F6 FIELD  F7 INPUT  F8 CMDS
```

Figure C-2. Sample of Enhanced Mode Screen

Task Mode

In this mode the DEFINITY Manager II presents task-oriented screens to the user (see Figure C–3). The task mode provides station administration for all predefined terminal types. In the task mode, DEFINITY Manager II can also be used from off-site locations via a dial-up connection to a TN492C remote interface port or the TN563 SCSI adapter.

```
add station 12345                                                  Page  1 of 7
                                      STATION

   Extension: 12345
        Type: ____________                              Origination: prime
   Equip Loc: __/ / / /__                 COS: ___      Termination: ______

        Name: __________________________

  FEATURE OPTIONS

            LWC Dstionation: -                  Call Coverage Group: ____
                  AP Number: -             Coverage Msg Retrieval? n
       AUDIX Machine Number: -                    Call Pickup Group: ___
              Auxiliary ANI? n                    Hunt-To Extension: _____

      Automatic Msg Waiting? n               Bearer Capability COS: 0
  Audible Auto Msg Waiting? n        Dedicated Switch Connection? n
       Attd Cont Rest Group: -



enter command:
F1 CANCEL  F2 REFRESH  F3 SUBMIT  F4 CLRFLD  F5 HELP          F7 NXT PG  F8 PRV PG
```

Figure C-3. Sample of Task Mode Screen

General Terminal Administration

General terminal administration simplifies the administration of multiappearance voice terminals and data modules by making it easier to specify how a voice terminal is equipped and by allowing new terminal types to be defined. Procedure 50 has been added to make this possible. Also, procedures 51 and 58 have been modified to take advantage of procedure 50. (For detailed information on using these procedures, please refer to the *System Administration Manual — 555-104-506.)*

Terminal Types

As in previous releases, voice terminals and data modules have "type" encodes for each different model. For example, a model 7405D voice terminal will always have a "type" encode of 45. Encodes 1 through 99 are reserved for AT&T terminal types (see Table C-A).

TABLE C-A. Terminal Type Encodes for DEFINITY Generic 2

Encode	Terminal Type	Encode	Terminal Type
1	PDM (Processor Data Module)	43	7403D
2	TDM (Trunk Data Module)	44	7404D
3	Dual Port Data	45	7405D
5	ADFTC (A/D Fac. Test Circuit)	46	7406D (without display)
7	EIA Port	47	7407D
10	Straight Line Set	55	7505 ISDN—BRI Set
15	10-Button MET Set	56	7506 ISDN—BRI Set
16	20-Button MET Set	57	7507 ISDN—BRI Set
17	30-Button MET Set	62	CALLMASTER Voice Terminal
23	7203H	64	7406D (with display)
25	7205H	70	7434D
33	7303S	91	510BCT
35	7305S	95	515BCT
41	7401D	99	DCP PC/PBX
42	7410D		

Encodes 100 through 200 are available to the administrator for defining "new"* terminal types. In practice, these definitions will generally be limited to other vendor's ISDN—BRI terminals or to future AT&T voice terminal models. Here are the attributes that are specified in order to create a new terminal type:

- **Circuit Pack Type**
  - — SN270/TN754 DCP port
  - — TN556 BRI port
  - — SN228/SN229 or TN742B/TN746 analog port
  - — ANN17/TN762B hybrid port (7300S series)
  - — SN261 facilities test circuit

* Terminals must be electrically similar to, and operate like, existing terminal types. General Terminal Administration is primarily intended to be an inventory tool for keeping track of what equipment is being used. This is accomplished by eliminating the need to assign a new type of terminal with an old terminal type.

- — SN238/TN726 ADU compatible data line port
- — TN735 MET line port

- Data Type
  - — No data
  - — Optional data
  - — Integrated data
  - — Data only
- Display Type
- Number of Buttons On Voice Terminal: 0 through 62.

Terminal Options

In addition to terminal type, a separate "option" encode has been introduced for specifying how a terminal is configured. For example, if the 7405D is equipped with a display and a DTDM, it will have an "option" encode of 10. A complete list of option encodes appears in Table C-B.

TABLE C-B. Option Encodes for DEFINITY Generic 2

Encode	Terminal Type
0	Data only
1	Voice only
2	Basic terminal, feature module
3	Basic terminal, coverage module
4	Basic terminal, feature module and coverage module
5	Basic terminal, display module
6	Basic terminal, data
7	Basic terminal, feature module and display module
8	Basic terminal, feature module and data
9	Basic terminal, coverage module and data
10	Basic terminal, display module and data
11	Basic terminal, feature module, coverage module, and data
12	Basic terminal, feature module, display module, and data

Notes:

Appendix D: Data Communications

Several different forms of digital communications services are provided by System 85 and DEFINITY Generic 2 features. These are digital switches. As such, they are particularly well suited to provide a variety of digital services. Tables D-A and D-B summarize the data communications features available with both the System 85 and DEFINITY Generic 2 as well as the services these features provide.

Digital Features and Services

TABLE D-A. Summary of Internal Digital Features and Services

Feature	Dgital Format Used	Service Provided
Data Call Setup	DCP (Digital Communications Protocol)	Data and digital voice calling and signaling within the local switching network.
Data Protection	Not Applicable	Protection from intrusion warning tone for internal data calls.
Data Communications Access	Analog	Analog line or trunk interface for local host computers.
Host Computer Access	DCP	Digital line or trunk interface for local host computers.
PC Interface	DCP	Switch feature and service access for local Personal Computers.
Modem Pooling	Analog/DCP	Inserts conversion resources for calls transiting between analog and digital formats.

TABLE D-B. Summary of Digital Interface Features

Feature	Digital Format Used	Service Provided
ACCUNET Service Interface	BOS	Trunking: CO or Special Access
DMI (Digital Mutiplexed Interface):	24th Channel Signaling	Host Computer Services:
	BOS	Provides 23, 64 Kbps clear data channels over standard DS1 (AVD) trunking facilities.
	MOS	Provides 23, 64 Kbps clear data channels over ISDN trunking facilities.
DS1 Interface		
Voice Grade Service	In Channel (Bit Robbed)	Trunking for CO, FX, ETN connections Supports digitized voice communications and data communications with modems. Data rates are limited to 19.2 Kbps.
AVD (Alternate Voice Data)	24th Channel Signaling	Trunking CO, FX, ETN. Supports digitized voice and data communications without modems up to 64 Kbps clear channel.
ISDN—BRI (Integrated Services Digital Network /Basic Rate Interface)	D-Channel MOS Signaling	Line side Voice and Data Services
ISDN—PRI (Intiegrated Services Digital Network /Primary Rate Interface)	24th Channel MOS Signaling	Trunk side Voice and Data Services
ISN (Information System Network) Interface	DCP/EIA	Data Communications Interface Services (between switch and ISN Local Area Network).

BOS (Bit Oriented Signaling)
MOS (Message Oriented Signaling)

Basic Digital Signals

Unipolar Signals

Basic digital signals are carried over low voltage dc (direct current) circuits. This is the type of signaling used at the I/O (Input/Output) ports of most data terminal devices. The most common form these signals take is unipolar. With this form of signaling, zero voltage is used as the reference level and a pulse deviation (plus or minus the carrier voltage) is used to indicate a bit change. Figure D-1 represents a unipolar bit train.

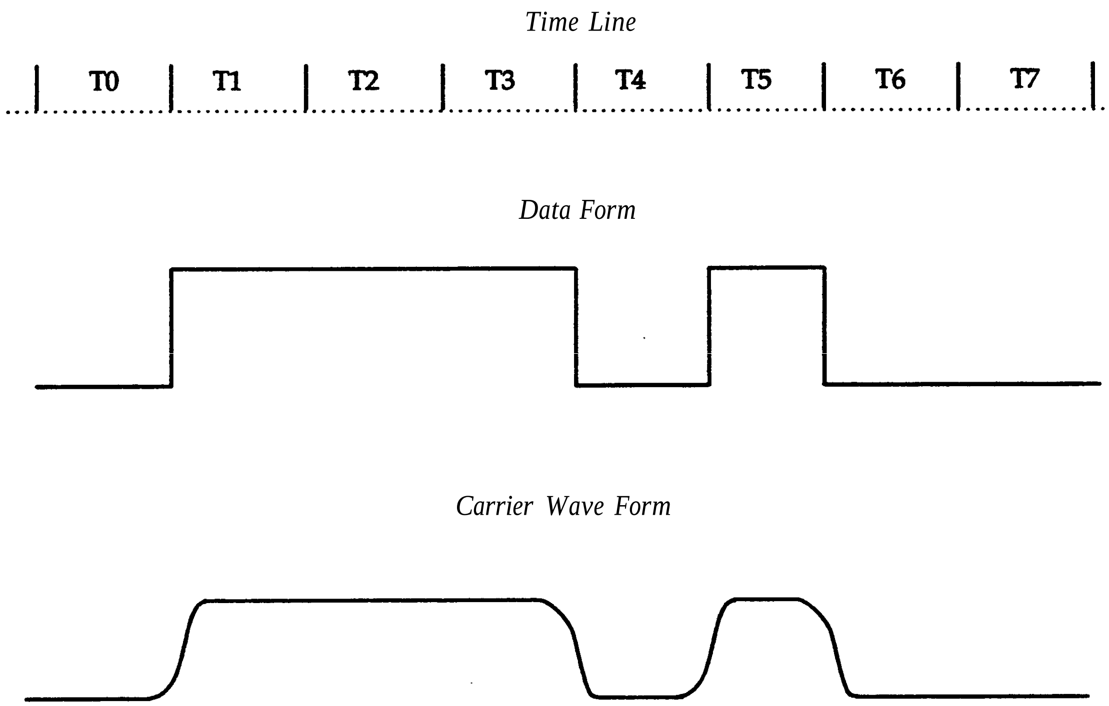


Figure D-1. Unipolar Digital Format

The pulse may represent either a bit (binary 1) or a no bit (binary 0).

This is a satisfactory arrangement for short range transmissions. However, when greater ranges are needed, problems arise. Voltages are typically in the range of 5 to 12 volts dc. Greater effective ranges can be achieved by raising the voltage levels. However, this increases the rise time for each pulse and has a detrimental effect on transmission rates. Regenerative repeaters can also be used; however, this increases costs and also limits available transmission rates. A cost effective way to overcome these problems is to convert the signals to an ac (alternating current) carrier.

Bipolar Signals

The use of bipolar signaling offers an attractive alternative to unipolar signaling.

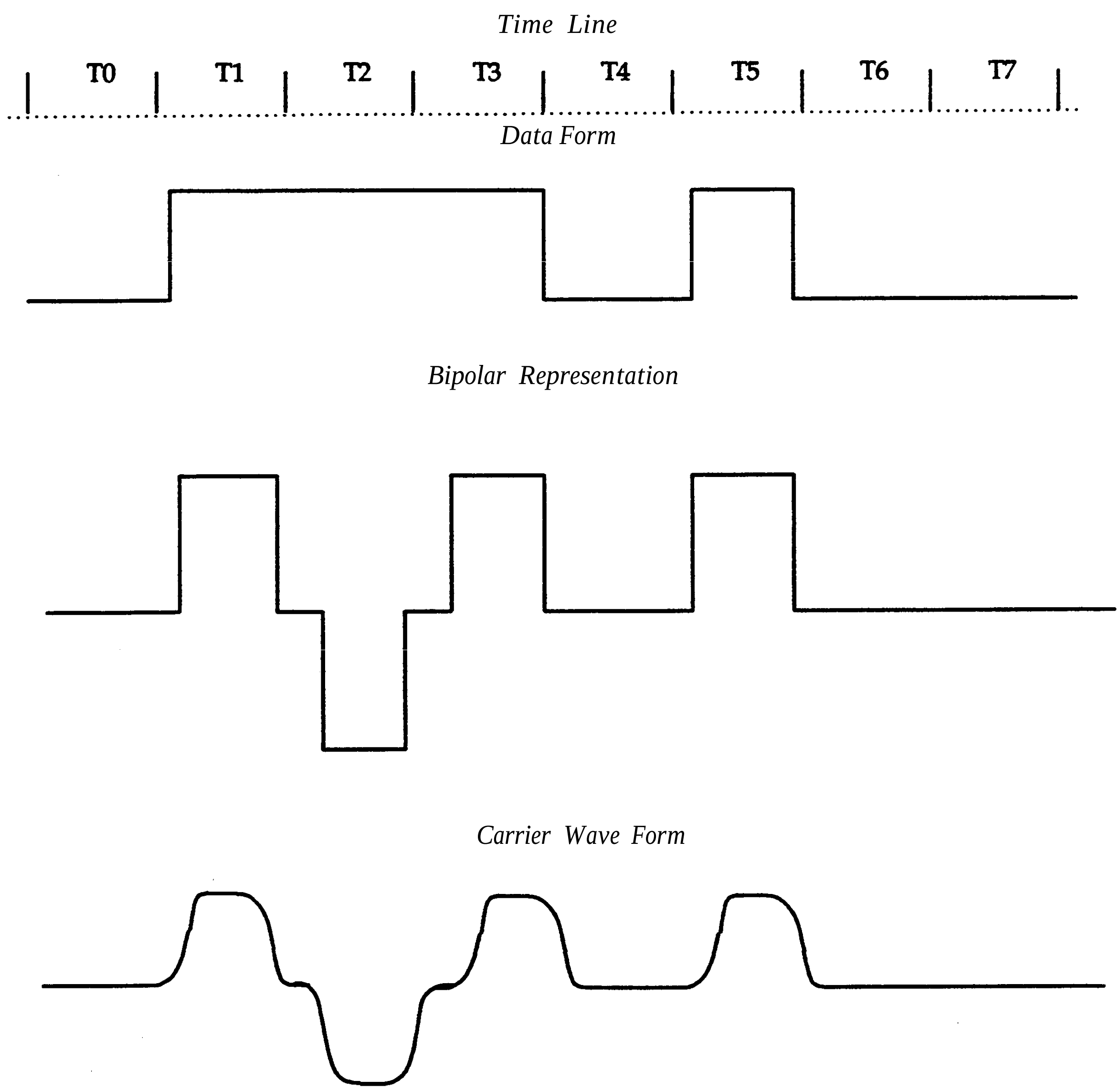


Figure D-2. Bipolar Digital Format

Data Communications Applications

Some features, such as ISN Interface, are directed toward specific data applications. Others are more general and provided opportunities for individualized applications. The following is an application that is derived from the more general feature, Data Call Setup.

Host Computer Dialing

The DCP (Digital Communications Protocol) and the Keyboard Dialing function of the Data Call Setup feature provide a simple and flexible call interface that can be used by program controlled computers as well as by human controlled terminals.

To take advantage of the terminal dialing capability available to a program controlled computer on the either a System 85 or DEFINITY Generic 2 switch, the programmer needs detailed knowledge of dialing formats, call progress messages, and control-lead state changes during the course of a call. The following provides the information needed to prepare a Host Computer Dialing program.

Data Modules

Several types of data modules are available for use with customer provided equipment. They all support keyboard dialing. However, three data modules are specifically applicable to the Host Computer Dialing application:

MPDM (Modular Processor Data Module)

The MPDM is a stand-alone DCE (Data Communications Equipment) device. This data module supports synchronous or asynchronous operation from 300 bps to 19.2 Kbps and synchronous operation at 56 Kbps and 64 Kbps. It also provides a "LOW" mode that operates by blind sampling at 19.2 K samples per second. Keyboard dialing is not supported at "LOW", 56 Kbps or 64 Kbps. The MPDM can be configured with an RS-366 autodialer interface.

MTDM (Modular Trunk Data Module)

The MTDM is a stand-alone device that provides a DTE (Data Terminal Equipment) interface. It supports the same speeds and modes of operation as the MPDM. The MTDM also supports a "switched" mode that is used with the Modem Pooling feature.

EIA Port Board and MADU (Multiple Asynchronous Data Unit)

The SN238, EIA Port Board, and MADU configuration forms a reduced cost device that provides a simplified DCE type interface. This is not a data module as such, but serves the same purpose. One RS-232 control lead is supported in each direction. Receive Line Signal Detect and Clear to Send are tied to Data Set Ready, and Data Terminal Ready is inferred as Request to Send. The EIA Port Board MADU configuration supports asynchronous-only operation from 300 bps to 19.2 Kbps. An optional speed-selection algorithm (autobaud) is also available.

For use with the Host Computer Access feature, the Z3A3, ADU can be used rather than the MADU. In effect, the MADU is eight ADUs combined in a common housing. The functionality is the same, but the MADU wiring and installation is more compact.

Keyboard Dialing

Keyboard dialing is supported in either asynchronous or synchronous mode at data rates from 300 bps to 19.2 Kbps. The ASCII character set is used. In ***synchronous mode,*** all characters are sent in start-stop form.

Control Functions

Control of keyboard dialing is accomplished through the combination of data module settings (such as, data rate and sync/async mode), interface control leads (Data Set Ready and Clear to Send), and data transfer (e.g., the dialed numbers and call progress messages).

For human users, the call progress messages are sufficient. However, computers can react more quickly than humans and can take advantage of information provided by interface leads.

Dialing Character Set

Address information is supplied by the calling device (data terminal or computer) on the TD (Transmit Data) lead [RD (Received Data) lead on the data module (MTDM)]. Call progress information is supplied by the switch on the RD lead (TD lead on the MTDM).

The following characters can be used in the dialing (called number) string:

0 to 9	The decimal digits.
A to Z **a to z**	Alphabetic characters, both upper and lower case. These are used for Mnemonic Dialing (Release 2, Version 3 and later).
# and *	The nondigit symbols on the touch-tone dial pad.
+	Control character — Wait for dial tone.
, or %	Control character — Pause, currently 1.5 seconds. The comma is the preferred symbol. However, it is not supported in old releases. The percent (%) will be phased out in later releases of System 85.
() - **and (sp)**	Readability characters (ignored by switch processing).

_ or (bs)	Printing and non-printing editing characters (destructive backspace). Either one (undersore or backspace) causes erasure of the last remaining (preceding) character in the entered string (including readability characters). Sequential erase characters effect sequential preceding characters, i.e., 577 31(bs)(bs)(bs)(bs)6 2118 yields 5762118 when outpulsed as a digit string.
(cr) or (lf)	Signals the completion of the dialing string. The System 85 takes no action on an entry until CR (carriage return) or LF (line feed) is entered.
@	Line-erase character. The current string is disregarded. If the previous character entered was not another "@," a new dial prompt is issued.

Differences Between Versions

- Release 2, Version 2 and Earlier Switches

  Mnemonic dialing is not available on Release 2, Version 2 and earlier switches. If unrecognized characters (i.e., letters) are entered and not erased before completion (line feed or carriage return), an error will be indicated and the call will be disconnected.

- Release 2, Version 3 and Later Switches

  Mnemonic dialing allows the use of letters in a dialing string. Up to ten alphabetic characters can be used.

Overview —The Dialing Sequence

Data Call Setup using keyboard dialing proceeds like voice call setup.

- The calling device (data terminal or computer) signals the data module to go off-hook and then waits for a prompt. (Equivalent to dial tone in a voice call.)
- When the DIAL prompt is received, the desired number is entered.
- Call progress indications are provided by the switch.
- When the called number answers, the data module enters a handshake sequence to set and verify consistent operating parameters.
- Upon successful completion of the handshake routine, the data module enters the data mode, and call setup is complete (login message from distant processor).
- If the handshake squence fails, the caller is informed by a message, and the call is torn down (DISCONNECTED).

Switchhook Control

Call Origination

The calling device signals the data module to go off-hook with a "break" (long space) of at least ten bit periods. If the data module is optioned for ***autoanswer,*** the data module will go off-hook automatically when alerted for an incoming call when DTR is asserted. If the autoanswer is disabled, a "break" is required.

Call Disconnect

The data module may be put in the on-hook state by dropping DTR for at least 50ms. The controlling device should hold DTR down until the data module responds by dropping DSR. If the long space disconnect option is enabled on the data module, a long space of 2 seconds or more will cause the data module to go on-hook and disconnect the call. In either case, disconnect is effected by dropping DTR or transmitting a 2-second break.

Call Progress Messages

Outgoing Call Messages

Call progress is indicated to the calling device (data terminal or computer) by a series of messages generated by the switch. Generally, these messages correspond to the audible tones that would be received during an equivalent voice call. The messages are shown in Table D-C.

TABLE D-C. Call Progress Messages — Call Origination

Leading Control Chars.	Message	Trailing Control Chars.	Meaning
(cr)(lf)	DIAL:	(sp)	This message corresponds to dial tone. The calling device (data terminal or computer) can enter the "address" or number to be called.
(cr)(lf)	RINGING		This message corresponds to ringback tone, and indicates that the called device is being rung.
(cr)(lf)	ANSWERED	(cr)(lf)	This message corresponds to the removal of audible ringback when the called device answers the call.
(cr)(lf)	BUSY		This message corresponds to busy tone.

TABLE D-C. Call Progress Messages — Call Origination (Contd)

Leading Control Chars.	Message	Trailing Control Chars.	Meaning
	CHECK(sp)OPTIONS		This message is unique to data call setup. It indicates that the data modules involved cannot find a mutually agreeable set of operating parameters. On an outside call, this may mean that no suitable modem pooling conversion resource is available. In any event, handshaking has failed, and this message is normally following by **DISCONNECTED.**
(cr)(lf)	DISCONNECTED	(bell)	This message appears when the data module goes on-hook (disconnects). It may be followed by an explanatory message: **-TRANSFER** The data call has been returned to a voice set via a data button press. **-OTHER(sp)END** The far end has hung up. **(sp)-CALL(sp)FAILED** The call has failed for some unspecified reason
(cr)(lf)	TRY(sp)AGAIN		This message corresponds to reorder tone (fast busy). It indicates that a facility needed for the call (e.g., modem pooling conversion resource) is not available. A **DISCONNECT** message will follow immediately. The call may be retried later.
(cr)(lf)	DENIED		This message corresponds to intercept tone, and indicates that the call is not permitted (e.g., undefined restriction). A **DISCONNECT** will follow immediately The call is guaranteed to fail again if retried.

TABLE D-C. Call Progress Messages — Call Origination (Contd)

Leading Control Chars.	Message	Trailing Control Chars.	Meaning
(cr)(lf)	CONFIRMED		This message corresponds to "confirmation" tone (three tone bursts), and indicates that a dialed entry (usually a feature dial access code) has been accepted by the switch. This message is usually associated with feature activation or deactivation (e.g., Call Forwarding, Automatic Callback, etc.). A **DISCONNECT** message will follow immediately.
(cr)(lf)(bell) (bell)(cr)(lf)	CALLBACK, PLEASE(sp)ANS-		These messages are given to the originating device (terminal or computer) when an automatic callback call is being returned. The call originating device should respond with a "BREAK" (long space) to bring its data module off-hook. The switch will then respond with an **ANSWERED** message, followed by appropriate call progress messages.
(cr)(lf)	AUTOCALL		This message is sent to the originating device in place of **DIAL:** when it is attempting to originate a call while it has been selected to receive an Automatic Callback call. This only occurs after the Automatic Callback sequence has begun (to the original calling station) but has not yet been completed. This message is followed by call progress message **RINGING** (referring to alerting taking place at the call originating station).
	CONNECTED	(cr)(lf)	This message is given to the originator of an Automatic Callback call, in place **ANSWERED,** when the called party actually answers first. This situation arises from a race condition when the called party goes off-hook (to originate a call) before the Automatic Callback caller responds to the **CALLBACK** message.

TABLE D-C. Call Progress Messages — Call Origination (Contd)

Leading Control Chars.	Message	Trailing Control Chars.	Meaning
	ANSWERED	(cr)(lf)	This message appears in response to an off-hook from the data module being called.
(cr)(lf)	WAIT,	(sp)	This message may be sent to the originator of a call requiring a Modem Pooling conversion resource when the response at the far end is delayed. It has no meaning other than as a "keep alive" message.
(cr)(lf)	CODE:	(sp)	This message is sent to a call originator when an Authorization Code is required to access facilities needed to complete the call (e.g., Default FRL is too low to use available trunks). Calling device must enter an authorization with a higher FRL, followed by (cr) or (lf) and wait for further call progress messages. Otherwise, the call may not complete.
	TOLL(bell)(sp)CALL!		This message indicates that the requested call is being routed over toll facilities, and the originating terminal is toll restricted. A **DISCONNECT** message will follow immediately.
(cr)(lf)	TIME(sp)OUT		This message indicates that an off-premises call has received no change in call progress for some time (usually 10-seconds). A **DISCONNECT** message will follow immediately.
(cr)(lf)	ANSWERED-NOT(sp)DATA		This message indicates that the far end has answered an outgoing modem pool call, but the "noise" detected is not recognizable as data answer tone. A **DISCONNECT** message will follow immediately.

Incoming Call Messages

When keyboard dialing is enabled on a data module, and the attached device (data terminal or computer) has DTR (data terminal ready) asserted, the switch will supply notification of incoming calls. The messages are shown in Table D-D.

TABLE D-D. Call Progress Messages — Terminating Call

Leading Control Chars.	Message	Trailing Control Chars.	Meaning
(cr)(lf)(bell)	INCOMING(sp)CALL-		This is the normal "ringing" message (the MPDM and DTDM also assert Ring Indicator). If autoanswer is not enabled, the data terminal or computer must enter a "BREAK" (long space) to bring the data module off-hook.
	ANSWERED	(cr)(lf)	This message is supplied by the switch when the data module goes off-hook
(cr)(lf)	PLEASE(sp)ANS-		This message is supplied when a data call is transferred to the data module (via the Data Button) from a voice terminal that originated the call. Ringing indicator is not inserted when this message appears (or would appear). If the attached data station does not assert DTR until RI is asserted, it will not know that a call is present.
	ABANDONED	(cr)(lf)	This message is supplied to the called device if the caller abandons the call before it is answered. This message may be preceded by a **DISCONNECTED** message if the caller abandons as the call is being answered.
(cr)(lf)(bell)	FORWARDED		This message is sent to a called device when an incoming call is received, and the Call Forwarding—Follow Me feature is active. It corresponds to "Ring Ping" given at a voice terminal as a reminder the forwarding is active.

TABLE D-D. Call Progress Messages — Terminating Call (Contd)

Leading Control Chars.	Message	Trailing Control Chars.	Meaning
	FORWARDED	(cr)(lf)	This message is sent to a called data station when an incoming call is forwarded by the Call Forwarding—Don't Answer feature. This message may be preceded by a **DISCONNECTED** message if the call is answered just as the switch decides to forward the call.
	TRANSFER	(cr)(lf)	This message is sent when an Unanswered incoming call is "intercepted" by a Return-to-Voice operation from a voice terminal. In the case of a race condition (data module answers just as the call is returned to voice), a **DISCONNECTED** message will precede the **TRANSFER** message.

Interface Lead States

Monitoring and control of the data module interface leads is important for proper operation of programmed devices (computers). In some cases, information is available from these leads that is not available from the text messages provided by the switch. In particular, after call setup is completed (the ANSWERED message is sent to the caller), the MPDM and the DTDM drop DCD (Data Carrier Detect) and CTS (Clear to Send) while a handshake sequence is conducted between the endpoints. Transmission of user data must wait until DCD and CTS return to the on state.

The interface leads abbreviations, their RS-232 pin assignments, and their applications are shown in Table D-E.

TABLE D-E. RS-232C Interface Leads

Abbreviation	Pin Number	Description
TD	Pin 2	Transmit Data - Data from a DTE (e.g., MTDM) to a DCE (e.g., MPDM).
RD	Pin 3	Receive Data - Data from a DCE to a DTE.
RTS	Pin 4	Request to Send - Request from a DTE to a DCE for permission to transmit.
CTS	Pin 5	Clear to Send - Permission from a DCE to a DTE for the DTE to transmit data.
DSR	Pin 6	Data Set Ready - Signal from a DCE to a DTE that the DCE is ready for operation. When a DCE type data module goes on-hook, this lead is dropped. If keyboard dialing is enabled in the DCE, DSR stays low for a least 50ms. DSR is then reasserted if DTR is asserted by the DTE.
SG	Pin 7	Signal Ground - Common return for all other signals.
DCD	Pin 8	Received Line Signal Detector - Indicates that communication is established with the other end. This lead is commonly called (Data) Carrier Detect.
CI	Pin 12	Signal Rate Indicator 1 - Indication from DCE (e.g., 2224A modem) to DTE (i.e., MTDM) of data rate.

TABLE D-E. RS-232C Interface Leads (Contd)

Abbreviation	Pin Number	Description
CI2	Pin 13	Signal Rate Indicator 2 - Indication from DCE (e.g., 2224A modem) to DTE (e.g., MTDM) of data rate.
TC	Pin 15	Transmitter Signal Element Timing - Clock provided from a DCE to a DTE for transmitted data (TD). Only used in synchronous operation, when the DCE provides the transmit clock.
RC	Pin 17	Receiver Signal Element Timing - Clock provided from a DCE to a DTE for receive data (RD). Used only in synchronous operation.
CH2	Pin 19	Data Rate Selector 2 - This lead, along with CH (Pin 23) permit selection by a DTE of the transmission rate of a DCE. This lead is supported only by the MTDM with the SW option selected.
DTR	Pin 20	Data Terminal Ready - Indication from a DTE to a DCE that the DTE is ready (e.g., to acceptor make a call). If the autoanswer option is selected on the DCE (e.g., MPDM), it will automatically answer a call when the DTE asserts DTR. The MTDM will drop DTR when it goes on-hook. If the PL option is selected on the MTDM, it will reassert DTR after 50ms. All DCE type data modules will go on-hook (and drop DSR) when the attached DTE drops DTR for at least 50ms.
RI	Pin 22	Ring Indicator - Indication from a DCE type data module to a DTE that an incoming call is present. The RI lead is asserted continuously (not pulsed) until the call is answered or abandoned.
CH	Pin 23	Data Rate Signal Selector - Indication from a DTE to a DCE to select one of two operating speeds of the DCE. The MTDM supports selection of three speeds when the SW option is selected.
XTC	Pin 24	Transmitter Signal Element Timing - External transmit clock supplied from a DTE to a DCE when the DTE is the source of timing.

For simplicity, Table D-E does not include the test indication and control leads supported by most data modules and modems.

Basic Operational Sequence

The normal sequence of operation when a DTE is connected to a System 85 or DEFINITY Generic 2 DCE with keyboard dialing enabled is as follows:

1. DTE asserts DTR.
2. Data module asserts DSR CTS, and DCD.
3. DTE transmits "BREAK" to bring data module off-hook.
4. DTE receives `DIAL:` prompt.
5. DTE transmits called number in ASCII characters (e.g., 82133), followed by a carriage return or a line feed.
6. DTE receives call progress messages.
7. When (or if) the called station answered, the data module DCE drops CTS and DCD while the handshake sequence is performed.
8. When (or if) handshake is successful, the data module reasserts CTS, DCD, and sets the CI lead, if appropriately optioned, to indicate the selected speed.
9. Data transmission takes place.
10. If the transmission path is interrupted for 4 seconds, the data module will drop CTS and DCD until the path is restored. If the SIGLS (signal loss disconnection option) is enabled, the data module will go on-hook and drop DSR, CD, and CTS, for at least 50ms.
11. When the call is completed, the DTE may disconnect by dropping the DTR for at least 50ms, or if the DISC (long space disconnect) option is selected, by transmitting a long break of at least 2 seconds. (Clearly, tipping DTR is easier for a program-controlled DTE.) The data module will respond by dropping DSR, CTS, and DCD for at least 50ms.
12. After DSR has been low for 50ms, the data module is ready to initiate or receive another call. If DTR is asserted by the DTE, the data module will assert DSR, CTS, and DCD.

The sequence for receiving an incoming call is as follows:

1. When the incoming call is received, the data module asserts RI.
2. If DTR is already asserted by the DTE and the data module is optioned for keyboard dialing the data module will have DSR, DCD, and CTS asserted, and the DTE will receive the `INCOMING CALL` message.

   If keyboard dialing is not enabled, the data module will have DSR, DCD, and CTS low.
3. If the auto-answer option is enabled, the data module will go off-hook whenever DTR is asserted by the DTE.

   If keyboard dialing is not enabled, the data module will raise DSR at this time.

4. If auto-answer is not enabled, the DTE must assert DTR and transmit "BREAK" to the data module to bring it off-hook. The data module raises DSR at this time if keyboard dialing is not enabled.
5. When the data module answers the call and keyboard dialing is enabled, the DTE will receive the `ANSWERED` message.
6. After it answers the call, the data module enters a handshake sequence with the calling data module. The data module asserts DSR, but holds DCD and CTS low during handshaking. Operation from this point is identical to that described in the origination sequence.

Keyboard Dialing Examples

The following diagrams illustrate the behavior of the major EIA control leads during keyboard dialing data call setup. These diagrams use the convention of showing ASCII control characters as lower case abbreviations, in parentheses. The first example, in Figure D-3, shows a station-to-station call.

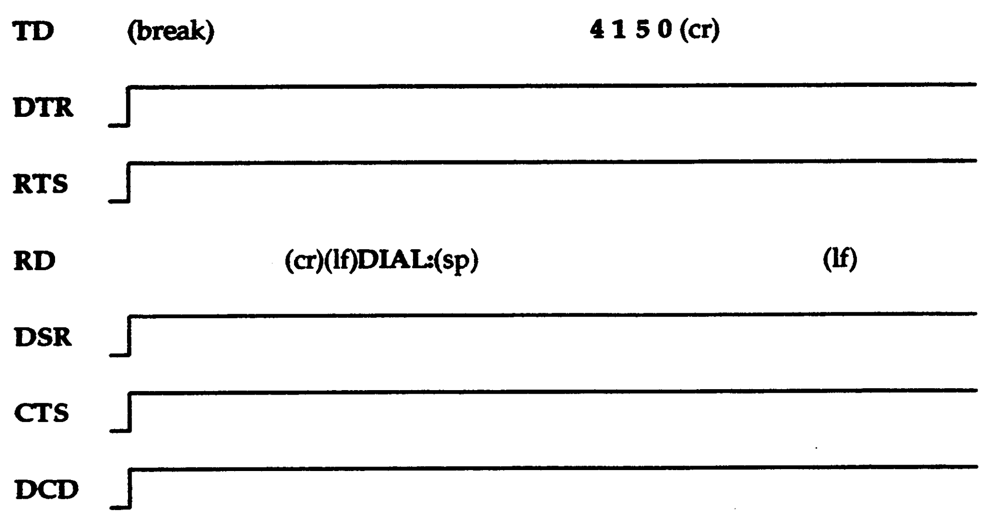
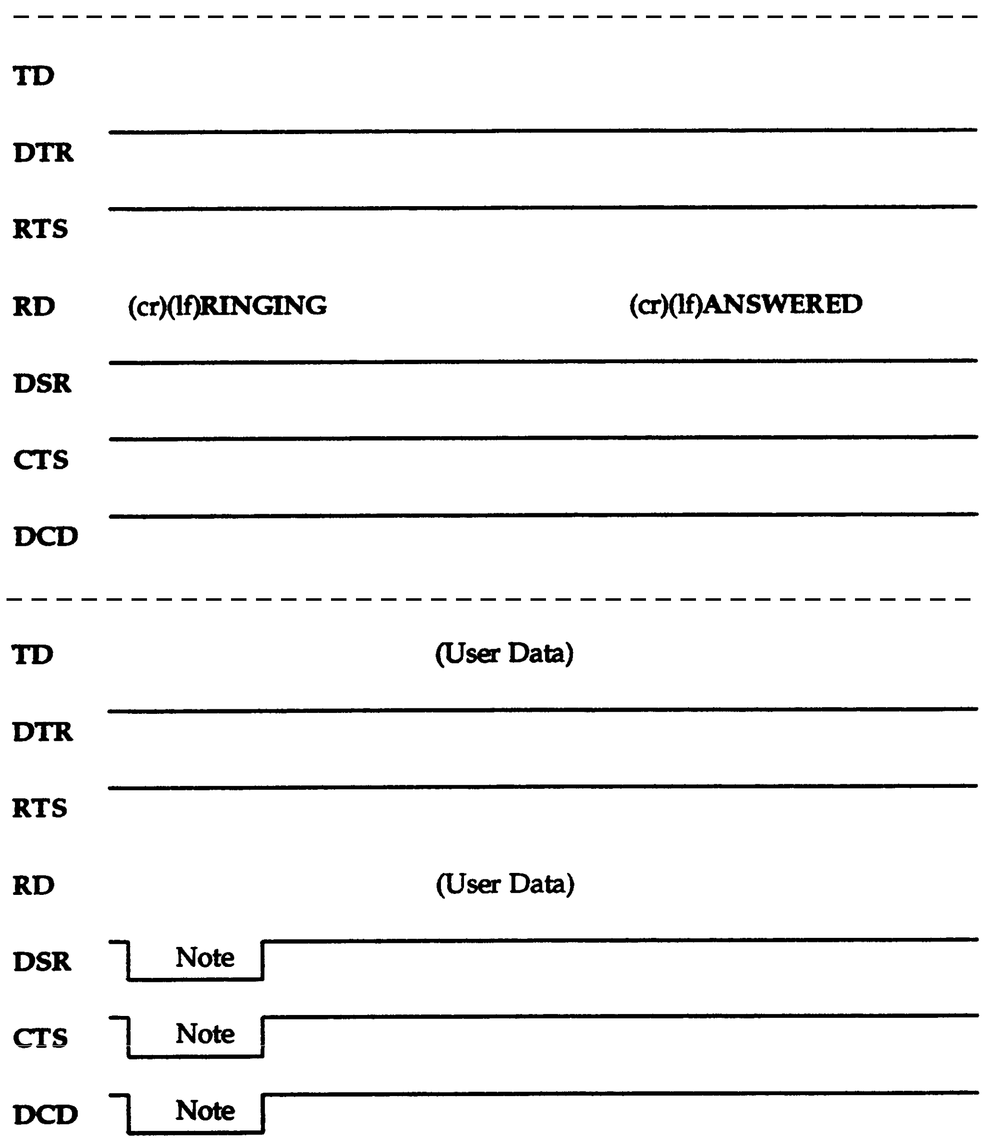


Figure D-3. Control Sequence — Station to Station Call (Sheet 1 of 2)



Note: Handshake between calling and called data modules.

Figure D-3. Control Sequence — Station to Station Call (Sheet 2 of 2)

Figure D-4 shows an outgoing off-premises call using Modem Pooling. The WAIT message may not appear if the call propagation time is short.

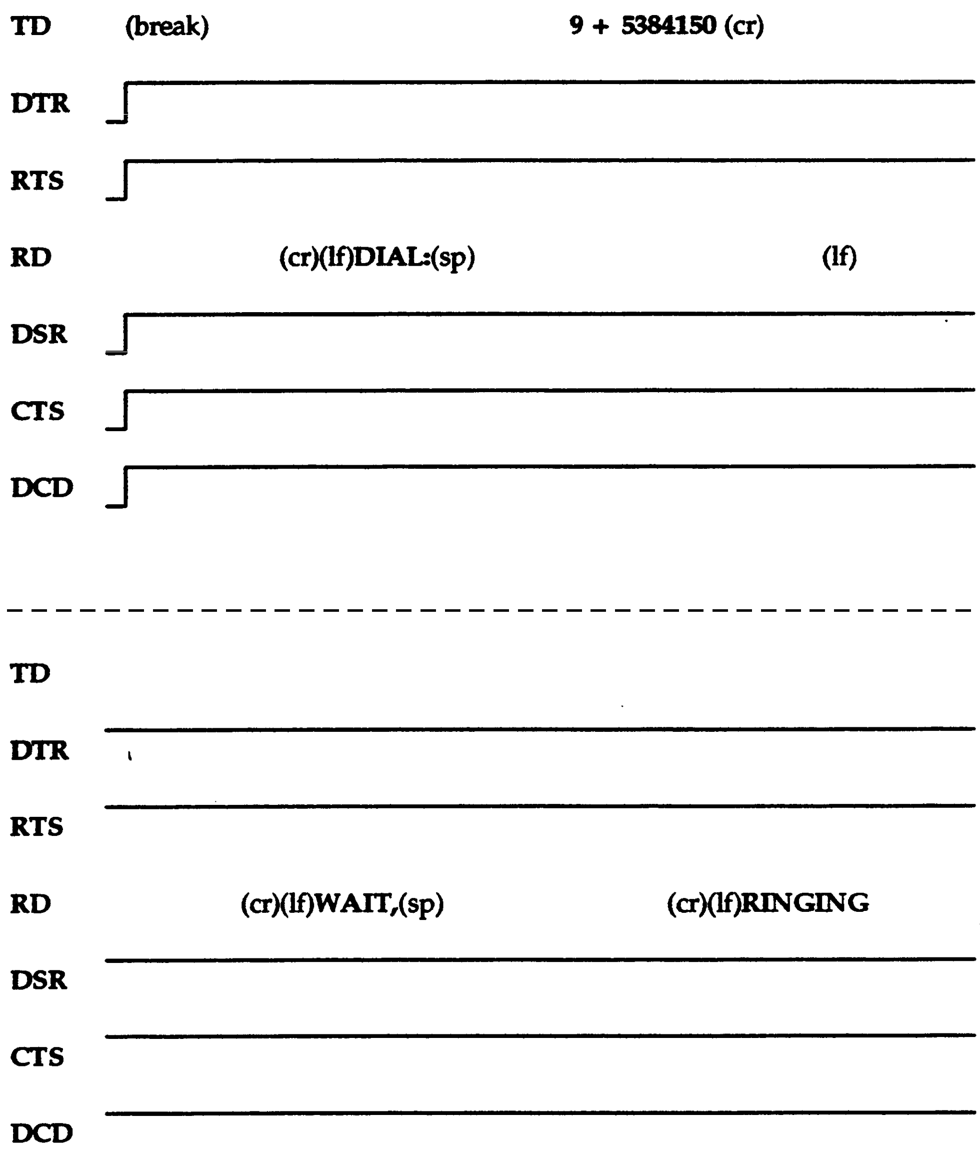
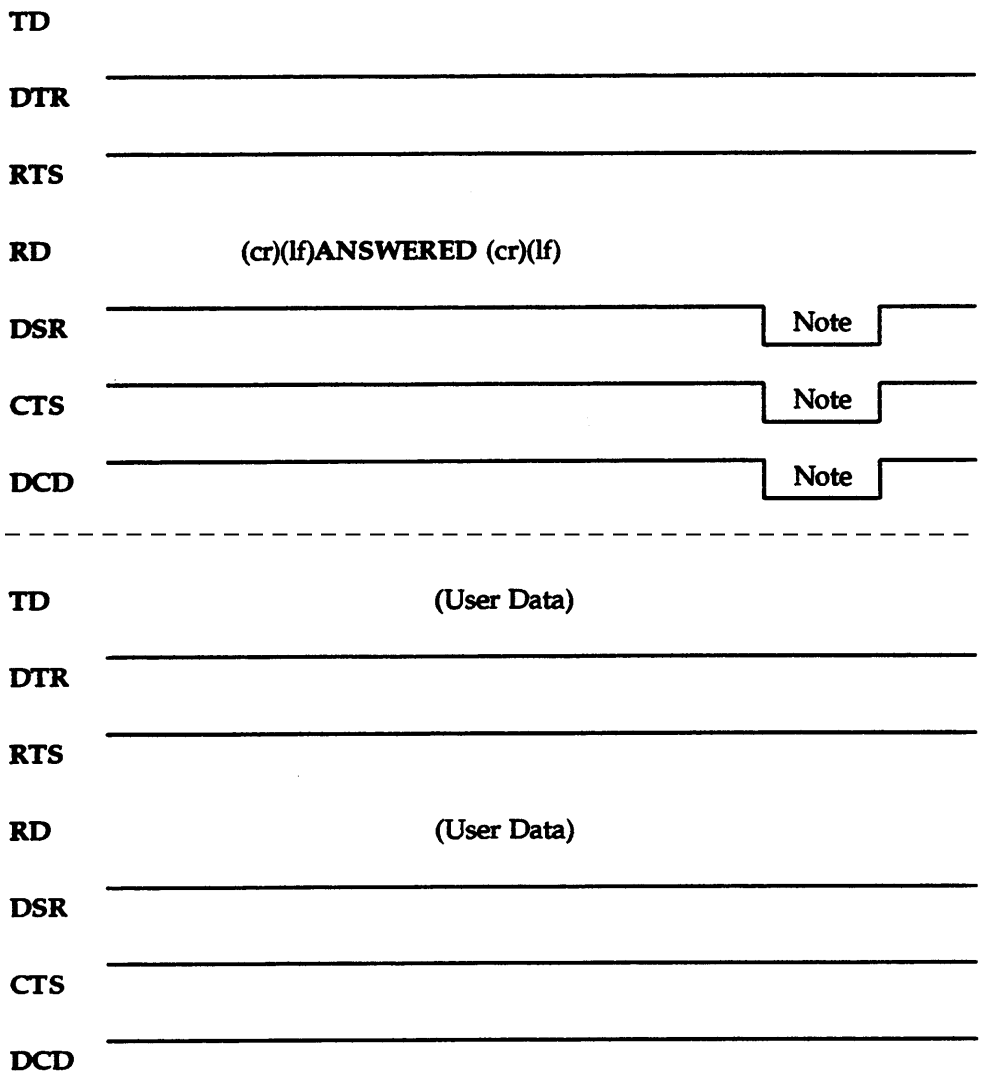


Figure D-4. Control Sequence — Off Premises Call (Using Modem Pooling)



Note: Handshake internal (data modules, then modems).

Figure D-4. Control Sequence — Off-Premises Call (Sheet 2 of 2)

Figure D-5 shows a failed station-to-station call with the called extension busy.

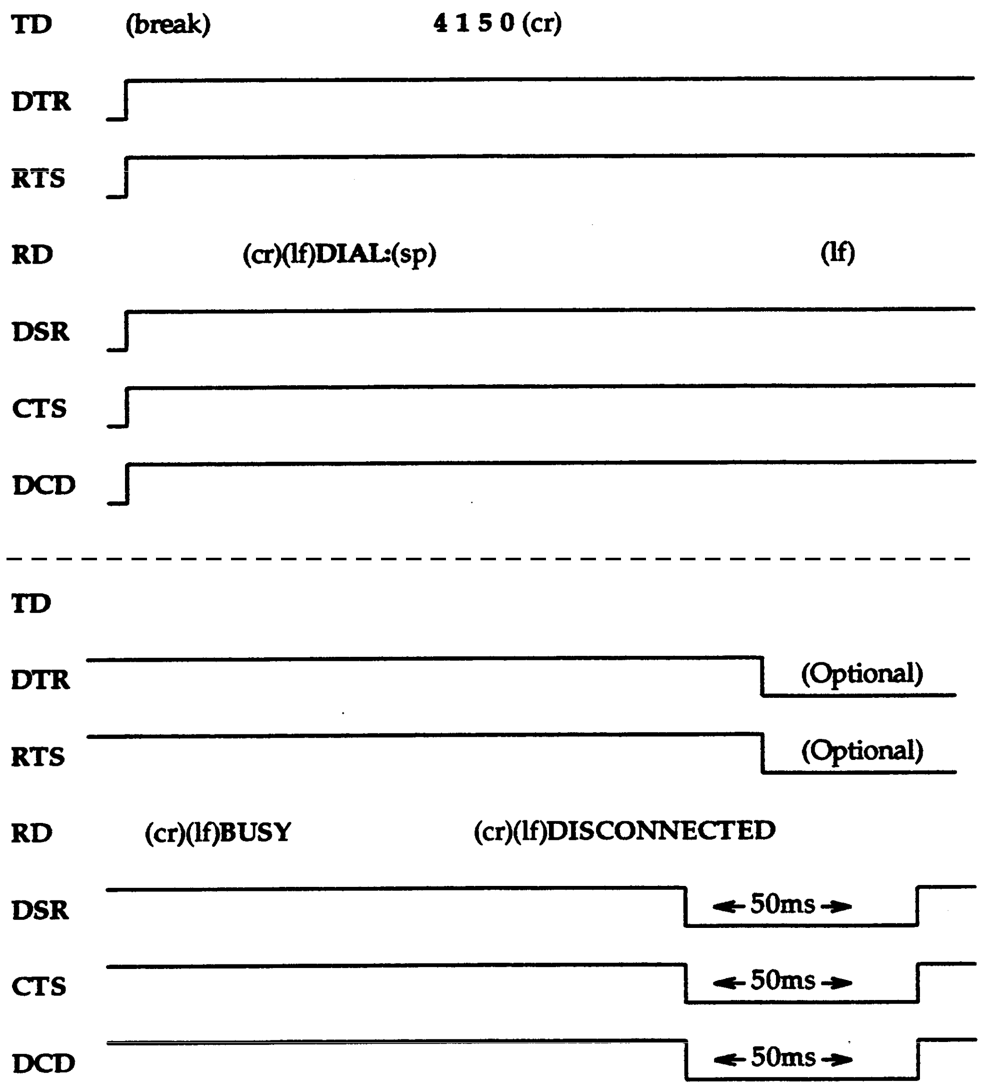
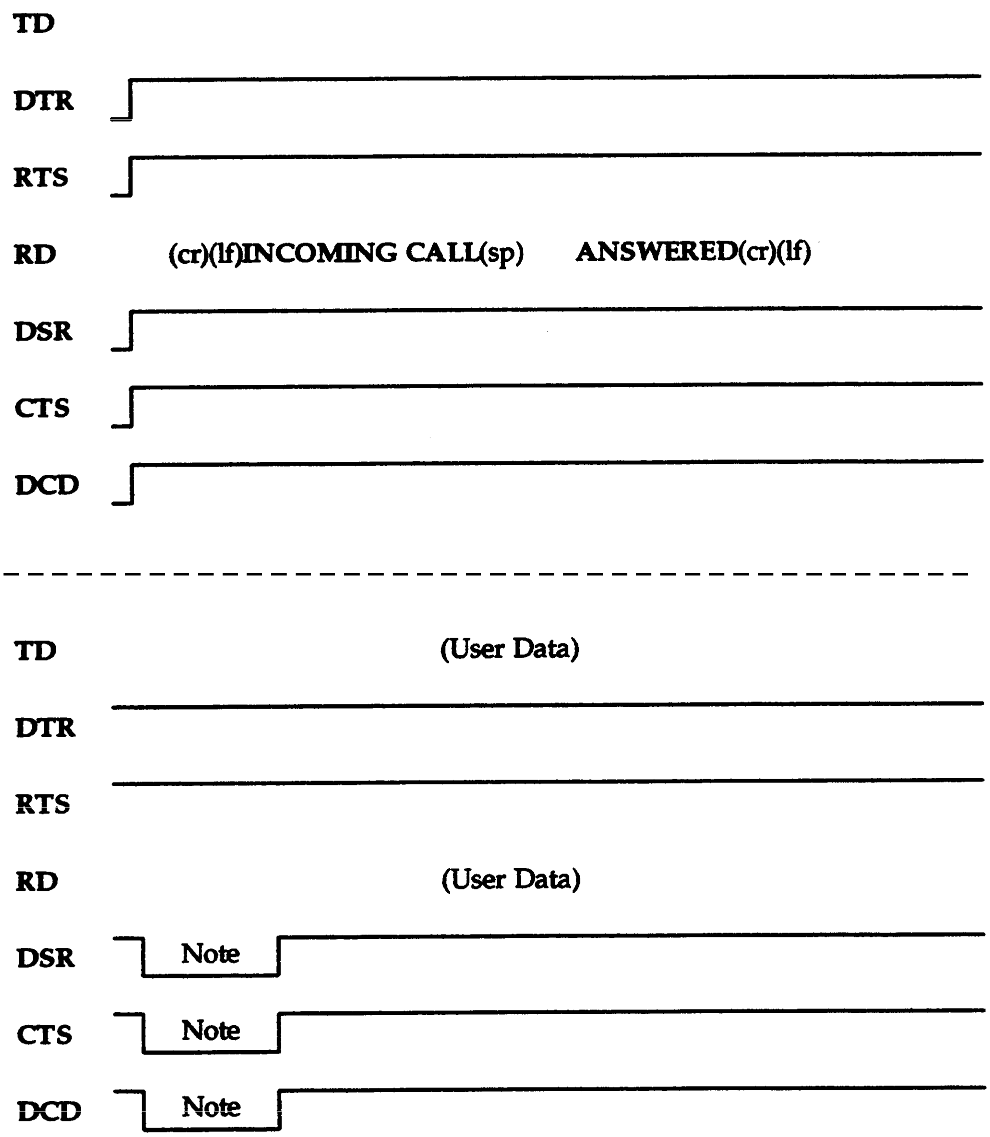


Figure D-5. Control Sequence — Station-to-Station Call, Called Extension Busy

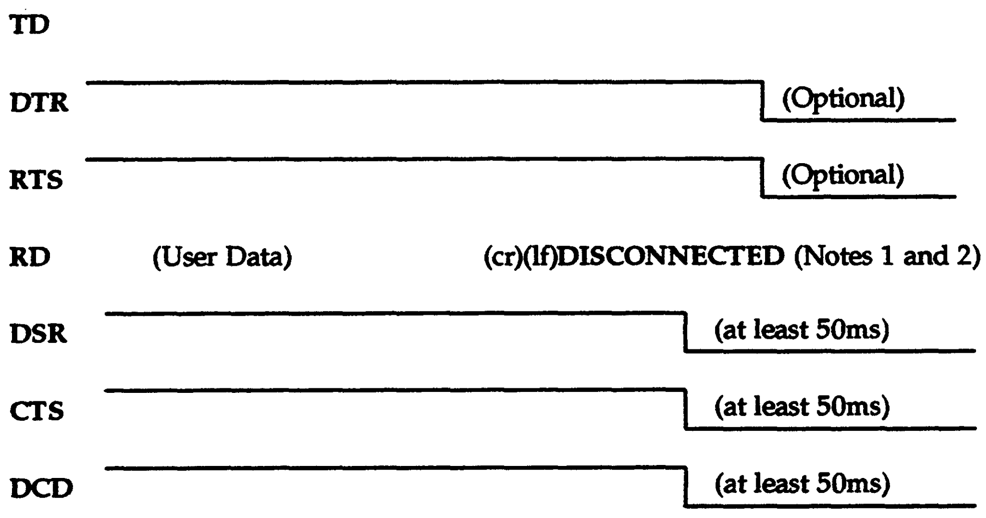
Figure D-6 shows the incoming call answer sequence at a data module that has keyboard dialing enabled and DTR is up.



Note: Data module handshake interval.

Figure D-6. Control Sequence — Incoming Call Answer

Figure D-7 shows the disconnect sequence at a data module with keyboard dialing enabled.



Note 1: A reason-for-disconnect message may follow the **DISCONNECT** before DSR drops.

Note 2: The **DISCONNECT** message is suppressed if operating in synchronous mode.

Figure D-7. Control Sequence — Disconnect With Keyboard Dialing Enabled

Notes:

Specific Signaling Formats

The B8ZS Format

The B8ZS format is recommended by the CCITT for use in ISDNs. The frame structure is based on recommendation G.733.

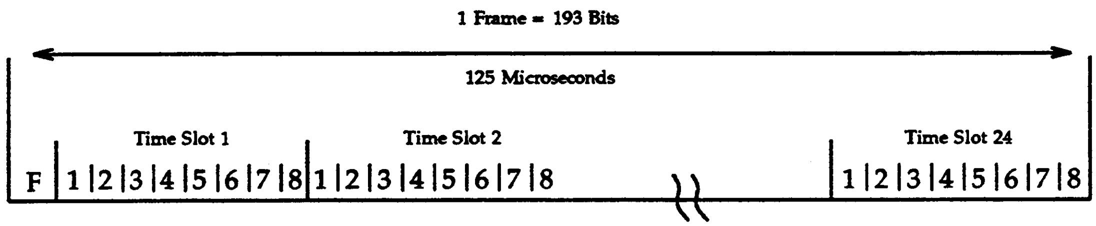


Figure D-8. The B8ZS Digital Frame Format

Each frame is 193 bits long and consists of an "F" (framing) bit followed by 24 consecutive time slots, numbered 1 to 24. Each time slot consists of 8 consecutive bits, numbed 1 to 8. The frame repetition rate is 8000 frames per second.

Time slot assignments are as follows:

Time Slot 24

Assigned to the "D" (data) channel for signaling purposes, if a D channel is present. Note that a D channel may not be required on all carriers. It may be possible for one D channel to support the signaling requirements of more than one 24 channel carrier. This is permitted by the CCITT standard. If a D channel is not used on a carrier, time slot 24 for that carrier can be used as a "B" bearer channel.

Time Slots 1 through 23

Assigned to correspondingly numbered "B" (bearer) channels.

D4 Framing

FRAME NUMBER	S-Bit	
	TERMINAL FRAMING Ft	SIGNALING FRAMING Fs
1	1	—
2	—	0
3	0	—
4	—	0
5	1	—
6	—	1
7	0	—
8	—	1
9	1	—
10	—	1
11	0	—
12	—	0

Figure D-9. D4 Framing Format

Extended Super Frame

The following is the Extended Super Frame (also called the Multiframe) format structure. Each superframe (or multiframe) is 24 frames long and is defined by the FAS (frame alignment signal) formed by every fourth "F" (framing) bit.

Multi-Frame Frame Number	F-Bits			
	Multi-Frame Bit Number	Assignments		
		FAS	R	CRC
1	0	—	m	—
2	193	—	—	e
3	386	—	m	—
4	579	0	—	—
5	772	—	m	—
6	965	—	—	e
7	1158	—	m	—
8	1351	0	—	—
9	1544	—	m	—
10	1737	—	—	e
11	1930	—	m	—
12	2123	1	—	—
13	2316	—	m	—
14	2509	—	—	e
15	2702	—	m	—
16	2895	0	—	—
17	3088	—	m	—
18	3281	—	—	e
19	3474	—	m	—
20	3667	1	—	—
21	3860	—	m	—
22	4053	—	—	e
23	4246	—	m	—
24	4439	1	—	—

FAS: Frame Alignment Signal (...001011...)
CRC: Cyclic Redundancy Check bits (e).

Figure D-10. The Extended Superframe

The "e" (error check) bits are used for error checking via a CRC (cyclical redundancy check) technique. This format is included in CCITT recommendation G.733. The "**m**" bits are reserved for further study by the CCITT (possible maintenance applications).

Notes:

Appendix E: Images, Appearances, and Extensions

Preface

This Appendix is intended to clarify the relationship between images, appearances, and extensions. Confusion has resulted from using the term "bridged appearance," because it leads the user to believe that the terminals ***sharing*** an appearance are physically bridged together. At one time, bridging was accomplished through physical (hard wire) connections, however, this is no longer true.

With modern technology, bridging is accomplished through software. No physical connection exists between bridged images or appearances. The term "shared appearance" (access point for two or more "images") is used to better describe the functionality of appearances for this application. The term "image" was introduced to distinguish between a particular "bridged appearance" (now shared appearance) and the point(s) of access to that appearance. It is hoped that, rather than compounding the confusion, this new terminology will simplify a somewhat difficult topic.

Definitions

Extensions

Extensions are unique numbers (3-, 4-, or 5-digits) by which calls are routed to a particular station. In effect, an extension is an address used for routing of a call. This number is primarily associated with a particular terminal (often the home terminal is the destination of the routed call).

Appearances

Appearances are points of access to an extension. (Appearances are also known as "occurrences.") Appearances are, in effect a further definition of the addressing function of an extension. System 85 and Generic 2 software allows as many as 12 appearances of an extension. However, a more common limit, imposed by the physical arrangement of most multiappearance voice terminals, is ten appearances per extension. As a matter of common practice, three appearances of an extension are usually assigned to a home voice terminal.

> **NOTE:** The software limit of 12 appearances can be reached using a 7205H or 7405D (equipped with the appropriate coverage module), a 7507, or a 7434D. This limit can also be reached by using more than one terminal (assigning different appearances to different terminals).

Shared Appearances

A shared appearance is an appearance that allows multiple terminals to access it (for example, an executive and his or her secretary may share the same appearance, each with an image of that particular appearance). On a multiappearance voice terminal, the point of access (or "image") to a shared appearance is called an appearance button. A shared appearance is an integral part of the Bridged Call feature which allows two points of access to the same appearance to participate in the same call at the same time.

Terminals With Shared Appearances

At one time, one "point of access" to every appearance was located on the home voice terminal. This is no longer necessary. However, for most applications it is still recommended. Failure to adhere to this practice can have an adverse effect on the functioning of the Bridged Call feature.

It should also be understood, that with the exception of an ISDN—BRI data appearance, these points of access (on the home terminal) are in no way preferred over (or superior to) the points of access on the other terminals. From the perspective of System 85 or Generic 2 software, (except for BRI data appearances), these points of access share in a "peer" relationship. Each point of access (or "image") is ***an equal*** in its ability to place and answer calls.

Images

Images are points of access to an appearance. System 85 software allows from one to 16 images of each appearance. An appearance with ***only one image*** is an unshared appearance. An appearance with ***from 2 to 16 images*** is a ***shared*** appearance.

Shared appearances are basic to the Bridged Call feature. With the exception of ISDN—BRI data calls, when an appearance is shared, one other person who shares the appearance where an active call resides can "bridge onto" the call.

> **NOTE:** An extension with ***one or more shared appearances*** is known as a ***shared extension.***

> **NOTE:** The System 85 and Generic 2 software allows for as many as 192 images of an extension (i.e., 12 appearances x 16 images per appearance).

Description of Images and Appearances

Figure E-1 shows the structure of a fictitious extension This extension, number "6234," has three appearances, and all three appearances have at least one image. Appearances 6234-1 and 6234-3 are shared appearances while appearance 6234-2 is unshared.

Notice the connections within the diagram. Each appearance's images are ***independent*** of the other appearance's images. For example, an image of appearance 6234-3 ***can only access*** appearance 6234-3, not 6234-1 or 6234-2. Likewise, when a call terminates to appearance 6234-3, visual alerting (and ringing, if assigned) will only occur for the three images of ***that*** appearance. When a call does terminate to the third appearance of Extension 6234, the call can be answered from any of the three assigned voice terminals.

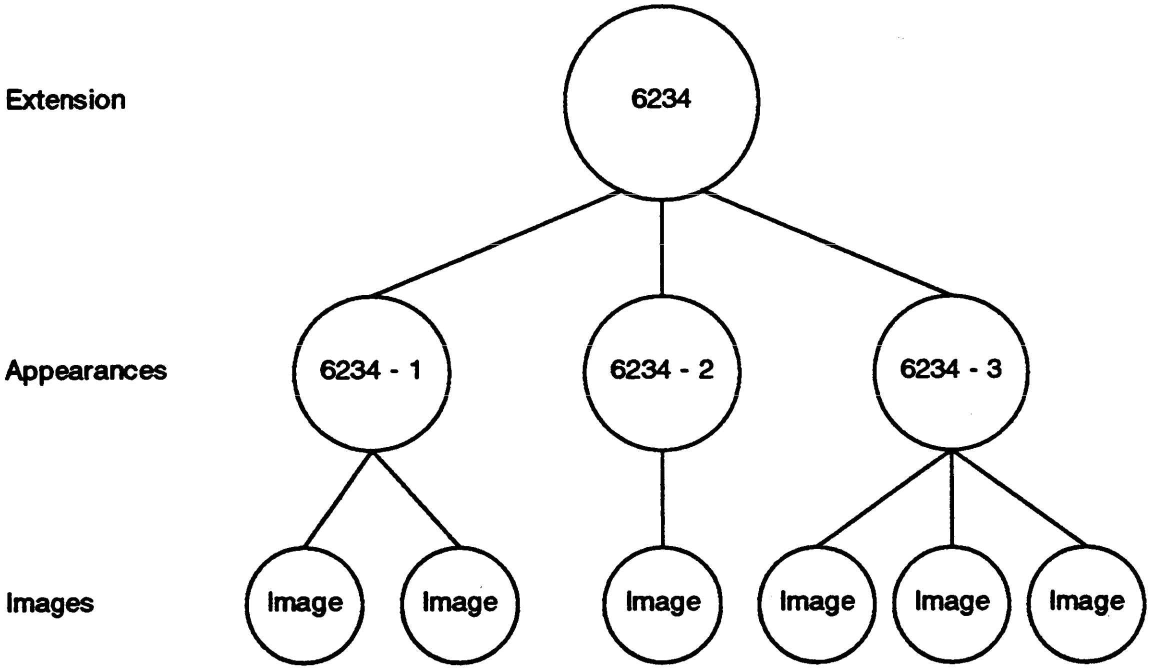


Figure E-1. Relationship of Images, Appearances, and Extensions

Figure E-2 is a simplified drawing containing eight voice terminals labeled A through H. The voice terminals labeled A through E represent multiappearance voice terminals, and F through H represent single-appearance terminals. Since the appearances on terminals F, G, and H are shared appearances, these terminals are known (and administered) as ***straight line sets.***

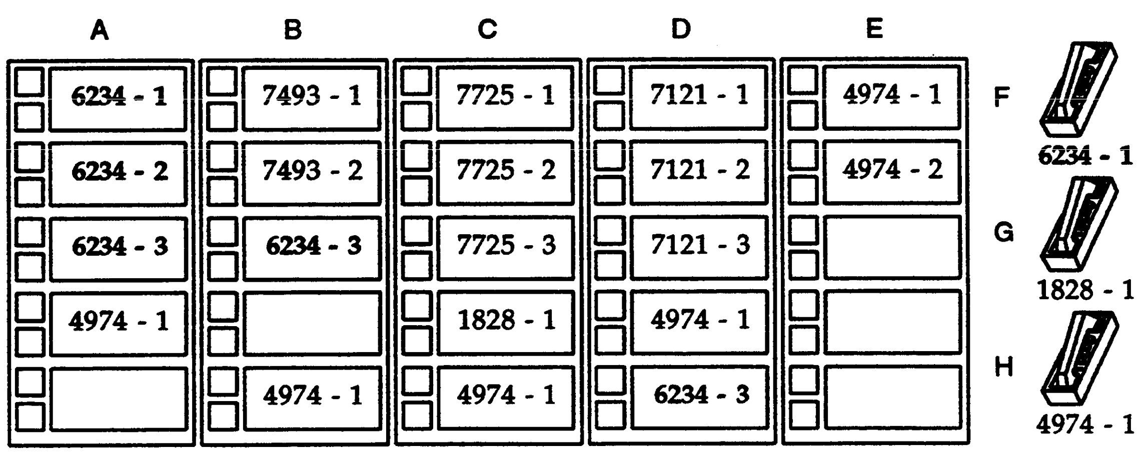


Figure E-2. Shared Extensions

Notice that Extension 6234, previously shown in Figure E-1, is also shown in Figure E-2 (in bold). For Extension 6234, voice terminal A would be the home terminals. An image of every appearance of extension 6234 is on this terminal. There are two images of the first appearance, located on terminals A and F. One image of the second appearance is located on terminal A. There are three images of the third appearance, located on terminals A, B, and D.

NOTE: The image configurations shown in Figure E-2 are ***administrable,*** but not necessarily ***functionally desireable.*** This figure is only meant to show some of the possibilities. It is left to the reader to devise functional image configurations to suit specific needs.

Another extension that is worth some discussion is extension 1828. The home terminal for this extension is terminal G. The two images of appearance 1828-1 are located on voice terminals G and C. Single-appearance voice terminal G must be administered as a straight line set in order to share the appearance with terminal C.

This arrangement is typical of an executive/secretary arrangement where the secretary performs many of the calling duties. Terminal G would be located on the executive's desk, and the secretary would handle terminal C. Terminal C is home to extension 7725 with three appearances. Using three appearances, the secretary can effectively handle most of the calling responsibilities. However, extension 1828 serves as a direct line to the executive. When this extension is unanswered, or when ringing is transferred to the secretary, the secretary can also answer these calls using the image on terminal C.

Visual Alerting

When a call terminates to an appearance of an extension, the green status lamp flashes for every image of that appearance on a multiappearance voice terminals.

Administration

Home Terminal Administration

- If "home terminal" is assigned to an image of one appearance on a voice terminal, the home terminal designation is applied to the image of every appearance of that terminal's extension number. An image can be designated as "home terminal" in Procedure 052, Word 1, Field 12. Note that for many applications, it is not necessary to designate a home terminal.

Send All Calls Group Administration

- When "Send All Calls Group" is assigned to one image of an extension on a voice terminal, this assignment is applied to every image of the same extension on the same voice terminal. "Send All Calls Group" is assigned to an extension's images residing on a specific voice terminal in Procedure 052, Word 1, Field 14.

Administrable Characteristics of Appearances

- Terminating / Originating or Originating Only

  When one of these attributes is assigned to an image of an appearance, every image of that appearance takes on the same attribute. This attribute is assigned in Procedure 052, Word 1, Field 13.

- A / D Ringing Encode

  A ringing encode is assigned for the Abbreviated and Delayed Ringing feature. When manual, automatic, or manual and automatic ringing transfer is assigned to an image of an appearance, the type of transfer chosen is applied to every image of the appearance. The type of transfer is assigned in Procedure 052, Word 2, Field 3.

Administrable Characteristics of Images

- **Ring Type**

  A type of ringing for the Abbreviated and Delayed Ringing feature (such as no ringing, ringing, delayed ringing, or abbreviated ringing) is independently specified for each image of an appearance. The ringing type is assigned in Procedure 052, Word 1, Field 11.

- **Ringing Transfer Encode**

  A ringing characteristic for the Ringing Transfer feature (such as, no ringing transfer, ring when active, or no ring when active) is independently specified for each image of an appearance. The ringing transfer encode is assigned in Procedure 052, Word 2, Field 4.

Images and the Bridged Call Feature

The software of the Bridged Call feature allows ***any two images*** of the same appearance to participate in the same voice call at the same time. However, the images of separate appearances are independent. That is, an image of one appearance (bridged or not) cannot bridge onto a voice call that is active on a *different appearance,* even though they are both appearances of the same extension.

Within the context of the Bridged Call feature, the following conditions hold true:

Any image of a particular appearance can place and / or answer a call on that appearance.

Once the call (either terminating or originating) is established, another image of the same appearance (except for ISDN—BRI data appearances) can bridge onto the call. (An image of a different appearance cannot bridge onto the call.)

A third image cannot enter an active call (while two images are active). When this is attempted, the switch returns reorder tone to the third image.

Images of ISDN—BRI Appearances

For ISDN—BRI appearances, some special rules apply. ISDN—BRI appearances can be either standard appearances (that support both voice and data calls) or data appearances (that support only data calls). Either type of appearance can have multiple images assigned. For a standard appearance with a voice call active, the general rules for the Bridged Call feature discussed above apply. However, if a standard appearance has a data call active, bridging on from another image is not allowed (reorder tone is returned).

For an ISDN—BRI data appearance, only data calls are allowed. While these appearances can be shared, actual bridging onto an active data call is not allowed (reorder tone). This is true for any image of the data appearance regardless of home terminal or data prime line designation. For further details, *see* Data Prime Line under the ISDN—BRI feature.

Appendix F: Enhanced Trunking

General

The Generic 2 switch connects to most types of trunks currently available. Trunking services that can be used with the Generic 2 include:

- Both analog and digital trunking to the local service area (Central Offices)
- Where ***direct*** or ***bypass*** access is available, both analog and digital trunking to class 4 toll offices
- ISDN—PRI (Integrated Services Digital Network-Primary Rate interface) service for both public and private networks
- Both analog and digital (including ISDN) tie trunk connections for private networks.

Enhancements

Most recent trunking enhancements were applied in System 85, Release 2, Version 4 and are continued in Generic 2. These include trunking services that improve trunk performance, trunk maintenance, and service operations. These rececent enhancements fall into three general categories:

- **ISDN trunking services**

  These services allow the System 85, R2 V4 and Generic 2 switches to participate in the emerging worldwide networking standard.

- **General trunking: Enhanced signaling and error recovery**

  These services address trunk signaling and trunk seizure failure problems.

- **Trunk Partitioning**

  Certain trunk types allow dedication to a partition or can be shared between partitions in a Tenant Services switch.

Enhancements for Generic 2

Most trunking enhancements introduced in Generic 2 are in the ISDN category. These include the following:

- ***Code Conversion:*** Required on ISDN—PRI spans for proper ISDN message handling between switches using dissimilar codesets (such as Codesets 6 and 7).
- ***D-Channel Backup:*** Used with NFAS (Non-facility Associated Signaling) to improve reliability of D-channels supporting multiple PRI spans.
- ***ISDN—FLOW Control:*** Provides protection to the switch processor from overloading due to excessive D-channel messaging activity.

- ***NFAS (Non-Facility Associated Signaling):*** Allows one (or two) D-channels to provide signaling support to multiple ISDN—PRI spans.

Many of the enhanced trunking services (both System 85, R2 V4 and Generic 2 enhancements) are interrelated. For instance, the general trunking enhancement, ***separation of trunk type and signaling*** (System 85, R2 V4), supports the ISDN ***Call-by-Call Service Selection capability.*** Also, the ***glare detection and handling*** enhancement is available only on ***true wink start, true delay dial,*** and ***ISDN trunks.*** Several general trunking services, like ***glare detection*** and ***retry,*** apply to ISDN trunks.

While many of these trunking services are most useful in large and complex networking operations, improvements in trunk services effects most switch users.

ISDN Trunking Services

The ISDN—PRI feature calls for several enhanced trunking services. Most of these services can be used for both public and private networks.

ISDN Trunk Signaling

An ISDN trunk uses a signaling protocol that is different from the traditional types of analog and digital (DS1) trunk signaling. Analog trunks use change of state (on-hook and off-hook) signals to pass information between switches. Standard digital (DS1) trunks pass the same information with "A" and "B" bit signals. ISDN trunks pass data in the form of ***message packets.*** This is called MOS (Message-Oriented Signaling). This messaging protocol provides more flexibility and increased "power" to the signals passed between switches.

Dynamic Trunk Type

‘The dynamic trunk type (120), is designed for use with ISDN trunks. Use of this trunk type is made possible by message type signaling and by the *separation of trunk type and signaling* enhancement. These functions allow the active trunk type of a trunk group with basic trunk type 120 to be specified by the networking feature (AAR, ARS, or WCR) in the pattern preferences. This capability is used with the ISDN Call-by-Call Service Selection function.

With the dynamic trunk type, the switch can change the features and services of a trunk based on requirements from the ISDN message packet or local call setup. The same trunk group can be used for a variety of purposes (for example, MEGACOM, WATS, or 800 service trunk, etc.) as the needs of the system dictate.

It is important to note that trunk type 120 ***is not*** the only trunk type than can be used for ISDN trunks, nor is it the only trunk type that can be used when implementing Call-by-Call Service Selection. Any trunk type that can use signaling type 20 (Table F-A and B) can be used as an ISDN trunk type.

> **NOTE:** On System 85, Release 2, Version 4 switches prior to release 2.0, some problems were experienced with trunk type 120 on incoming calls. These problems related to the number of address digits received and were experienced

only on incoming calls. This difficulty has been corrected and trunk type 120 functions properly for both incoming and outgoing calls on Version 4 (2.0 and later) and Generic 2 switches.

Generic 2 ISDN Trunking Enhancements

ISDN—Flow Control

ISDN flow control in Generic 2 is based on the System 85, Release 2, Version 4 enhancement ***Scanning and Recovery Mechanism for Hyperactive DS1 Trunks.*** ISDN flow control applies to ISDN D-channel messaging activity and monitors activity on both PRI and BRI D-channels.

Codeset Conversion

IEs (Information Elements) contained in Codesets 6 and 7 have changed between the System 85, R2 V4 and Generic 2 switches. These changes resulted from a significant expansion in the use of these codesets. Codeset conversion applies to DEFINITY Generic 2 switches.

The Codeset Conversion capability allows the Generic 2 switch to recognize that the switch at the distant end of a particular PRI span "speaks a different language." The DEFINITY Generic 2 switch can then provide the needed translations to avoid confusion or misinterpretation of messages passed between these switches.

Codeset conversion is introduced to support compatibility between System 85, R2 V4 and DEFINITY Generic 2 switches. It is also needed on ***direct access*** or ***bypass access*** PRI spans between a Generic 2 switch and a 4 ESS (4E11) class 4 toll office switch (the 4 ESS, 4E11 uses the same codesets used by the System 85, R2 V4). A future benefit of the Codeset Conversion capability will be to allow optimum ISDN functionality on PRI spans between Generic 2 switches and other manufacturers' switches that may not use the same IEs in codesets 6 and 7. The IEs in codesets 6 and 7 are not specified by CCITT standards. These codesets are for local network uses and terminating interface use (user-to-user information) and could vary considerably between manufacturers.

NFAS (Non-Facility Associated Signaling)

The NFAS enhancement in DEFINITY Generic 2, allows a single D-channel to provide signaling support for up to 20 ISDN—PRI spans. In effect, non-facility associated signaling uses the excess signaling capacity usually available on a D-channel to provide signaling support for additional B channels.

With NFAS, ISDN circuit boards are administered in one of two possible configurations: the standard ISDN configuration of 23 B plus D, or the communications intensified configuration of 24 B. Either the ANN35 (for traditional modules) or the TN767 (for universal modules) can be used for a 24 B configured span. When the TN767 is configured as 24 B, the packet adjunct board (TN555) is not required. For each NFAS group (up to 20 spans), there must be at least one (and at most two) spans configured as 23 B plus D.

A side effect of this enhancement is that, for an NFAS arrangement that covers 20 PRI spans, it releases up to 19 D-channels that can now be converted to B-channels for additional trunk capacity at no added cost.

D-Channel Backup

One potential drawback to the NFAS enhancement is that if the D-channel in an NFAS arrangement fails, up to 479 B-channels (trunks) could be taken out of service. The D-channel backup enhancement increases the reliability of an NFAS arrangement by providing a ***backup signaling channel.*** The relationship between NFAS and D-Channel Backup is that you must have NFAS to use D-Channel Backup; however, you are not required to use D-Channel Backup with an NFAS arrangement.

With D-Channel Backup, a primary D-channel (D1) provides signaling for an NFAS arrangement (2 or more PRI spans). A second D-channel (D2), located on a separate PRI span in the same NFAS arrangement is designated as backup for D1. If the primary (D1) should fail, the backup (D2) automatically takes over call-control and signaling for the NFAS arrangement. When this happens, the backup becomes primary (D1), and when the previous primary is returned to service it returns as the backup (D2).

General Trunking: Enhanced Signaling and Error Recovery

The trunk signaling enhancements consist of:

- The separation of trunk type and signaling, which adds flexibility to trunk administration
- Three added trunk signaling protocols: true wink start, true delay dial, and ISDN MOS. The ISDN MOS protocol has already been discussed.

Separation of Trunk Type and Signaling

Prior to System 85, Release 2, Version 4, the "trunk type" defined the signaling, features, and services available on a trunk inteface. This system provides little flexability when selecting trunk types to match feature and service requirements.

The separation of trunk type and signaling enhancement allows the trunk and signaling type to be selected independendy from the features and services requirement. This enhancement provides the flexibility needed for ***Direct*** and ***Bypass Access*** networking arrangements and for the ISDN capability ***Call-by-Call Service Selection.*** A given trunk can be used for a variety of purposes, without regard to the type of trunk signaling used to set up or maintain the connection. As an added, long range benefit, the separation of trunk type and signaling will decrease the pressure and complexity of adding new trunk types to provide for new features and services as they are developed in the future.

Limitations:

This enhancement does not allow the use of any type of signaling with any type of trunk. It does, however, significantly increase the flexibility available in assigning signaling protocols to different trunk types.

Call-by-Call Service Selection Application:

The separation of trunk type and signaling enhancement supports the ISDN Call-by-Call Service Selection capability. With Call-by-Call Service Selection, the same trunk group can be assigned to different network routing preferences. The signaling type and NSF (Network Specific Facilities) codes are assigned to the network (AAR, ARS, or WCR) preferences and these assignments are then used to build the ISDN Message when a preference is selected for call routing.

Applicable Trunk and Signaling Types:

Tables F-A and B list the trunk types to which the separation of trunk type and signaling enhancement can be applied. Table F-A applies to System 85, Release 2, Version 4 while Table F-B applies to DEFINITY Generic 2.

Table F-C lists the signaling type numbers (used in Table F-A and F-B) and their associated signaling protocols.

Signaling Compatibility

Table F-D shows the compatibility between signaling types on the DEFINITY Generic 2 switch. The first column is the frame of reference for directional compatability. Compatibility is indicated as follows:

- * Two way compatibility
- -> One way compatibility outgoing
- <- One way compatibility incoming.

TABLE F-A. Trunk Types and Signaling Characteristics for System 85, R2V4

Trunk Type and Description	Signaling					
	E&M	GS	RB	LS	ISDN	Other
All 2-Way APLT Trunks:						
12 APLT	* 9					
13 APLT	* 10					
14 APLT	* 8					
15 APLT	* 5					
Regular CO Trunks						
16 1-Way In Attendant Completing	4,21†	* 1		19	20	
17 1-Way Out DOD	4,21†	* 1		19	20	
18 1-Way Out DOD w/Party Test		* 2				
19 2-Way In Attendant Completing/DOD	4,21	* 1		19	20	
20 2-Way With Party Test		* 2				
Foreign Exchange Trunks:						
21 1-Way In Attendant Completing	4,21†	* 1		19	20	
22 1-Way Out DOD	4,21†	* 1		19	20	
23 1-Way Out DOD w/Party Test		* 2				
24 2-Way In Attendant Completing/DOD	4,21†	* 1		19	20	
25 2-Way with Party Test		* 2				
WATS Trunks						
26 1-Way In Attendant Completing	4,21†	* 1		19	20	
27 1-Way Out DOD or Toll Terminal Access for TSPS	4,21†	* 1		19	20	
28 1-Way Out DOD with Party Test		* 2				
DID Trunks:						
30 Immediate Start DID	4		‡* 3		20	
31 Wink Start DID	11		* 3		20	

* = Default Signaling.

† = Direct connections to class 4 offices (4 ESS switches) using multifrequency signaling require wink start for incoming trunk seizure. Because of operational characteristics, the 4 ESS switch will send at least one address digit. However, the System 85 will ignore this digit.

‡ = Call processing will eliminate the wink to distant end.

TABLE F-A. Trunk Types and Signaling Characteristics for System 85, R2V4 (Contd)

Trunk Type and Description	Signaling					
	E&M	GS	RB	LS	ISDN	Other
E&M Trunks:						
32 1-Way In Dial Repeating	* 4					
33 1-Way Out Automatic	* 4					
34 1-Way Dial Repeating Out	* 4	1				
35 1-Way In Automatic	* 4	1				
36 2-Way Dial Repeating Both Ways	* 4					
37 2-Way Dial Repeating In, Auto Out	* 4					
38 2-Way Auto In, Dial Repeating Out	* 4	1				
39 2-Way Auto Both Ways	* 4	1				
40 1-Way In Dial Repeating, Delay Dial	*27					
41 2-Way Dial Repeating, Wink Start In Wink Start Delay Dial Out	*26 11,21,22				20	
42 1-Way In Wink Start	*26 11,21,22				20	
43 1-Way Out Wink Start or Delay Dial	*26 11,21,22				20	
44 2-Way Dial Repeating, Delay Dial In	*27					
45 2-Way Dial Repeating, Delay Dial In Wink Start In Auto Out	*27					
46 2-Way Dial Repeating In, Delay Dial Wink Start Out	*24 12,21,22 25					
47 2-Way Dial Repeating Delay Dial In, Wink Start Delay Dial Out	*24 21,2 23				20	
Special Trunks:						
50 Remote Access 2-Way Dial Tone In, Ground Start and Dial Tone Out	4,21	* 1			20	
51 Telephone Dictation Interface						* 7
52 Recoded Announcement Interface						* 7
53 Code Call Interface						* 7
54 Loudspeaker Paging Interface						* 7
55 Touch-Tone Sender						0
57 CAS Release Link Trunk	* 13					
58 ANI Interface						* 6
62 Music On Hold Interface						0
65 Contact Interface						0
66 CAS Release Link Trunk In at Main, 1-Way In	* 14					
67 Audio Interface						0
68 UCD Delay Recorded Announcement						0

* = Default Signaling.

TABLE F-A. Trunk Types and Signaling Characteristics for System 85, R2V4 (Contd)

Trunk Type and Description	Signaling					
	E&M	GS	RB	LS	ISDN	Other
Special E&M Trunks:						
70 1-Way In Immediate Start	* 15					
71 1-Way Out Immediate Start	* 15					
72 2-Way Immediate Start	* 15					
73 1-Way In Wink Start	* 16					
74 1-Way Out Wink Start	* 16					
75 2-Way Wink Start Both Ways	* 16					
76 1-Way Delay Dial In	* 17					
77 1-Way Delay Dial Out	* 17					
78 2-Way Delay Dial Both Ways	* 17					
90 ACD First Announcement	* 7					
91 ACD Second Announcement	* 7					
92 ACD Queue of Origin Announcement	* 7					
93 Malicious Call Trace	* 7					
Data Trunks:						
100 Tone Detector						0
101 Analog Data Modem Pool	* 4					
102 Digital Data Modem Pool						* 18
103 Host Access PDM						* 18
104 Host Access TDM						* 18
105 3BAP (AP32) DCPI						* 18
106 EIA Port						* 18
107 ISN/EIA Port						* 18
108 DMI Wink In, Auto Out	* 5				20	
109 DMI Wink In, Wink Out	* 11				20	
ISDN Trunks:						
120 ISDN Dynamic					* 20	

* = Default Signaling.

TABLE F-B. Generic 2 Trunk Types and Signaling characteristics

Trunk Type and Description	Signaling					
	E&M	GS	RB	LS	ISDN	Other
All 2-Way APLT Trunks:						
12 APLT	* 9					
13 APLT	* 10					
14 APLT	* 8					
15 APLT	* 5					
Regular CO Trunks:						
16 1-Way In Attendant Completing	28,29‡†	* 1		19	20	
17 1-Way Out DOD	28,29‡†	* 1		19	20	
(18) 1-Way Out DOD w/Party Test		* (2)				
19 2-Way Attendant Completing/DOD	28,29‡	* 1		19	20	
(20) 2-Way With Party Test		* (2)				
Foreign Exchange Trunks:						
21 1-Way In Attendant Completing	28,29‡†	* 1		19	20	
22 1-Way Out DOD	28,29‡†	* 1		19	20	
(23) 1-Way Out DOD w/Party Test		* (2)				
24 2-Way Attendant Completing/DOD	28,29‡†	* 1		19	20	
(25) 2-Way with Party Test		* (2)				
WATS Trunk						
26 1-Way In Attendant Completing	28,29‡†	* 1		19	20	
27 1-Way Out DOD or Toll Terminal Access for TSPS	28,29‡†	* 1		19	20	
(28) 1-Way Out DOD with Party Test		* (2)				
29‡ Automatic In WATS (no admin)		* 1				
DID Trunks:						
30 Immediate Start DID	4		* 30‡		20	
31 Wink Start DID	11		* 3		20	

* = Default signaling.

† = Direct connections to class 4 offices (4 ESS switches) using multifrequency signaling requires wink start for incoming trunk seizures. The 4 ESS switch will send at least one address digit; however, the Generic 2 and System 85 ignore this digit.

(n) = Denotes a signaling or trunk type that is not valid on universal modules.

‡ = Denotes new or changed information (for Generic 2).

TABLE F-B. Generic 2, Trunk Types and Signaling Characteristics (Contd)

Trunk Type and Description	Signaling					
	E&M	GS	RB	LS	ISDN	Other
E&M Trunks:						
32 1-Way In Dial Repeating	* 4					
33 1-Way Out Automatic	* 31† 32,27†					
34 1-Way Dial Repeating Out	* 4	1				
35 1-Way In Automatic	* 28†	1				
36 2-Way Dial Repeating Both Ways	* 4					
37 2-Way Dial Repeating In, Auto Out	* 31† 32,27†					
38 2-Way Auto In, Dial Repeating Out	* 28†	1				
39 2-Way Auto Both Ways	* 32†	1				
40 1-Way In Dial Repeating, Delay Dial	* 8†					
41 2-Way Dial Repeating, Wink Start In Wink Start Delay Dial Out	* 26 11,21,22				20	
42 1-Way In Wink Start	* 26 11/21/22				20	
43 1-Way Out Wink Start or Delay Dial	* 26 11,21,22				20	
44 2-Way Dial Repeating, Delay Dial In	* 8†					
45 2-Way Dial Repeating, Delay Dial In Wink Start In Auto Out	* 8†					
46 2-Way Dial Repeating In, Delay Dial Wink Start Out	* 24 12,21,22, 25					
47 2-Way Dial Repeating Delay Dial In, Wink Start Delay Dial Out	* 24† 21,22,23				20	
Special Trunks:						
50 Remote Access 2-Way Dial Tone In, Ground Start and Dial Tone Out	4,21	* 1			20	
51 Telephone Dictation Interface						* 7
52 Recorded Announcement Interface						* 7
53 Code Call Interface						* 7
54 Loudspeaker Paging Interface						* 7
55 Touch-Tone Sender						0
57 CAS Release Link Trunk	* 13					
58 ANI Interface						* 6
62 Music On Hold Interface						0
65 Contact Interface						0
66 CAS Release Link Trunk In at Main, 1-Way In	* 14					
67 Audio Interface						0
68 UCD Delay Recorded Announcement						0

* = Default Signaling.

† = Denotes new or changed information (for Generic 2).

TABLE F-B. Generic 2, Trunk Types and Signaling Characteristics (Contd)

Trunk Type and Description	Signaling					
	E&M	GS	RB	LS	ISDN	Other
Special E&M Trunks:						
70 1-Way In Immediate Start	* 15					
71 1-Way Out Immediate Start	* 15					
72 2-Way Immediate Start	* 15					
73 1-Way In Wink Start	* 16					
74 1-Way Out Wink Start	* 16					
75 2-Way Wink Start Both Ways	* 16					
76 1-Way Delay Dial In	* 17					
77 1-Way Delay Dial Out	* 17					
78 2-Way Delay Dial Both Ways	* 17					
90 ACD Frist Announcement	* 7					
91 ACD Second Announcement	* 7					
92 ACD Queue of Origin Announcement	* 7					
93 Malicious Call Trace	* 7					
Data Trunks:						
100 Tone Detector						0
101 Analog Data Modem Pool						* 27†
102 Digital Data Modem Pool						* 18
103 Host Access PDM						* 18
104 Host Access TDM						* 18
105 3BAP (AP32) DCPI						* 18
106 EIA Port						* 18
107 ISN/EIA Port						* 18
108 DMI Wink In, Auto Out	* 5				20	
109 DMI Wink In, Wink Out	* 11				20	
ISDN Trunks:						
120 ISDN Dynamic					*20	

* = Default Signaling.

† = Denotes new or changed information (for Generic 2).

TABLE F-C. Trunk Signaling Type Number Definitions

Signaling Number	Definition
0	No Signaling Required
1	Ground Start
2	Ground Start with Parity Test
3	Loop/Reverse Battery, Wink Start
4	E&M Immediate Start In and Out
5	E&M Wink Start In and Immediate Start Out
6	ANI ( )
7	Auxiliary Equipment
8	E&M Immediate Start In and Out
9	E&M Immediate Start In and Wink Start/Delay Dial with Dial Tone Out
10	E&M Wink Start In and Wink Start/Delay Dial with Dial Tone Out
11	E&M Wink Start In and Wink Start/Delay Dial Out (Universal Sequence)
12	E&M Immediate Start In and Wink Start/Delay Dial Out
13	E&M Release Link Trunk Out
14	E&M Release Link Trunk In
15	E&M Main Satellite, Immediate Start
16	E&M Main Satellite, Wink Start
17	E&M Main Satellite, Delay Dial
18	"S" Channel Signaling, Host Access, GPP
	"S" Channel Signaling, Host Access, EIA
19	Loop Start
20	DMI (Digital Multiplexed Interface), ISDN (Integrated Services Digital Network), MOS (Message Oriented Signaling)
21	E&M Wink Start In and Out
22	E&M Delay Dial In and Out
23	E&M Delay Dial In, Wink Start/Delay Dial Out
24	E&M Delay Dial In, Wink/Delay Dial Out with Fail On Time-Out
25	E&M Immediate Start In, Wink/Delay Dial Out with Fail On Time-Out
26	E&M Wink Start In, Wink/Delay Dial Out with Fail On Time-Out
27	E&M Delay Dial In, Immediate Start Out with AUTOVON ***(R2V4)***
27	Analog Line Loop ***(DEFINITY Generic 2)***
	New Signaling Types for DEFINITY Generic 2 Only
28	E&M Auto in Immediate Start out
29	E&M Auto in Wink Start Out
30	Loop/Reverse Battery Immediate Start
31	E&M Immediate Start In, Auto Out
32	E&M Auto In and Out

TABLE F-D. Trunk Signaling Type Compatibility For Generic 2

Signal Type	00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16
00	*																
01		*															
02			*														
03				*													
04					*	←			*				→				
05					→					←	←	←	*				
06							*										
07								*									
08					→					←	←	←	*				
09					→	→			→	*	*	*	*				
10					→	→			→	*	*	*	*				
11					→	→			→	*	*	*	*				
12					*	*			*	→	→	→	→				
13															→		
14														←			
15																*	
16																	*
17																	
18																	
19																	
20																	
21						→				←	*	*	←				
22									→	*	←	←	←				
23					→	→			→	*	*	*	*				
24						→			→	*	*	*	←				
25					←	*			*	→	→	→					
26						→			→	*	*	*	←				
27																	
28					→								→				
29						→					→	→					
30																	
31					←	←			←								
32																	

TABLE F-D. Trunk Signaling Type Compatibility For Generic 2 (Contd)

Signal Type	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	32
00																
01																
02																
03																
04									→			←			→	
05					←		←	←	*	←			←		→	
06																
07																
08						←	←	←	*	←					→	
09					→	*	*	*	*						→	
10					*	→	*	*	*	*			←		→	
11					*	+	*	*	*	*			←		→	
12					→	→	→	→	→	→		←			→	
13																
14																
15																
16																
17	*															
18		*														
19			*													
20				*												
21					*		←	←	←	*			←			
22						*	*	*	←	←						
22					→	*	*	*	*	*					→	
24					→	*	*	*	←	*						
25					→	→	→	→		→		←				
26					*	→	*	*	←	*			←			
27											*					
28									→						*	←
29					→					→					←	←
30														*		
31												*	→			→
32												→	→		←	*

True Wink Start and Delay Dial Signaling on Network Trunks

The Universal Trunk Sequence

Prior to V4, the System 85 used the "Universal Trunk Sequence" protocol rather than a true wink start or delay dial signaling protocol on E&M trunks. The universal trunk sequence is a combination of the wink start and delay dial trunk signaling protocols that allows easy administration of trunk signaling. It is compatible with a wide range of older switching equipment and provides a "forgiving" timing algorithm. The wink signal or delay dial signal provides an integrity check for the originating switch. The start of the delay dial signal tells the originating switch that the distant switch has heard its bid for use of the trunk. The wink or the end of the delay dial signal indicates that the distant switch is ready to receive digits. As an added benefit, it also shows that the signaling path (the trunk, intervening switches, and the distant switch) is working properly.

TABLE F-E. Universal Trunk Sequence Timing

Access To Trunk	Timing
Direct Dial Access	No signal from far end: switch waits 2 to 4 seconds and begins sending digits. If signal received from far end: The switch sends digits after the far end on-hook signal is received, or waits 10 to 12 seconds before declaring seizure failure. With "Universal Time-Out Sequence" and no signal from far end: The switch waits 2 to 4 seconds before declaring seizure failure. Caller receives reorder tone.
AAR/ARS	No signal from far end: The switch waits a ***customer set period*** and sends digits. If signal received from far end: The switch sends digits after the far end on-hook signal is received, or waits 10 to 12 seconds before declaring seizure failure. With "Universal Time-Out Sequence" and no signal from far end: The switch waits a ***customer set period*** before declaring seizure failure. Caller receives reorder tone.

Universal Time-Out Sequence

Prior to the Universal Time-Out Sequence (Winkless Solution), the universal trunk sequence would wait for a set time (described in Table F-E) and then send digits. With the Universal Time-Out Sequence, the customer can set a time-out bit that controls the switch response to the absence of return signals.

With the Universal Time-Out Sequence, the customer has the option of treating the time-out as an error, or allowing the call to complete (outpulse digits). If the time-out is treated as an error (time-out bit set to 1), the sequence fails and the caller hears reorder tone. The call must be placed again.

While the universal trunk sequence works well and simplifies administration, there is one significant problem with this protocol: its inability to detect far end trunk failures either because of transmission line failure or distant switch failure. This is because it assumes that the far end is ready even when the start signal is not received. Also, it is not compatible with new services such as, ***glare detection*** and ***retry on failure of outgoing trunk seizure.***

E&M Wink Start

A true wink start signal is an off-hook to on-hook signal that lasts from 0.14 to 0.29 seconds. The end of the "wink" tells the calling switch that it may start sending digits. The beginning of the "wink" must not start earlier than 0.14 seconds after the incoming seizure and must occur within 5 seconds of seizure (for System 85 and DEFINITY Generic 2 switches). Figure F-1 shows a typical E&M Wink Start signaling sequence.

True wink start signaling enables the DEFINITY Generic 2 (or System 85) to detect far end trunk failures (wink not received). It also works with the new services ***glare detection*** and ***retry on failure of outgoing seizure.***

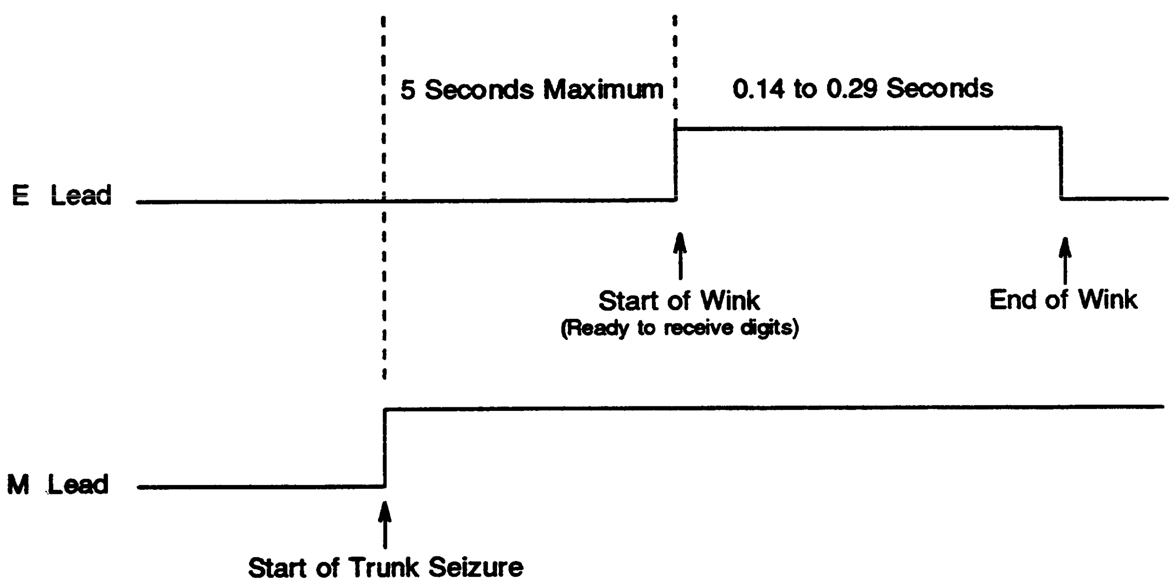


Figure F-1. E&M Wink Start Signaling Sequence

E&M Delay Dial

A delay dial signal is an off-hook signal that tells the calling switch that trunk seizure is acknowledged, followed by an on-hook signal that shows that the called switch is ready to receive digits. The off-hook must occur within 0.15 seconds of trunk seizure. The delay (off-hook pulse) must last at least 0.14 seconds. For Generic 2 (or System 85) switches, if the delay is longer than 5 seconds the call is assumed to be blocked or in glare. Figure F-2 shows a typical E&M delay dial signaling sequence.

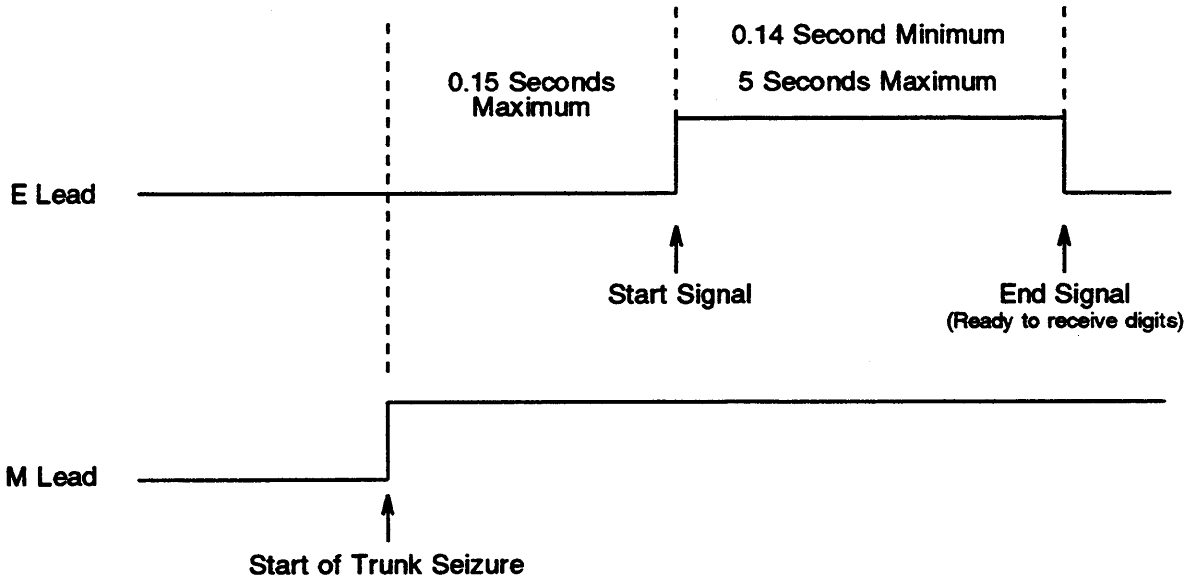


Figure F-2. E&M Delay Dial Signaling Sequence

The true delay dial signaling protocol, like the true wink start, enables the DEFINITY Generic 2 (or System 85) to detect far end trunk failures (the start signal or ready to receive signal is not received) and also works with the new ***glare detection*** and ***retry on failure of outgoing trunk seizure*** services.

Considerations

Direct Access to Delay Dial and Wink Start Trunks: For trunks selected directly (by trunk-group dial access code rather than AAR or ARS) it may be necessary for the calling party to wait for a call-progress tone for up to 5 seconds. For example:

- The system 85, R2 V4, switch will wait up to 5 seconds for the start of wink or delay dial signals. If these signals are not detected, the call could fail with reorder tone or retry could be attempted.
- If the return signals are detected, glare could be possible. The delay dial sequence takes 5 seconds before declaring glare. The wink start sequence is quicker. Wink start declares glare after 0.35 seconds. If possible, wink start is the recommended signaling sequence.

Trunk Seizure Problems

Trunk seizure problems can result from a variety of conditions including circuit failure, loss of signaling supervision, problems at the distant switch, or glare. In the past, the user was required to hang up and place the call again. The general trunking enhancements improve switch response to these problems and reduce the impact on users when trunk seizure problems occur.

The general trunking enhancements that deal with trunk seizure and trunk failure problems apply to both analog and digital trunks. There are eight specific enhancements that fall into this category:

- Glare detection and handling
- Retry on failure of outgoing trunk seizure
- Automatic detection, busy out, and service alerting, for improper trunk signaling
- Far-end removal from service on 2-way trunks when local busy-out is in effect
- Customer administerable treatment of permanent incoming seizures
- Enhanced call processing error recovery for DS1 interface failure
- Scanning and recovery mechanism for hyperactive DS1 interface
- Enhanced ground start signaling sequence.

Glare Detection and Handling

Glare occurs when both switches (both ends of a 2-way trunk) attempt to seize the same trunk at the same time. In the past, seizure failures resulting from glare were treated in the same way as seizure failures resulting from other problems. The caller received intercept tone and was required to go on-hook (hang up) and place the call again.

Both the System 85, Release 2, Version 4 and the DEFINITY Generic 2 are capable of *glare detection*. These switches can distinguish a failure because of glare from failures that result from other problems. For private network tie trunks, the switches are administered in either of two ways: one switch is given priority and is allowed to seize the trunk when glare occurs or the other switch is administered to back off of a seizure with glare and can then use the ***retry on failure*** enhancement to seize a different trunk.

The glare detection enhancement applies only to private network trunks that use ***true wink start, true delay dial,*** or ***ISDN trunk signaling.*** For non ISDN trunks, administration at both ends of the trunk must be coordinated to ensure that the opposite ends of each trunk will respond differently (one goes ahead while the other backs off).

For ISDN trunks, the glare detection and retry on failure services function as part of the ISDN ***channel negotiation*** service. Glare handling on ISDN/PRI trunks is more robust than for other types of trunks. This is because of the negotiation capabilities of ISDN. When both sides of the trunk are seized at the same time and SETUP messages cross on the D-channel, a couple of rules are used to decide which side "wins" (succeeds in making a call on that channel) and which side "loses" (backs off or moves to a different trunk).

The *first rule* is evaluated using a parameter that is indicated in the Channel ID IE, called the "preferred/exclusive" option. This option specifies that either the channel is the only one that can be used for this call, or that the call can be completed over a different channel, specified by the destination switch if the indicated channel is busy. The full benefits of using the preferred option are only reached if both switches can negotiate. The System 85, R2 V4 and DEFINITY Generic 2 switches having full negotiation capabilities, and send a preferred option indication, with one exception: In a DCS environment negotiation is impossible, so the exclusive option is used. The 4E13, in ISDN Phase 2, will also use the preferred option. In terms of glare, if both calls are exclusive, or both calls are preferred, the second rule is used to decide who wins. If one call is exclusive and the other call is preferred, the exclusive call wins.

The *second rule* is based on the translation field "Interface Type" in Procedure 262 Word 1. This field has opposite settings on either side of the PRI. When glare occurs and both calls are preferred or both are exclusive, the network side wins.

If the losing call requested preferred, it is negotiated to a different trunk in the same trunk group, otherwise it "dies" and the user hears reorder.

Retry on Failure of Outgoing Trunk Seizure

With previous systems, when a trunk seizure fails the caller receives either intercept or reorder tone. In either case, the caller must hangup and place the call again.

With this enhancement, when a trunk seizure failure occurs, the switch attempts to place the call a second time without invoking the user.

This enhancement applies only to **private network trunks** that use ***true wink start, true delay dial,*** or ***ISDN trunk signaling.***

Automatic Detection, Busy Out, and Service Alerting for Improper Trunk Signaling

In previous systems, outgoing trunk seizure failures from signaling error may or may not be identified in maintenance error logs. With this enhancement, users can administer an option to provide automatic alarms to either AT&T supplied or customer owned service centers. Another option is available that will automatically remove problem trunks from service (busy out).

Far-End Removal From Service on 2-Way Trunks When Local Busy-Out Is In Effect

With previous systems, when a 2-way trunk is maintenance busied out, the control is applied only to the local switch end of the trunk. It is still possible for the far end switch to attempt to seize the busied out trunk for an incoming call.

This enhancement allows the System 85, R2 V4 and the DEFINITY Generic 2, to set a ***reverse make busy*** on 2-way private network tie trunks. Both ends of the trunk must terminate on either a System 85, R2 V4 or Generic 2 switch administered for far-end removal from service (a "1" in Field 6, Word 3, Procedure 100, permanent incoming seizure). This allows the trunk to be removed from service at both ends without causing an alarm at the distant switch. This enhancement applies a trunk seizure on maintenance busied 2-way tie trunks. At the distant switch, this becomes a ***permanent incoming seizure.***

CAUTION: *A caller can inadvertently initiate far-end removal. If a call receives intercept tone and does not go on hook within 20-seconds for example, lays the* *receiver down while intercept tone is being received to look up a number), the distant switch may assume that far-end removal is being initiated and go into the **permanent incoming seizure** processing when it is not intended.*

Customer Administrable Treatment of Permanent Incoming Seizures

A permanent incoming seizure can be caused by several conditions. Two basic condition are addressed by this enhancement. One is when an incoming trunk seizure is connected to a tone source (such as, intercept tone) and disconnect supervision is not returned. The other is when the distant switch initiates a 2-way maintenance busy out seizure.

Two administrable options are available:

Make Idle Option: The first involves timing the incoming seizure. Incoming trunk calls that progress to a tone state are timed. Once the established interval has expired, the call is assumed to be a permanent incoming seizure. Intercept treatment is removed and an attempt is made to idle (abort) the trunk. If the trunk goes idle at both ends, it is returned to service. If the trunk does not go idle at both ends, it is passed to an alarm escalation routine. This routine attempts to release the trunk once each minute for the first 15 minutes and, if not successful, raises a warning alarm. If the trunk remains in the seized state for 2 hours it escalates to a minor alarm.

Reverse Make Busy Option: If the trunk is adminstered for the reverse make busy option (this option is administrable on a trunk group basis), the trunk is scanned to ensure the near-end is on-hook. If the near-end is idle, the trunk is placed in a state that waits for the far-end to go on-hook. No alarms are created, and the trunk returns to service when the seizure is released.

Enhanced Call Processing Error Recovery for DS1 Interface Failures

This enhancement improves the handling of various alarm conditions recorded on the DS1 interface board. Most of these alarms are DS1 specific. The alarms that this enhancement responds to are:

- Loss of signal (conductor broken)
- Loss of frame alignment (red alarm)
- Remote frame alarm, far end cannot frame (yellow alarm)
- Processor sanity
- Other, which includes:
  - Loss of multiframe alignment, near end cannot recover signaling (not applicable to ISDN).
  - Far end is remote loop back or in a maintenance state (blue alarm).

— Remote multiframe alarm, far end cannot recover signaling (not applicable to ISDN).

When any of these alarm conditions are detected on the DS1 interface status register, the remaining idle trunks on that circuit are put into a transient state to make them unavailable for use. Call processing periodically checks the interface status. If the alarms are cleared, the trunks are returned to service. The remaking channels on the interface (channels that are either in use or in the process of disconnecting) will dynamically return to service with the normal disconnect process.

This enhancement eliminates ineffective call attempts on the faulty circuits until the problem is resolved.

Scanning and Recovery Mechanism for Hyperactive DS1 Trunks

A hyperactive DS1 trunk is a DS1 interface circuit that is rapidly sending repeated on-hook off-hook transitions to call processing. With the high-speed signaling capability of the DS1 interface, this problem can quickly overload the call processing capability of a switch. This enhancement detects hyperactive DS1 circuits and takes them out of service before they cause processing problems. This applies to DS1 trunks that use bit-oriented (A and B bit) signaling. It does not apply to ISDN trunks that use message-oriented signaling.

ISDN Flow Control

The Flow Control enhancement, added in DEFINITY Generic 2, is the ISDN versions of the ***Scanning and Recovery Mechanism for Hyperactive DS1 Trunks*** introduced in System 85, R2 V4. Flow control applies to ISDN D-channel activity. ISDN flow control does not apply to B-channels. The flow control mechanism monitors ISDN facilities and keeps back of the message flow rates by type of application. Actual message flow rates are checked against standards for each facility type. These standards are maintained and updated statistically by the switch as experience factors develop.

If a particular ISDN facility significantly exceeds the standards for its type over a designated period, that facility is flagged as an application hyperactivity suspect. If suspect status continues, an additional flag is raised, the offending facility is automatically removed form service and reported on a trouble audit.

Flow control applies only to the DEFINITY Generic 2 switch and there are not plans at this time to retrofit this mechanism to System 85, R2 V4 switches. Flow control is sometimes referred to as ***Hyperactivity Management.***

Enhanced Ground Start Signaling Sequence

In earlier versions, a ground start outgoing seizure would wait indefinitely for a return tip ground signal. If this ready signal is not received, the calling party must go on-hook and try again. With System 85, R2 V4 and DEFINITY Generic 2, the ground start sequence has been changed to fail the call after a 5-second wait if the calling party does not go on-hook sooner.

NOTE: Glare detection and retry capability has not been developed for ground start trunks.

Enhancement Application by Signaling Type

The specific enhancements that apply to a particular trunk depend on the signaling protocol used.

Standard Trunk Signaling Protocols

The following trunk signaling protocols are affected by one or more of the general trunking enhancements available with System 85, R2 V4.

E&M (Ear and Mouth) Wink Start and Delay Dial Trunks

Signaling, retry, automatic maintenance busy out, and permanent seizure enhancements apply to all types of incoming trunks. Glare detection and retry applies to networking trunks only (trunk types 41, 42, 43, 46, and 47) for System 85, R2 V4 and Generic 2.

Ground Start

Outgoing seizure signaling sequence is enhanced to provide an automatic timeout if the distant end does not return a ready signal.

Loop/Reverse Battery

This signaling type is used for 1-way incoming (DID) service in DEFINITY Generic 2 (or System 85). The permanent incoming seizure enhancement applies to this signaling type.

Digital (DS1) Trunks

Digital trunks use the same signaling protocols used by analog trunks plus the new ISDN signaling protocols. Where an analog signaling protocol is used, the same enhancements apply to both analog and digital trunks. Also, the specific DS1 enhancements apply to digital trunks, and where trunks are used for ISDN interface the specific ISDN enhancements apply.

Partitioned Trunk Types

The following trunk types can be dedicated to or shared between partitions in a Tenant Services switch:

- 12 - 50
- 70 - 78 (Tie-trunks)
- 103 - 109 (PDM, TDM, AP 32 DCPI, EIA, ISN, and DMI trunks)
- 120 ISDN — Dynamic Trunk Type.

Appendix G: Integrated Services Digital Network

Overview

ISDN with System 85 and DEFINITY Generic 2

ISDN was first introduced on System 85 with Release 2, Version 4. At that time, only ISDN—PRI (Primary Rate Interface) was available.

With DEFINITY Generic 2, the range of ISDN features and services has been expanded as follows:

- **Bearer Capability:** A new feature has been added which expands and enhances a capability that was present as a subset of *line class of service* in System 85, Release 2, Version 4. The Bearer Capability feature is discussed in detail in Section 23.
- **ISDN—BRI (Basic Rate Interface):** A new feature for DEFINITY Generic 2 that provides ISDN connectivity and services at the terminal level. The ISDN—BRI feature is described in detail beginning in section 64.
- **ISDN—PRI (Primary Rate Interface):** The PRI feature was introduced in System 85, Release 2, Version 4. It has been enhanced in DEFINITY Generic 2 as follows:
  - ***Flow Control*** has been added to reduce potential overloading of the switch processor by applications hyperactivity. Flow control applies to messaging activity for both the PRI and BRI features.
  - ***IEs (Information Elements) and Codesets,*** used in ISDN messaging, have been expanded and some of the earlier IEs have been moved (from codeset 7 to codeset 6).
  - ***Codeset Conversion*** has been implemented. This capability allows the DEFINITY Generic 2 switch to communicated effectively with the earlier PRI switches despite the expansion and repositioning of IEs that has taken place. In addition, this capability allows the DEFINITY Generic 2 and later switches to communicate over PRI spans with other manufacturers' switches, even though those switches may not conform to the same codeset structure.
  - ***NFAS (Non-Facility Associated Signaling)*** has been added. NFAS allows a D-channel on one PRI span to provide signaling support for B-channels on different PRI spans. This permits more efficient use of channel resources and also allows signaling for remote channels.

— ***D-channel Backup*** is added in association with the NFAS enhancement. D-channel backup allows a second D-channel to act as an alternate or backup signaling channel for a group of B-channels supported by a primary NFAS D-channel. The use of a backup D-channel is not required by the NFAS enhancement; however, it significantly improves the reliability of NFAS arrangements.

The ISDN—PRI feature was introduced in System 85, Release 2, Version 4. It has been expanded and enhanced for DEFINITY Generic 2. ISDN—PRI is discussed in detail in Section 66.

The ISDN Concept

The ISDN concept envisions a world wide standard for ***integrated voice/data networks.*** Figure G-1 shows the telecommunications networking services that will be available from fully developed ISDNs.

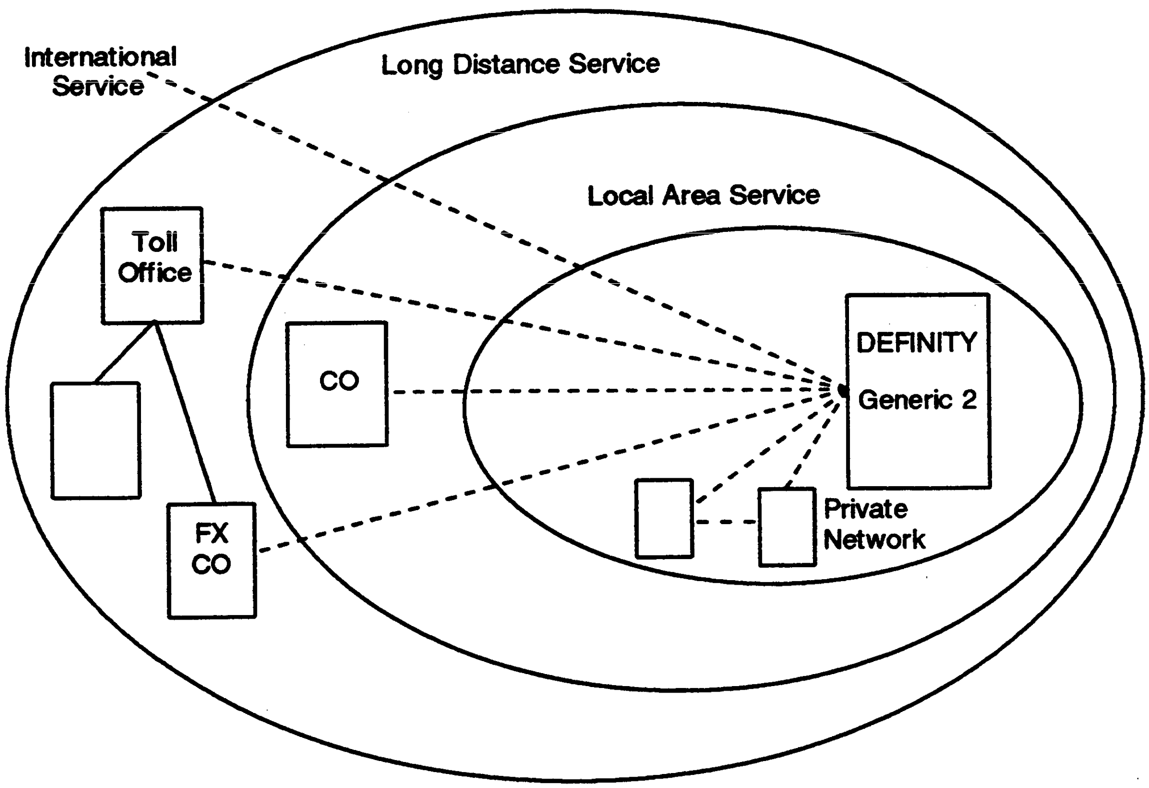


Figure G-1. The ISDN Service Concept

The CCITT (International Telegraph and Telephone Consultive Committee)

International standards for ISDN are set by the CCITT. The ISDN concept was first proposed to the CCITT in 1979 and has been evolving ever since. As an evolving system, the ISDN standards are not fully defined at all levels. Current standards are contained in CCITT recommendations in the I.420 series, I.430 series, Q.920 series, and the Q.930 series.

Terms:

Bearer Capability

The Bearer Capability concept was first introduced in System 85, Release 2, Version 4. It refers to the capability of a facility (such as a trunk and especially an ISDN trunk) to support specific types of traffic. Bearer capability applies to all calls but is of primary significance for data calling. It is used by the AAR, ARS, and WCR features in pattern and preference selection for call routing. The bearer capability concept has been expanded considerably between System 85, Release 2, Version 4 and DEFINITY Generic 2. In System 85, Release 2, Version 4, bearer capability was used only by the network routing features. In DEFINITY Generic 2, it is also used within the switch to make call processing decisions such as call support resource requirements (for example, modem pooling resource requirements) and call offering options.

Bearer Capability Class

In System 85, Release 2, Version 4 there are five bearer capability classes (or codes). These are assigned to trunk groups and specifically identify the type(s) of traffic that trunk group can support.

BCCOS (Bearer Capability Class of Service)

In DEFINITY Generic 2, the five bearer capability classes are replaced by 256 BCCOSs. Nine of these BCCOSs are predefined for use as ***defaults.***

The uses for bearer capability information have also been expanded. In DEFINITY Generic 2, the BCCOS is used for network routing as were the bearer capability classes in System 85, Release 2, Version 4; however, this application has been further refined to provide a greater degree of selectivity. Additionally, BCCOS is also used to assist in switch call processing. The BCCOS provides the switch and BRI end points with information needed to identify and support requirements, such as baud rate and modem pooling, and the specific end point to which a call should be offered (data terminal vs. voice terminal).

Table G-A compares the default BCCOSs for the DEFINITY Generic 2 with the System 85, Release 2, Version 4 bearer capability classes. Bearer capability is discussed in greater detail under the Bearer Capability feature.

TABLE G-A. Default Bearer Capability Classes of Service

Default BCCOS DEFINITY G2	BCC System 85 R2V4	Type of Call Supported
0	(0)	Voice only*
1	2	Mode 2 Data
2	—	BRI Voice / Data
3	—	Unknown Digital
4	(0)	Unknown Analog
5	(0)	Voice Grade Data*
6	4	Mode 0 Data
7	1	Mode 1 Data
8	3	Mode 3 Data

* Bearer services available on System 85, Release 2, Version 4 but in a single BCC.

Channel

In ISDN terms, a channel is a discrete communications link. A given channel may be physically separate and distinct from other channels or it can be (and usually is) virtual (logically separated from other channels on the same physical link).

ISDN uses two general types of channels:

- ***B (Bearer) Channels***

  B-Channels are the communications links in an ISDN. They provide 64 Kbps digital communications service and are used with both the BRI (Basic Rate Interface) and PRI (Primary Rate Interface).

- ***D (Data) Channels***

  D-channels are the signaling links in an ISDN. D-channels carry call control and call related information to the interfaces. D-channels are used with both the BRI and PRI, however, them are differences between the D-channels that support these separate interfaces.

  — With the BRI, the D-channel operates at 16 Kbps and supports 2 B-channels. In some cases, the D-channel in a BRI application is also used for part time packet switching applications; however, this use is not supported on the DEFINITY Generic 2 switch.

  — With the PRI, the D-channel operates at 64 Kbps and normally supports 23 B-channels

  — With PRI using NFAS (Non-Facility Associated Signaling), the D-channel still operates at 64 Kbps, but supports up to 478 B-channels.

Channel Negotiation

Channel negotiation is an ISDN capability that allows the member not in control to influence the specific facilities (channel) used for a connection that is in the process of being set up. Channel negotiation works differently for the two types of ISDN interfaces:

ISDN—PRI: With ISDN—PRI, channel negotiation provides the called (terminating) switch with a say in selecting the B (Bearer) channel (trunk) to be used for an incoming call. If the B-channel originally selected by the calling switch is not acceptable to the called switch, the called switch can request a change in the channel and the call can still be completed. Without this capability, the called switch has no alternative but to reject the call. Channel negotiation is not used for DCS calls or Trunk Verification calls.

Glare Handling: Channel negotiation can be use when glare occurs on an ISDN trunk. Glare occurs when the two switches at opposite ends of a trunk group both attempt to seize the same trunk at the same time. Because of the channel negotiation capability, glare handling on ISDN—PRI trunks is more robust than for other types of trunks. This is assuming that both switches are capable of negotiating (not all switches that support ISDN—PRI are capable of channel negotiations). For a detailed description of how Glare Handling through channel negotiation works for ISDN—PRI trunks, see Appendix F: Enhanced Trunking.

ISDN—BRI: With ISDN—BRI, channel negotiation is extended to calls between the switch and a BRI terminal. In this application, channel negotiation occurs between the switch and the BRI terminal. Two separate scenarios exist for BRI channel negotiation.

When a call is ***originated by the BRI terminal,*** either a preferred or exclusive channel is indicated.

- **Preferred Channel Option**

  If the preferred B-channel is available, it will be used for the call; otherwise the other B-channel will be used (if available).

- **Exclusive Channel Option**

  With the exclusive channel option, if the requested B-channel is available, it will be used; otherwise the call is denied.

Channel negotiation between the BRI and the switch is performed automatically by resident firmware. This is not a user administrable function.

Channel negotiation is supported on System 85 for the ISDN—PRI feature in both System 85, Release 2, Version 4 and DEFINITY Generic 2. Channel negotiation for the ISDN—BRI feature is available on DEFINITY Generic 2 switches.

Call Reference Value

A number between 0 and 32767 that is used to identify a specific ISDN call. The call reference value associates the D-channel messaging with the actual B-channel call.

Codepoint

A specific IE (Information Element) within a specific codeset.

Codeset

One of eight possible groups of IEs (Information Elements) used to form ISDN D-channel messages. Each codeset can consist of up to 133 code points or IEs. Codesets and IEs are discussed in more detail under the "Message-Oriented Signaling" section of this appendix

D-Channel Backup

D-channel backup is a recent development (DEFINITY Generic 2) that uses two D-channels (one a primary and the other standby) to control the same set of B-channels. This arrangement is used with non-facility associated signaling to enhance reliability through redundancy.

IE (Information Element)

An information element is part of a standard ISDN message. IEs provide specific elements of information. For example, a specific IE is used to convey ***calling party name*** information from the origination point to the termination point. Another IE (the Bearer Capability IE) identifies call type (voice, data mode, etc.), data rate, and other call related information.

Interface

An interface is the means by which two ***independent systems*** come together and communicate with each other. In reference to ISDN, two ***systems*** would be the network and its subscribers.

Interfaces are both physical (the pin and wiring arrangement) and protocol arrangements (the set of rules that control what signals are carried on which pin and wire and how these signals are structured).

The CCITT recommendations call for two types of interface:

- **BRI (Back Rate Interface)**

  This is a station level interface that allows individual subscribers to connect to the ISDN. With DEFINITY Generic 2, an ISDN—BRI station level interface is also available within the switch network. This provides ISDN end-to-end connectivity down to the station level (not available in System 85, Release 2, Version 4).

- **PRI (Primary Rate Interface)**

  This is a switch level interface. The PRI allows a switch to provide a common or shared interface to the ISDN for its separate stations. This type of interface can also be used by a processor (see the DMI feature). The physical part of this interface is the T1 configuration described in the DS1 Interface feature.

These interfaces are described in greater detail in separate chapters of their own.

ISDN (Integrated Services Digital Network)

The CCITT defines an ISDN in the early part of the "I." series recommendations. Two significant elements of that definition provide the following:

- An ISDN is a network, evolving from integrated digital networks. An ISDN provides end-to-end digital connectivity to support a wide range of services, including both voice and data communications. Users have access to an ISDN through a limited set of standard, multipurpose user-network interfaces.

- An ISDN is recognized by the service characteristics available through user-network interfaces, rather than by its architecture, configuration, or technology. This concept plays a key role in permitting user and network technologies and configurations to evolve separately.

Interworking

Interworking is an important concept in an ISDN, particularly as it relates to the PRI. In general, interworking is the process by which the ISDN side of the interface communicates with the Non-ISDN side of the interface. For example, with the System 85 switch the internal protocol used is DCP (Digital Communication Protocol). Interworking allows information and call control signals to be passed back and forth between the ISDN message based software and the stimulus based DCP software on the System 85 switch. Interworking applies equally to both the BRI and PRI. The interworking function is discussed in greater detail in the ISDN—PRI chapter.

Octet

This is the international (CCITT) term for an 8-bit digital word. It is generally equivalent to the term "byte" which is more commonly used in North America.

Background

The ISO (International Standards Organization) Model

ISDN uses a layered protocol that conforms to layers 1, 2, and 3 of the ISO OSI (International Standards Organization Reference Model for Open Systems Interface) shown in Figure G-2.

This reference model is frequently called the ISO Model. It has been used as the standard for packet switching networks and is the standard for the international packet switching protocol X.25. The ISO model is intended to be the standard for public networks around the world. The intent of both the ISO OSI model and ISDN is to provide an open system digital network arrangement that is independent of any particular vendor.

The Open Systems Concept

The term ***Open System*** means a system that is open or available to anyone. The standards used in such a system are public domain, and anyone who wants to can use them. In this way, any vendor is free to produce equipment or develop systems that will work with the open system. The opposite would be a ***proprietary system*** that can be used only by the developer or under license (with the developers permission).

While some services and features (for example, display services) are inherently hardware related, general access to a public ISDN and its network capabilities and services should be independent of the specific hardware used. That is, network capabilities and services should be available to all users without regard to who manufactured their equipment.

Also, a user with equipment made by one manufacturer should be able to communicate, without difficulty or special permission, with another user whose equipment was made by a different manufacturer. This goal however is by no means assured. The ISDN standard deals with only the first three layers of the ISO Model. There can be significant differences in protocol used in the higher layers.

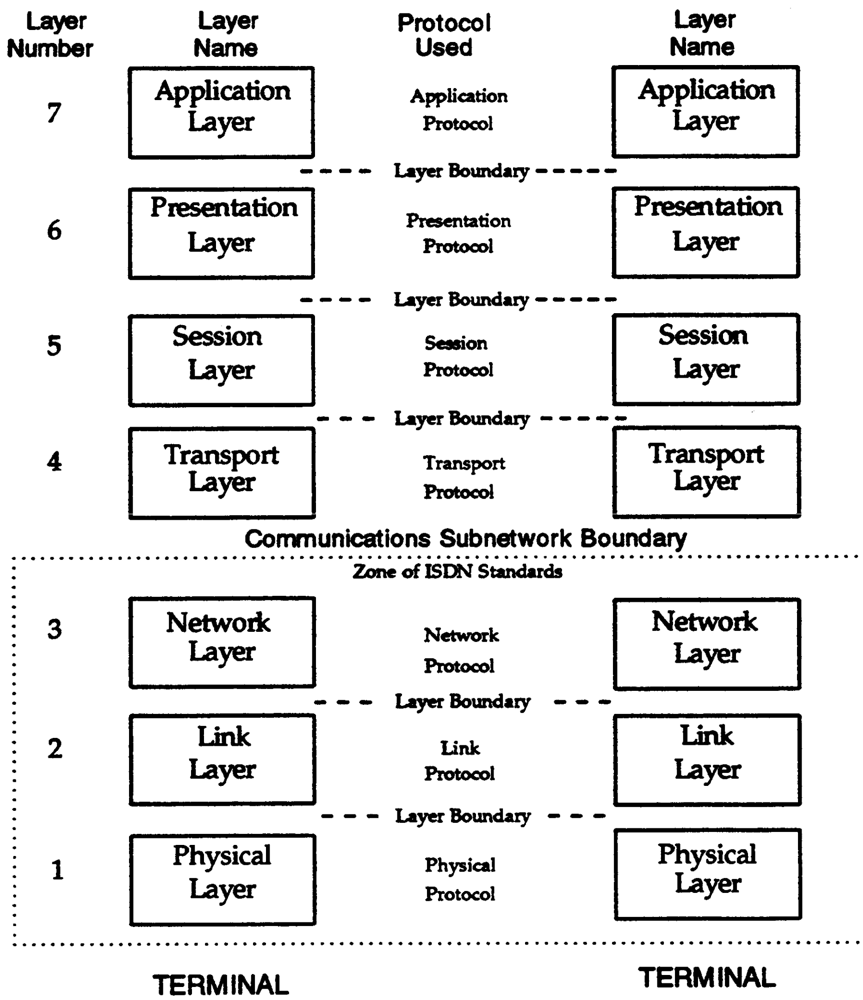


Figure G-2. The ISO Reference Model

The Layered Protocol Concept

In a layered protocol, each separate layer is independently responsible for specific functions and tasks. Each lower layer supports the layers above it, and each successive layer is independent of how the layers below it perform their functions.

For example, the Physical Layer (layer 1 in Figure G-2) provides physical connectivity support to the layers above it. The Session Layer (layer 5 in Figure G-2) performs the functions of setting up and monitoring the communications session between the communicating terminals. The Session Layer at each terminal then passes action to the Presentation Layers which in turn pass action to the Applications Layers at the different terminals.

Layers 4 through 7 (the Transport Layer, Session Layer, Presentation Layer, and the Applications Layer) communicate with the like layer at each terminal, without concern for how the layers below them perform their functions. Each layer "does its own thing" under the assumption that all the supporting (lower) layers are doing their jobs. The higher layers do not enter the picture until the lower layers have done their jobs. The upper layers (above the ***Communications Subnetwork Boundary*** are independent from lower level software within the network (levels 1, 2, and 3) and correspond only to the matching layers at the sending and receiving termination points. The termination point may be a switch (PRI) or may be a terminal-associated interface unit (BRI).

Network Features and Services

The ISDN standards provide for a limited set of ***network*** features and services. Note that these are Network Features and Services rather than switch features and services. As such, they may differ in some respects from similar features and services provided by the switch.

Where these ISDN capabilities and switch features and services match (for example, Calling Number and Display Voice Terminal), these matching features and services can be used on interswitch calls rather than being limited to local (or DCS) calls. Table G-B lists some ISDN features and services that apply to DEFINITY Generic 2, and the implementing switch features.

TABLE G-B. ISDN Capabilities and Services in DEFINITY Generic 2

CCITT Recommended Service Capability	Implementing Service or Feature	Level of Current Implementation
Call-by-Call Service Selection	ISDN—PRI with AAR, ARS, WCR, and Generalized Routing	Fully Implemented
Channel Negotiation	ISDN—BRI and PRI	Fully Implemented
Calling and Connected Party Number and Name Display	ISDN—BRI and PRI, and Display—Voice Terminal	Fully Implemented
ETN Compatibility	ISDN—PRI and AAR, ARS, WCR, with Generalized Routing	Fully Implemented
End-to-End ISDN Connectivity	ISDN—BRI and DMI—MOS	Fully Implemented
Tandemed User-to-User Information	ISDN—PRI	Partially Implemented
Maintenance Services for ISDN Facilities	ISDN—BRI and PRI	Fully Implemented

Keep in mind that on any ISDN call, the availability of a particular feature or service depends on both ends of the connection, and all intervening nodes and links, supporting that capability or service. For example, with ISDN—PRI the switch can support Calling and Connected Party Name and Number Display. This is accomplished via interworking with the Display — Voice Terminal feature for DCP station and via direct (end-to-end

ISDN connectivity) for ISDN—BRI stations. However, if the switch (or station) at the far end of the connection does not provide the IEs (Information Elements) needed for these figures, the information cannot be displayed on voice terminals.

ISDN capabilities and services supported by the System 85 or DEFINITY Generic 2, as well as switch features that interact with ISDN features, are identified under the appropriate switch feature discussions.

Call-by-Call Service Selection

The ISDN capability, Call-by-Call Service Selection, is of particular significance in utilizing the capabilities of ISDN connections. This capability allows a single (direct or bypass access) ISDN trunk group to be used for a variety of network features or services on a call-by-call basis. This can result in a much more efficient use of these ISDN trunking facilities.

Call-by-Call Service Selection works with the network routing features AAR, ARS, and WCR. A single ISDN trunk group is assigned to several different network routing patterns and preferences. Part of the preference definition is an NSF value. For System 85 and Generic 2.1 switches, this is done in Procedure 309, Word 5 or 321, Word 5. For Generic 2.2 switches, the NSF value is assigned in Procedure 322, Word 1. This NSF value is used to create an NSF IE in the call setup message that is sent to the NSO (Network Service Office). This NSF IE identifies the ***Network Nodal Service*** be used for the associated call.

NSF (Network Specific Facilities)

In System 85, Release 2, Version 4, the NSF Values are fixed (their meanings are in a memory table and cannot be changed through administration). In DEFINITY Generic 2, NSF Values are administrable and must be established through administration. The Nodal Features and Services that can be specified with an NSF Value and the fixed NSF Values that are used in System 85, Release 2, Version 4 are given in the chapter on ISDN—PRI.

Administrable Network-Specific Facilities Values

The ability to administer the NSF values gives the switch administrator considerable flexibility and significant responsibilities. Choices of features and services are, of course, limited to those offered by the serving network. The major value here is that the administrator can define new NSF values to take advantage of new and improved network features and services as they become available. Also, interexchange carriers, other than AT&T, may use different NSF values when they offer the Call-by-Call Service Selection feature. The ability to administer these values independently, will allow the user to access these features and services through other interexchange carriers.

One problem that the switch administrator will face is knowing how to define an NSF value to cause it to generate the desired NSF IE. The NSF value itself is not passed from one switch to another, and if it were would mean nothing. It is merely used within the sending switch to generate an NSF IE which is then passed to the NSO for evaluation. The elements coded in Procedure 279, Word 1 produce the hard information contained in the NSF IE associated with each NSF value.

Nodal Features and Services

The Nodal Features are a subset of the features and services available on a call-by-call basis from the AT&T Communications System Network.

Features vs Services

In network terminology, a ***feature*** provides a specific capability. The capability to pass the CPN (Calling Party Number) or SID (Station Identification) to the call destination (either switch or terminal) is considered a feature.

This is as opposed to a network ***service*** which provides a group of capabilities. The capabilities provided by WATS (Wide Area Telecommunications Service) are considered a service.

As a matter of practical resolution, one is best advised to "look it up" to determine if a particular "thing" is a feature or service. Table 63-C, in the **ISDN—PRI** chapter, lists the currently available Network Features and Services as well as the NSF Values that apply to System 85, Release 2, Version 4 switches.

Binary vs Parameterized

Another concept in dealing with NSFs that is not immediately obvious is the distinction between *Binary* and *Parameterized* features and services. The term binary is the same word used in mathematics and means the same thing. It refers to something that can have only one of two states. In mathematics, a binary number is one that contains digits that are either "0" or "1." There are no other possibilities. In NSF, a binary feature is one that can be in only one of two possible states; that is, it can either be "on" or "off." There is no other possibility.

A parameterized feature or service, on the other hand, has a range of possibilities (or parameters). The basic example is WATS which can be provided on the basis of "Bands" which are then the parameters for WATS.

The following examples show how NSF values are assigned and associated with a specific network feature or service.

Nodal Services

SDN (Software Defined Network)

An SDN is a *virtual private network.* This is a service that is not currently tariffed by AT&T Communications. SDN uses common carrier facilities to operate a functioning private network. Defining the NSF value for this service is much the same as for the feature in the previous examples.

```
                 ENAHNCED MODE - PROCEDURE: 279, WORD: 1
                        NETWORK-SPECIFIC  FACILITY

 1. ISDN Network Service Value:  nnn

ISDN  NETWORK  DEFINITION
   2. Parameterized - Binary: 1
   3.      Feature - Service: 1
   4. Facility Coding Value: 1

PARAMETERS
   5. Parameter 1: --
   6. Parameter 2: --
   7. Parameter 3: --
   8. Parameter 4: --
   9. Parameter 5: --
  10. Parameter 6: --
  11. Parameter 7: --



Connected to CC0 ON-LINE  MAJOR   MINOR  RUN TAPE  BUSY OUT  IN USE  WAIT

enter  command:
                    F3 DATA           F5 HELP  F6 FIELD  F7 INPUT  F8 CMDS
```

- Field 1 contains the NSF value that is to be assigned to this service. Any number from 0 through 511, not otherwise assigned, can be used.
- In field 2 the appropriate encode is "1" for binary. At the present time, all network nodal services, except WATS are binary.
- In field 3 the appropriate encode is "1" for service.

  The entries in field 4, Facility Coding Value, are not as complicated as they appear. In fields 2 and 3, the type (service) and form (binary) have already been established. Field 4 identifies the network service represented. The ***Administration Manual: Procedures*** (555-105-506) lists the available encodes by type and form. For SDN, the encode is "1."
- Fields 5 through 11 do not apply.

WATS (Wide Area Telecommunications Service)

WATS has been available from the AT&T Communications Network for a long time. It is not currently tariffed for ISDN. When tariffed for ISDN, the NSF will provide the ability to obtain WATS over shared trunks on a call-by-call basis.

```
                 ENHANCED MODE - PROCEDURE: 279, WORD 1
                        NETWORK-SPECIFIC FACILITY

 1. ISDN Network Service Value: nnn

ISDN NETWORK DEFINITION
   2. Parameterized - Binary: 0
   3.      Feature - Service: 1
   4. Facility Coding Value: 1

PARAMETERS
   5. Parameter 1:  2
   6. Parameter 2:  5
   7. Parameter 3: --
   8. Parameter 4: --
   9. Parameter 5: --
  10. Parameter 6: --
  11. Parameter 7: --


Connected to CC0 ON-LINE   MAJOR   MINOR   RUN TAPE   BUSY OUT   IN USE   WAIT

enter command:
                    F3 DATA            F5 HELP   F6 FIELD   F7 INPUT   F8 CMDS
```

WATS is the only parameterized service currently available. Defining this NSF value for WATS is similar to other features and services that can be specified by an NSF value.

- Field 1 contains the NSF value that is to reassigned to this service. Any number from 0 through 511, not otherwise assigned, can be used.
- In field 2 the appropriate encode is "0" for parameterized. Currently Out-WATS (band specific) is the only parameterized service.
- In field 3 the appropriate encode is "1" for service.
- The entries in field 4, *Facility Coding Value,* are not as complicated as they appear. In fields 2 and 3, the type (service) and form (parameterized) have already been established. Field 4 identifies which service is represented. The ***Administration Manual: Procedures*** (555-105-506) lists the available encodes by type and form. For Out-WATS, the encode is "1."
- Fields 5 through 11 apply to a parameterized service. The parameters are leading justified. That is, start entering parameter values in field 5 and continue until all parameter values have been entered. WATS Band 25 is specified by entering "2" in field 5 and "5" in field 6. The remaining fields are not needed and are left blank.

Message-Oriented Signaling

Much of the capability of an ISDN comes from the use of messages. Messages are used to convey call control signaling and other call related (and in some cases non call related) information between switching nodes and between end points. It is this messaging facility that gives ISDN its power and versatility.

Codesets

In the ISDN message structure, there are eight possible codesets, numbered 0 through 7. There codesets are standardized by the CCITT as follows:

- **Codeset 0**

  A set of basic IEs (Information Elements) defined by the CCITT (call control and signaling). This codeset should be standard for all ISDN applications, regardless of country where used or vendor.

- **Codesets 1 through 4**

  Reserved for future CCITT standards expansion.

- **Codeset 5**

  Reserved for national use. This codeset would be under the control of the national level telecommunications authority [such as, the FCC (Federal Communications Commission) in the U. S., and the Bundispost in The Federal Republic of (West) Germany]. The structure and content of this codeset could differ from one country to another, and therefore IEs from this codeset would probably not be passed on international calls.

- **Codeset 6**

  For IEs specific to the local serving network. The local serving network in this case is the terminating or serving switch (in the DEFINITY Generic 2 time frame). These are PRI terminations and IEs deal with maintenance and management information, and switch supported features and services such as calling and called party name and number display information. The structure of this codeset could vary from one vendor to another.

- **Codeset 7**

  For user-specific IEs. These IEs are used at the call termination point (such as, voice or data terminal, host computer, etc.). These IEs are specifically vendor-oriented and relate to terminal type features. The structure of this codeset will, most certainly, vary from one vendor to another.

IEs (Information Elements) or Codepoints

Each codeset contains up to 133 specific IEs. A specific IE within a specific codeset is also referred to as a ***codepoint.*** The IEs identify specific information contained in the ISDN message. This information is related to things such as terminal capabilities, lamp and ringing information, button and switchhook state changes, data rates, and so forth.

Codeset 0 is significant as it contains the CCITT standardized call processing IEs needed to make basic communications work within an ISDN. Tables G-C through G-G list and describe the IEs and related encodes that form codeset 0 and are used in System 85, Release 2, Version 4 and in DEFINITY Generic 2.

TABLE G-C. ISDN Message Oriented Signaling — Codeset 0

IE	Name and Parameter	Description
BC	**Bearer Capability**	Network provided protocol to support user-network interface for voice and data.
	Coding Standard	CCITT Q.931 (only valid source of coding).
	Information Transfer Capability	Type of information to be passed, i.e., speech, 3.1 Khz audio, mode 2 data, etc.
	Transfer Mode	Circuit (only valid method for DEFINITY Generic 2).
	Information Transfer Rate	64 Kbps; fixed channel rate for DEFINITY Generic 2.
	Layer ID	Layer 1
	User Layer Protocol	μ-law speech (only valid parameter)
CA	**Cause**	Used to provide reason for status or disconnect message.
	Location	User, Local Network etc.
	Class	Normal, Network Congestion, etc.
	Class Value	See Table G-F.
CDN	**Called Party Number**	Used to provide called party on redirection.
	Type of Number	Local, National, International.
	Number Plan	Private, ISDN, Telephony, etc.
	Number	Digits (ASCII).
CGN	**Calling Party Number Number**	Used to provide calling party on redirection.
	Type of Number	Local, National, International.
	Number Plan	Private, ISDN, Telephony, etc.
	Number	Digits (ASCII).
CI	**Channel Identification**	Used to identify a channel within the interface.
	Channel Selection	0=this channel preferred; 1=this channel only. (For channel negotiation purposes).
	Channel Information	0=no channel; 1=B1; 2=B2; 3=any channel.

TABLE G-C. Codeset 0 (Contd)

IE	Name and Parameters	Description
CN	**Connected Number**	Returns 10-digit number of receiving party to calling party.
CS	**Call State**	Used to provide current call state information.
	Call Status Value	For BRI calls see Table G-E for call state codes.
KEY	**Keypad**	Used to send keypad information to network.
	Keypad Information	IA5 Characters, maximum of 40.
KC	**Keypad Control**	Used to inform user which call reference to associate keypad information with.
	Call Reference Type	0=Null Call Reference 1=Non-Null Call Reference.
LS	**Look Shift**	Used to shift to new codeset (code sets 6 and 7 are valid possibilities).
	Shift Identifier	Fixed at 1.
	New Codeset Identifier	Either 6 or 7.
MT	**Message Type**	Function of message being sent.
	Function	See Table G-G for description of message types.
PI	**Progress Indicator**	Used to describe call related progress through the network
	Location	User, Local Network (switch), etc.
	Progress Description	Not end-to-end ISDN, etc.
PD	**Protocol Discriminator**	Used to indicate message protocol.
	Protocol Value	1=Maintenance; 8=Call Control; etc.
SIG	**Signal**	Used to request user generated tones.
	Value	See Table G-H for list of valid tones.
SWH	**Switchhook**	Used to inform the network (switch) of the users switchhook state.
	Switchhook Value	0=on-hook; 1=off-hook.

TABLE G-C. Codeset 0 (Contd)

IE	Name and Parameters	Description
LLC	**Low Layer Compatibility**	Used to identify low layer features and services used for data calls.
	Coding Standard	CCITT Q.931 is only valid standard.
	Information Transfer Capability	Unrestricted, restricted, 3.1 Khz audio, etc.
	Transfer Mode	Circuit is only valid mode for DEFINITY Generic 2.
	Information Transfer Rate	64 Kbps is only valid rate.
	Structure	8 Khz integrity.
	Configuration	Point-to-point is only valid configuration in DEFINITY Generic 2.
	Establishment	Demand (default); Permanent (for Dedicated Switch connections).
	Symmetry	Bidirectional symmetric (only valid option)
	Layer ID	1=Layer 1; 2= Layer 2; 3= Layer 3.
	User Information Protocol	For layer: 1. µ-law speech 2. Q.921 LAPD 3. Q.931.
NSF	**Network Specific Facilities**	Identifies network features and services that are to be used (have been used) for the call.
TNS	**Transit Network Selection**	Identifies the common carrier to be used for the call.
UUI	**User-to-User Information**	Provides end user to end user signaling information (ASCII Characters); maximum of 64 characters.

TABLE G-D. ISDN Message-Oriented Signaling — Call States

ISDN Terminal (BRI) Call States			
Terminal State (Encode)	**State Name**	**Direction**	**Description**
U0 (U0)	Null	N/A	No call exists (status only).
U1 (00001010)	Call Initiation	Outgoing	Call origination establishment.
U2 (00001010)	Overlap Sending	Outgoing	Called party digits being sent.
U3 (*)	Outcall Proceeding	Outgoing	All called number digits sent.
U4 (*)	Call Delivered	Outgoing	Ringing (alerting) is applied to called station.
U7 (U7)	Call Received	Incoming	Local ringing (alerting).
U8 (00001010)	Connect Request	Incoming	Local station answer.
U9 (*)	Incall Proceeding	Incoming	Call proceeding, wait for answer.
U10 (00001010)	Active	Both	Call is active.
U11 (U0)	Disconnect Request	Both	Local disconnect
U12 (*)	Disconnect Indication	Both	Disconnect request received.
U19 (*)	Release Request	Both	Request release of B-channel.

* State not implemented or used in DEFINITY Generic 2.

Switch (Network) Call States (For BRI Only)			
State	**State Name**	**Direction**	**Description**
N0	Null	N/A	No call exists.
N1	Call Initiation	Outgoing	Call origination establishment.
N2	Overlap Sending	Outgoing	Waiting for called party digits.
N3	Outcall Proceeding	Outgoing	All called party digits received.
N4	Call Delivered	Outgoing	Called party (far end) ringing.
N6	Call Present	Incoming	Call offered, waiting acceptance.
N7	Call Received	Incoming	Called party (local) ringing.
N8	Connect Request	Both	Called party (local) answered.
N9	Incall Proceeding	Incoming	Called party (local) call acceptance.
N10	Active	Both	Active call.
N11	Disconnect	Both	Request disconnect sent (local).
N12	Disconnect Indication	Both	Received a to disconnect request (local).
N19	Release Request	Both	Request B-channel release (local).

TABLE G-E. ISDN Message-Oriented Signaling — PBX Cause Class Values

Number	Class	Value	Cause	Diagnostic (Optional)
16	001	0000	Normal	None
17	001	0001	User Busy	None
18	001	0010	No User Responding	None
21	001	0101	Call Rejected	User Supplied
22	001	0110	Number Changed	New Destination Address
25	001	1001	Call Resumed	None
26	001	1010	Invalid Destination Address	None
28	001	1100	Invalid Number Format	None
29	001	1101	Requested Facility Rejected	Facility Identification
34	010	0010	No Channel Available	None
42	010	1010	Network Congestion	Network Identity
43	010	1011	User Information Discarded	None
50	011	0010	Requested Facility Not Subscribed	Network Identity, Facility
54	011	0110	Incoming Calls Barred User Specified Information	Destination Address
58	011	1110	Bearer Capability Not Presently Available	None
65	100	0001	Bearer Service Not Implemented	None
66	100	0010	Channel Type Not Implemented	Channel Type
68	100	0100	Message Not Implemented	None
69	100	0101	Requested Facility Not Implemented	Network Identity, Facility
81	101	0001	Invalid Call Reference Value	None
82	101	0010	Identified Channel Does Not Exist	Channel Identity
88	101	1000	Incompatible Destination	Incompatible Parameters

TABLE G-E. ISDN Message Oriented Signaling — PBX Cause Class Values (Contd)

Number	Class	Value	Cause	Diagnostic (Optional)
96	110	0000	Mandatory Information Element is Missing	Information Element Identifier
97	110	0001	Message Type Non-existent or Not Implemented	Message Type
98	110	0010	Message Not Compatible with Call State	Message Type
100	110	0100	Invalid Information Element Contents	Information Element
127	111	1111	Interworking Unspecified	None

NOTE: Class Categories are:

000 — Normal

001 — Normal Event

010 — Resource Unavailable

011 — Service or Option Not Available

100 — Service or Option Not Implemented

101 — Invalid Message (i.e., parameter out of range)

110 — Protocol Error (i.e., unknown message)

111 — Interworking.

TABLE G-F. ISDN Message-Oriented Signaling — Message Types

Message	Source	Description
ALERTing	Both	Indicates user ringing (alerting) initiated.
CALL PROCeeding	Both	Indicating call setup in progress.
CONFerence	User	Requests the Conference feature.
CONFerence ACKnowledge	Network	Acknowledges conference connection.
CONFerence REJect	Network	Rejects conference request.
CONNect	Both	Indicates called party answer.
CONNect ACKnowledge	Both	Acknowledges acceptance of called party answer.
DISConnect	Both	Request called or calling party disconnect.
DROP	User	Requests last conference party be dropped.
DROP ACKnowledge	Network	Acknowledge last conference party dropped.
DROP REJect	Network	Rejects dropping last conference party.
ENDpoint SERVice	Both	Requests maintenance service state change.
ENDpoint SERVice ACK	Both	Acknowledges requested maintenance service state change.
HOLD	User	Requests other party be place on hold.
HOLD ACKnowledge	Network	Acknowledges that other party is placed on hold.
HOLD REJect	Network	Rejects placing other party on hold.
INFOrmation	Both	Call control information.
MANagement INFOrmation	Both	Maintenance and management message.
PROGress	Network	Indicates the progress of call setup.
RECONNect	User	Request to reconnect held call or for bridging.
RECONNect ACKnowledge	Network	Acknowledges reconnection or bridging.
RECONNect REJect	Network	Rejects reconnect or bridging request.
REDIRect	Network	Requests return to overlap sending state.
RELease	Both	Requests release of call in progress.
RELease COMplete	Both	Indicates release of call is completed.
RESTart	Network	Request to restart terminal in unknown state.
RESTart ACKnowledge	User	Acknowledges restart of terminal.
SETUP	Both	Request for call setup.
SETUP ACKnowledge	Network	Acknowledge that more setup information is needed.
STATUS	Both	Indicates the current state of the terminal.
STATUS ENQuiry	Both	Request for the current state of the terminal.
TRANSfer	User	Request to transfer a call.
TRANSfer ACKnowledge	Network	Acknowledges that the call has been transferred.
TRANSfer REJect	Network	Rejects the request to transfer a call.

TABLE G-G. ISDN Message-Oriented Signaling — Tone Signals

Encode	ISDN Tone	Tone Description
00000000	Dial tone on	Dial tone
00000001	Ringback tone on	Ringback
00000010	Intercept tone on	Intercept tone
00000011	Reorder tone on	Reorder tone
00000100	Busy tone on	Busy tone
00000101	Confirmation tone on	Stutter dial tone
00000110	Answer tone on	Data Carrier tone
00000111	Call waiting tone on	Utility tone
00001000	Offhook warning tone on	No equivalent
00001001	Custom tone on	Utility tone
00001011	Busy verify tone on	Utility tone
00001100	Error tone on	No equivalent
00001110	Tones off	No tone connected
Encode	**ISDN Alerting**	**Ringing Description**
01000000	Normal alerting	Standard ringing (single burst)
01000001	Intergroup distinctive	Distinctive ringing (two burst)
01000010	Priority distinctive	Priority ringing (three burst)
01000011	Intercom	Intercom (repeated buzz)
01000100	Ping Ring	Ring Ping
01000101	Precedence Call	No equivalent
01000110	Alerting Pattern 6	No equivalent
01000111	Alerting Pattern 7	No equivalent
01001111	Alerting off	No ringing

Codeset Conversion

As part of the evolution of ISDN, changes have occurred in codeset designation between the System 85, Release 2, Version 4 and DEFINITY Generic 2. These changes are not limited to AT&T PBX switches but are the result of an upgrade in the AT&T ISDN—PRI specification and affect all AT&T products that use ISDN messaging.

Specifically, IEs that were contained in codeset 7 for System 85, Release 2, Version 4 have been moved to codeset 6 in DEFINITY Generic 2, while a new set of IEs has been established in codeset 7. This necessitates codeset conversion for messages being passed between System 85, Release 2, Version 4 and DEFINITY Generic 2 switches. Toll office 4 ESS (4E11 and 4E13) switches are also effected by this codeset change. Codeset conversion applies specifically to ISDN—PRI and is discussed in more detail under that feature.

The Implementation

In DEFINITY Generic 2, ISDN is implemented through the following features:

- ISDN—BRI (Basic Rate Interface)
- ISDN—PRI (Primary Rate Interface).

Application Features

The previously listed features represent the direct implementation of ISDN in DEFINITY Generic 2. In addition, the following features use ISDN capabilities or are direct applications of one or more ISDN feature.

- Bearer Capability
- DMI—MOS (Digital Multiplexed Interface with Message-Oriented signaling)
- Look Ahead Interflow.

Common Elements

With the standard interfaces (BRI, PRI, or DCP), ISDN uses common elements.

- Both the BRI and PRI use digital, 64 Kbps B channels for communications.
- Both also use common channel (out of band), message based signaling.
- Both BRI and PRI are compatible with the same networks and with each other.
- All implementations of ISDN use Message-Oriented Signaling.
- All of the ISDN capabilities and services rely on the standard ISDN message sets. For some special application, user peculiar messages are implemented; however, these are also provided for within the ISDN standards (Codeset 7).

Maintenance Services for ISDN Facilities

ISDN recommendations provide for both local and remote testing of ISDN facilities. This is supported on System 85 by both ***on demand*** and ***automatic*** testing of ISDN hardware connections and the service level (bit error rate) provided by ISDN facilities. For example, checks are made automatically (by DEFINITY Generic 2) for hyperactive ISDN channels (D channels), and if hyperactivity is detected, the faulty channels are automatically taken out of service and reported.

ISDN Compatible Switches

The ISDN can be used for either public or private networks. The following AT&T switches support (or will soon support) the ISDN standards:

- 4 ESS Switch (4E11 and later) — Commonly used for Toll Offices in public networks.
- 5ESS Switch [5E4.1 (BRI only), 5E4.2 and later, and 5E5 (both BRI and PRI)] — Commonly used as Central Office switches in public networks but also available for large PBX applications.
- System 85 Switch [R2 V4 (PRI only) and DEFINITY Generic 2 (both BRI and PRI)] — Commonly used for medium to large PBX requirements
- DEFINITY Generic 1 — Commonly used for medium PBX sizing requirements.

Appendix H: The DCIU (Data Communications Interface Unit)

General Concepts

This appendix provides a general description of the DCIU and its functions. Some recommended solutions for typical DCIU administration situations are also provided. The DCIU is used with the following features:

DCS (Distributed Communications System)

- Direct connection between two switch processors
- Indirect connection between two switch processors with one or two intervening processors (hop connection)
- Centralized messaging services in a DCS
- Connection to a switch processor that is not a System 85 or DEFINITY Generic 2
- Alternate routing connections.

Applications Processor Features

- Call Management System
- Leave Word Calling
- Message Center Service.

AUDIX.

The solutions recommended for given applications are not necessarily the only way the DCIU could be administered for the example; however, they should work and can be used to clarify DCIU administration requirements.

The DCIU

The DCIU is a special purpose processor that operates as a packet switch. It is located in the Common Control Cabinet of the System 85 or DEFINITY Generic 2. The DCIU receives information from the switch processor, assembles this data into units called packets, and then routes these packets of data to the appropriate distant processor.

Data Paths: Packets of data are transmitted over specially designated ***virtual circuits*** called data paths. They are virtual circuits because they are composed of logically or software defined elements rather than discrete physical components. A data path consists of all the circuit elements that provide the communications connection between the local switch processor and a specific external application. Data paths are discussed in more detail later in this appendix.

The DCIU also receives packets from distant processors. It routes this information either to the local switch processor or to another distant processor. The routing of packets is based on internal instructions contained in the DCIUs firmware (entered through the DCIU Administration process).

Releases

Two DCIU releases are available. The DCIU used in a given application will depend on the switch to which it is attached.

Release 1 DCIU

The Release 1 DCIU provides up to 4 links with 20 logical channels each and is used by DIMENSION System FP8 switches and System 85, Release 1 switches.

Release 2 DCIU

The Release 2 DCIU provides up to 8 links with up to 64 logical channels each and is used by System 85, Release 2 and DEFINITY Generic 2 switches.

Both DCIU releases support the same general applications; however, the Release 2 DCIUs provide an increased capacity (more links with more logical channels on each link) and a somewhat increased functionality. For example, in the DCS application the Release 2 DCIU provides an alternate routing capability while the Release 1 version does not. alternate routing is described in the DCS feature chapter of this manual and later in this appendix.

SCI (Switch Communications Interface)

System 75 and DEFINITY Generic 1 switches do not use a DCIU but have their own circuit, the SCI, that performs the same functions. The SCI is compatible with the DCIU link and is functionally similar to the Release 1 DCIU. The SCI provides up to 4 DCS links with up to 64* logical channels each.

Figure H-1 depicts the Release 2 DCIU as a box with nine links (0 through 8). The outside links (1 through 8) connect to other DCIUs, and the internal link (link 0) connects to the local switch processor. The intend link is referred to as the switch link to distinguish it from the outside DCIU links.

The most significant difference between the switch link and the DCIU links is the protocol they use. The switch link uses DMA (Direct Memory Access), while the DCIU links use the BX.25 transmission protocol. The logical channels on the switch link are called ***ports.*** Logical channels or ports on different links are associated through virtual circuits called network channels.

* The DCS links on the System 75, R1 V2 XE have 22 logical channels. The links on the R1 V3 XE have 18 logical channels. The links on the R1 V3, Issue 1.4 XE have 64 logical channels. The link on the rest of the System 75 versions DEFINITY Generic 1 have 64 logical channels, or the switch can be upgraded so that it does.

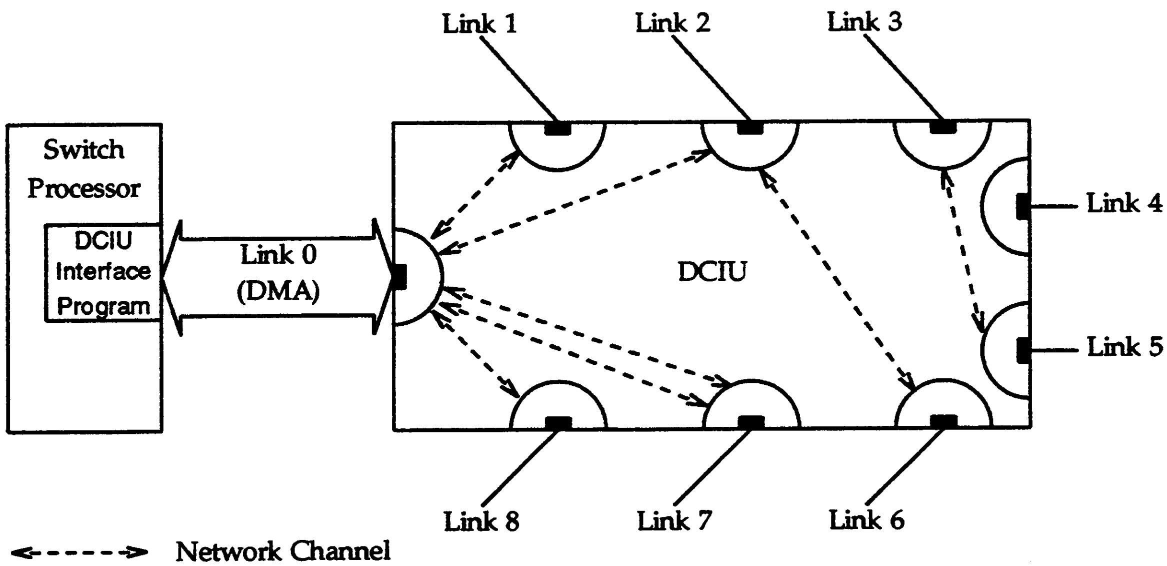


Figure H-1. Release 2 DCIU

Data Paths

In the context of DCIU communications, data paths are virtual circuits that form the complete route used by applications messages (packets) from the time they enter the DCIU environment (incoming port) to the time they are sent on to the final processor (outgoing channel or port at the destination switch, adjunct or applications processor). This concept sometimes appears to work differently in a DCS environment than in other DCIU applications. The differences are in the complexity of the data paths used and are more apparent than real.

In the DCS Environment

Each node in a DCS Cluster must have a data path to every other node. In a DCS, these data paths allow the switch processors to exchange call processing information, and it is this exchange of information that enables the network to function as one switch. With centralized messaging service, there must also be a data path between the messaging adjuncts and the switches they serve. ***DCIU Links*** and ***Network Channels*** provide these data paths. These data paths may use multiple DCIU links and network channels and can sometimes be setup dynamically. These arrangements are described in detail in the DCS chapter of this manual and later in this appendix under ***DCIU Linkage Options.***

The DCIU Link

The DCIU link is the physical or hardware connection between the DCIU and an external processor. External processors include APs, AUDIX Adjunct, other DCIUs, and SCIs. The DCIU links are identified as links 1 to 8 on Release 2 DCIUs (links 1 to 4 on Release 1 DCIUs and SCIs). The DCIU link is sometimes called a ***signaling link.*** It provides a full duplex, synchronous data path between the DCIU and the external processor.

The DCIU link must be a clear channel dedicated circuit. Traditionally this has meant that each DCIU link was a discrete physical circuit. However, with the availability of digital Dedicated Switch Connections, a DS0 channel in a DS1 carrier can serve as the physical connection between two DCIU circuits. (This connectivity can provide long-haul transmission of the DCIU protocol at a lower cost.)

The DCIU link is a major part of the ***data path*** but is only part of the data path. A full data path consists of a switch link and port, one or more network channels, and one more DCIU link logical channel pairs. A ***link logical channel pair*** is a specific logical channel on a specific DCIU link. The DCIU links carry applications information packets between processors via ***logical channels.***

Logical Channels

A logical channel is a segment of the data stream carried on a link. A logical channel carries a packet or frame of data. Packets and frames, in the DCIU context, are essentially the same thing. A frame actually contains some additional header and trailer information that is used by the BX.25 protocol but does not otherwise affect the information content of the packet.

Switch Link Ports

For the Switch Link (Link 0), logical channels are called ***ports.*** For Release 2 DCIUs used on System 85, Release 2, Version 3 and earlier switches, these ports have fixed reservations. These reservations are shown in Table H-A and must be used as reserved. That is, Port 1 is reserved for Message Center Service on AP 1. Port 1 must be used for this purpose and only this purpose. If AP 1 does not support Message Center Service (or there is no AP 1), Port 1 cannot be used.

Logical Channel Limitations

For the switch link and DCIU links when both ends of the link connect to a Release 2 DCIU, each link supports up to 64 software-assigned logical channels. If either end of a DCIU link connects to a Release 1 DCIU, the link is limited to 20 logical channels.

Network Channels

A DCIU passes data between links along internal virtual circuits called network channels. Like the DCIU links, network channels are another major part of the data path. Network channels are set up in administration (Procedure 257, Word 1) and provide routing for packets through the DCIU (from the incoming link logical channel or port to the outgoing link logical channel or port).

When a packet is received, it is routed via a network channel to the appropriate logical channel (or port) on the outgoing link. For example, data intended for the local switch is routed to a designated port on the switch link (link 0), and data intended for a distant processor is routed to the appropriate logical channel and DCIU link. For packets intended for a distant switch in a DCS, this can work differently depending on the release DCIU involved.

- In a Release 1 DCIU, network channels are given fixed assignments. That is, a specific network channel always associates packets received on a particular link/logical channel or port with a specific logical channel on the link leading to a distant switch or other application. This arrangement is called a ***fixed network channel*** or PVC (Permanent Virtual Circuit).
- In Release 2 DCIUs, the packet routing concept is modified to allow both fixed network channels and ***alternate routing.*** The fixed network channels work the same as in a Release 1 DCIU. With alternate routing, the DCIU can set up network channels based on the ***destination routing code*** contained in header information in the frame. With alternate routing up to three routes (a primary route and one or two alternate routes) can be assigned to each destination routing code. Alternate routing is discussed in detail later in this appendix and in the DCS chapter of this manual.

TABLE H-A. Release 2, DCIU Switch Link Port Reservations (NOTE)

PORT	APPLICATION	PORT	APPLICATION	PORT	APPLICATION
1	MCS 1	23	DCS 3	44	DCS 15
2	LWCH 1	24	DCS 4	45	CLK 6
3	LWCL 1	25	DCS 5	46	MCS 6
4	AMWL 1	26	DCS 6	47	LWCH 6
5	TRAF	27	DCS 7	48	LWCL 6
6	TEST 1	28	DCS 8	49	AMWL 6
7	SMDR	29	DCS 9	50	DCS 16
8	CLK 1	30	MCS 4	51	DCS 17
9	CLK 2	31	LWCH 4	52	DCS 18
10	MCS 2	32	LWCL 4	53	CLK 7
11	LWCH 2	33	AMWL 4	54	MCS 7
12	LWCL 2	34	DCS 10	55	LWCH 7
13	AMWL 2	35	DCS 11	56	LWCL 7
14	CLK 3	36	DCS 12	57	AMWL 7
15	MCS 3	37	CLK 5	58	DCS 19
16	LWCH 3	38	MCS 5	59	AUDIX 1
17	LWCL 3	39	LWCH 5	60	AUDIX 2
18	AMWL 3	40	LWCL 5	61	AUDIX 3
19	CLK 4	41	AMWL 5	62	AUDIX 4
20	TEST 2	42	DCS 13	63	SPAR 3
21	DCS 1	43	DCS 14	64	SPAR 2
22	DCS 2				

NOTE: The numbers to the right of the application refer to the adjunct number for that application. That is, MCS 1 refers to Message Center Service on AP 1.

LEGEND:

MCS	Message Center Service
LWCH	Leave Word Calling, high priority
LWCL	Leave Word Calling, Low Priority
AMWL	Automatic Message Waiting Lamp
TRAF	Traffic data
SMDR	Call Detail Reporting and Recording
CLK	AP clock synchronization
DCS	Distributed Communication System
TEST	DIP/DCIU test
AUDIX	Audio Information Exchange
SPAR	Spare, not assigned

BX.25 Protocol

The DCIU uses the BX.25 protocol to pass messages to other DCIUs. The BX.25 protocol is a layered protocol, like the international standard (X.25) for packet switching networks. Layering separates specific functions into distinct levels that communicate with matching levels at a distant processor. Layering also enables successive levels of the protocol to deliver specific services to higher levels, without concern for the details of underlying layers. The BX.25 protocol segments data to be transmitted into packets, appends framing and routing information to each packet, and queues them into a buffer where they await delivery.

Packets are numbered, and the DCIU keeps a copy of each packet. As it sends out a packet, the DCIU sets a timer. If confirmation of delivery is not received within a specific time period (this time period is set in administration, Procedure 256), the packet is retransmitted. The DCIU link can also be reset and restarted if packets are not delivered or arrive out of sequence.

DCIU Linkage Options

Two DCIU linkage options are available: direct and indirect linkage. Figure H-2 illustrates both of these types of linkage in a DCS. The DCIU link between node 1 and node 2 is a direct linkage connection, and the connection between node 1 and node 3 is indirect.

General Application

In most DCIU applications, the linkage used will be ***direct.*** In direct linkage, the DCIUs for the sending and receiving processors are connected directly by a single carrier. That is, there is a direct physical connection between the link leads of the sending and receiving DCIUs. There are no intervening DCIUs or other processors. This does not exclude necessary interfacing devices such as modems or adapters (see **The DCIU-to-DCIU Interface** later in this appendix). Direct linkage is used where both processors are colocated (on-premises) such as local APs and adjunct processors, and where only two processors are involved (i.e, a two node DCS).

DCS Applications

In the DCS application, two options are available for DCIU linkage. These are direct and indirect linkage. Direct linkage is the same as described in the preceding paragraph. Indirect linkage uses one or two hops between nodes. The term "hop" is used when the data path passes through an intervening DCIU before reaching its destination. This is similar to the tandeming process for tie trunks but should not be confused with tie trunk tandeming. In this appendix, the term "**hop**" is used to distinguish DCIU data paths from communications paths.

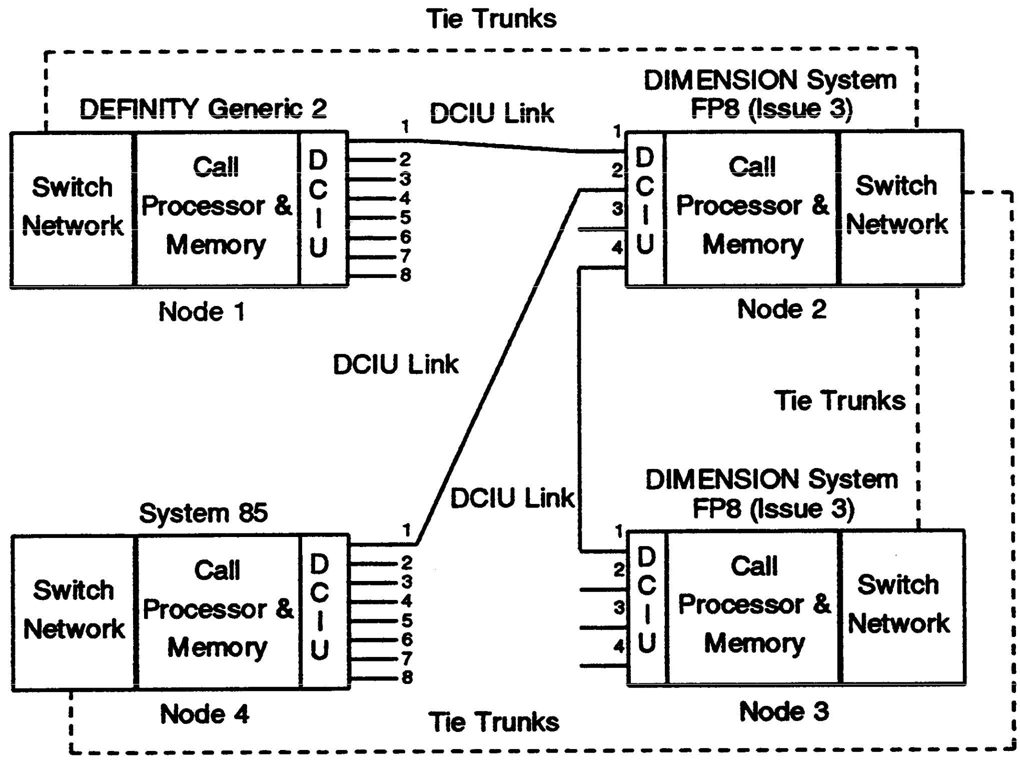


Figure H-2. Link Minimization DCS Configuration

Indirect Linkage

Figure H-2 shows a DCS cluster with a minimum linkage configuration. Link minimization uses indirect linkage to provide the minimum essential data paths needed for DCS service. At optimum, there would be only one data path possible between any two nodes. Notice that there are no direct data links available between nodes 1 and 3, 1 and 4, or between nodes 3 and 4. All data paths between nodes 1, 3, or 4 are indirect. They must pass or hop through node 2. Because this arrangement requires less hardware, it is less expensive than direct linkage. It uses fewer link connections at each DCIU (except the hub in a star configuration) leaving more links available for other applications. However, minimum linkage is less reliable than direct linkage. For example, failure of the data link between nodes 2 and 4 (Figure H-2) will cause loss of transparency between node 4 and all other nodes in the network.

Direct Linkage

Direct linkage provides ***"no hop"*** data paths between nodes in a DCS. A fully implemented direct linkage configuration would provide a no hop data path between every node in a DCS. In a large or highly disbursed network, fully implemented direct linkage is not always practical and usually some combination of direct and indirect linkage is used. The principle advantage to direct linkage is reliability. In the example shown in Figure H-3, if the data link between node 1 and node 2 fails, transparency is lost only between these two nodes. If ***alternate routing*** were available, transparency could be maintained even between nodes 1 and 2.

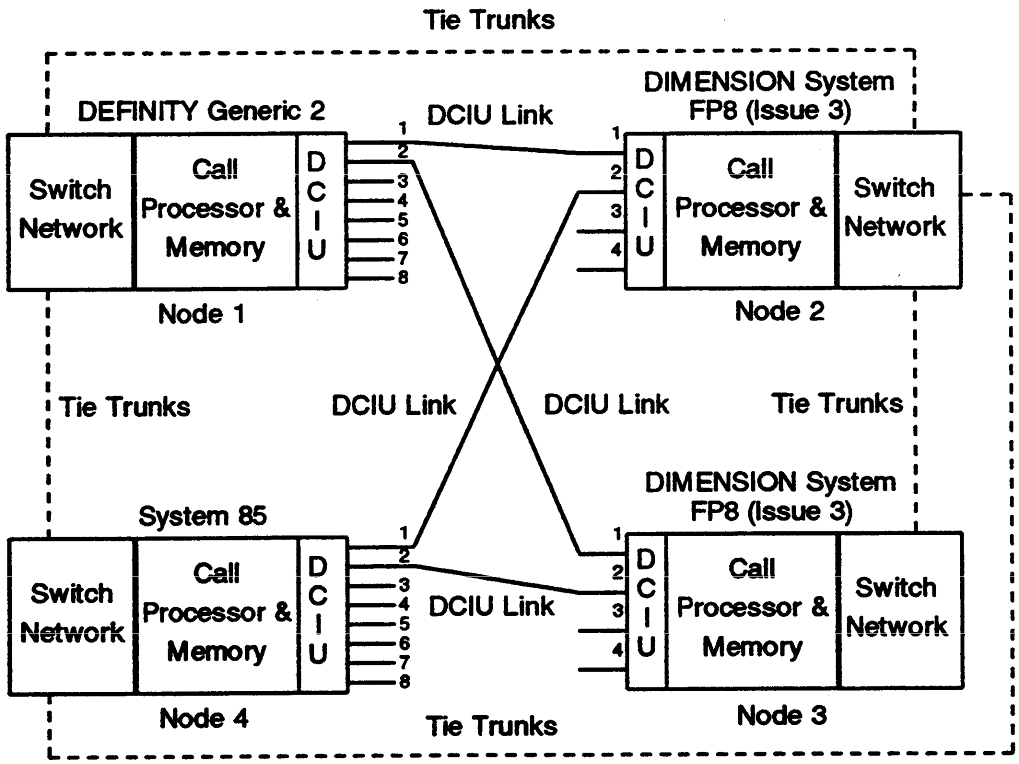


Figure H-3. Direct Linkage DCS Configuration

The problems with direct linkage are: for a system with three or more nodes, direct linkage requires more hardware and is, therefore, more expensive than a link minimization arrangement; direct linkage limits the number of nodes that can be included in a DCS; and direct linkage uses DCIU connections that maybe needed for other applications.

Alternate Routing of DCIU Messages

Alternate routing is assigned on a per-network channel basis and provides a way for DCIU messages (DCS applications) to bypass a failed DCIU link. Alternate Routing is available only on data paths where Release 2 DCIUs are used at both ends. The use of link minimization configurations tends to limit or eliminate the effectiveness of alternate routing.

Network Channels

Alternate routing is enabled for a network channel using switch administration Procedure 257, Word 1. When a network channel is assigned alternate routing, the second link and logical channel (Component B) is filled with dashes. The alternate routing network channel is then assigned a ***destination routing code*** in Procedure 257, Word 2. In effect, the destination routing code takes the place of the second logical channel link pair for alternate routing network channels. When alternate routing is used, the packet header includes the destination routing code and postage.

Destination Routing Code

The destination routing code identifies the node for which a packet is intended. It must be common throughout the network. That is, a given destination routing code must identify the same destination switch from any node in the system. The destination routing code is used at each alternate routing DCIU to select up to three routes (a primary and two alternate routes) that can be used from that DCIU to reach the destination switch. At the DCIU serving the destination switch, only one route (the primary) is used. This route passes the packet to the designated port on the switch link. Alternate routing network channels are assigned to destination routing codes at each alternate routing node using Procedure 257, Word 4.

Postage

Postage is the system's way of keeping track of the number of hops that have been used a data path is limited to a maximum of two hops. When the DCIU receives a packet that is using alternate routing it decrements the postage; and if the result is negative, it discards the message (the two hop limit has been used up). If the result is positive, the DCIU selects a network channel from the routing table that serves the destination routing code of the packet.

Fixed Network Channels

Alternate routing paths can use fixed routing network channels or PVCs at hops or intervening nodes. When an alternate routing path passes through a node that uses a PVC, it works just like a nonalternate routing data path. That is, it passes from the incoming link logical channel to a specific, predesignated outgoing link logical channel. However, alternate routing packets must begin and end on alternate routing nodes. The additional header information used for alternate routing (destination routing code and postage cannot be provided or deleted by a fixed network channel switch.

Routing on Failure

Alternate routing uses a technique called routing on failure. The alternate routing DCIU tests a selected link before sending a packet. The primary data path (first-choice routing) is used until a link failure is encountered. When a link failure occurs, the DCIU checks for an alternate (second-choice route). If an alternate route is available, that link is tested and so on. Up to two alternate routes (a second and third choice) may be assigned to serve the same destination routing code at each alternate routing DCIU.

Data Path Looping

A logical infinite loop can occur if a data path returns to a point it has already passed through and is routed over a link logical channel that it has already used. The use of alternate routing introduces the possibility of setting up a logical infinite loop. This is because alternate routing data paths are not fixed but are set up dynamically at each DCIU as the packet progresses from one link to the next. The best way to avoid an infinite loop in an alternate routing pattern is by careful network engineering. However, should a looping pattern start, the ***postage*** in the header information will eventually run out, and the loop will be stopped automatically. Additional discussion and some examples of alternate routing patterns are given in the DCS chapter of this manual.

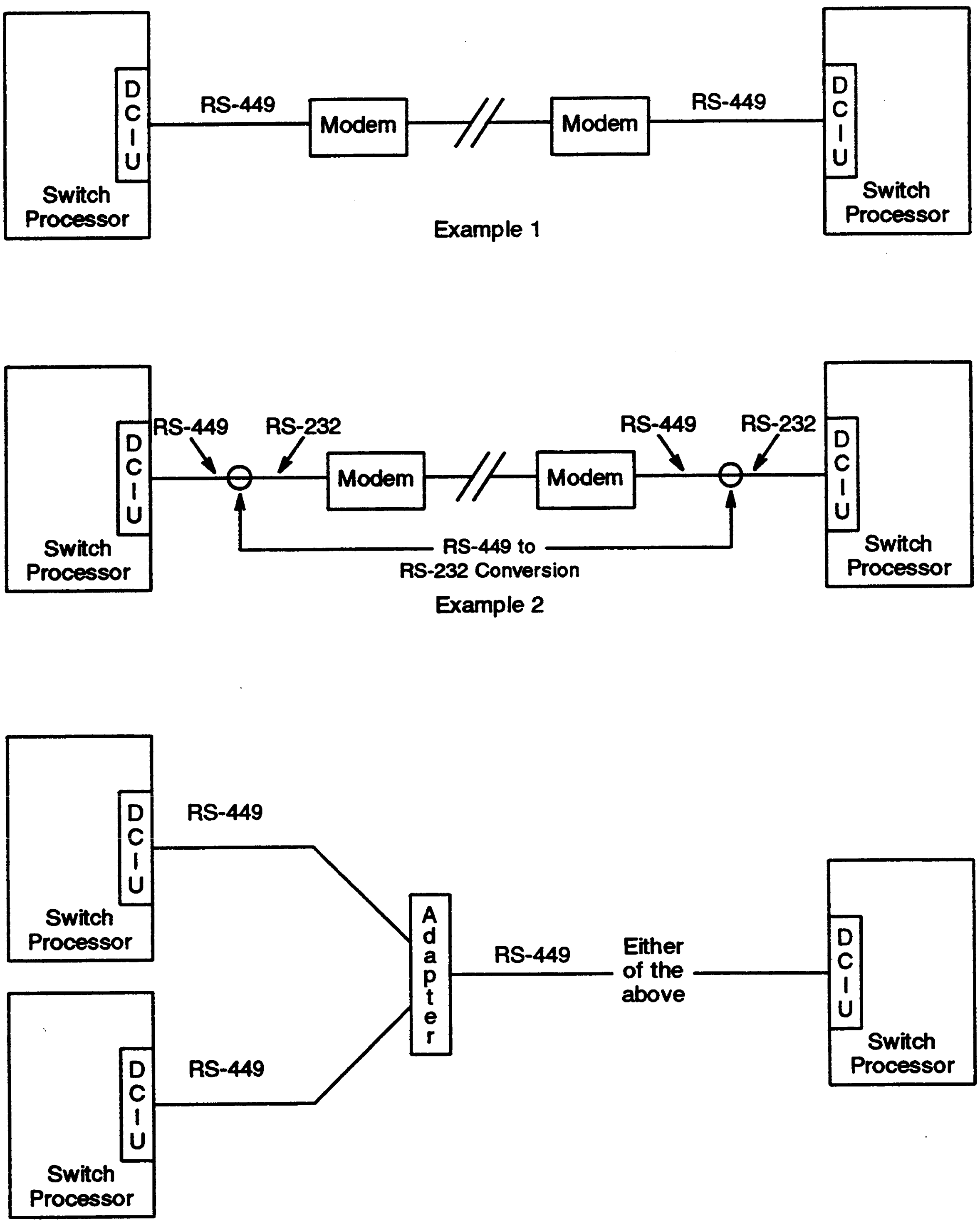


Figure H-4. DCIU Connecting Arrangements

The DCIU-to-DCIU Interface

Figure H-4 shows three examples of DCIU-to-DCIU connections. Example 1 shows a common arrangement. This data link uses standard RS-449 cable and a modem that support up to 9.6 Kbps full duplex, synchronous data. Modems are required in this type of-data link unless the connection meets all the following conditions:

- Both switches are located in the same building.
- Both switches have the same single-point ground.
- The cable distance between the DCIUs is less than 200 feet.

Example 2 shows a data link using RS-232 modems. This configuration requires the use of RS-449 to RS-232 converters (shown by the circles) because the output of the DCIU is RS-449. This is actually the same type of connection shown in Example 1 except for the type of modem used.

Example 3 shows a duplicated configuration. If a switch has a duplicated processor, the DCIU must also be duplicated but not the data links. The duplicated DCIU output is filtered by an adapter to remove duplicate packets, and a single output packet stream is sent on to the next DCIU. From this point on, the data link can use either of the arrangements shown in Examples 1 and 2.

Examples 2 and 3 assume that the conditions required for a connection not using modems are not met. These requirements are the same for all DCIU-to-DCIU connections.

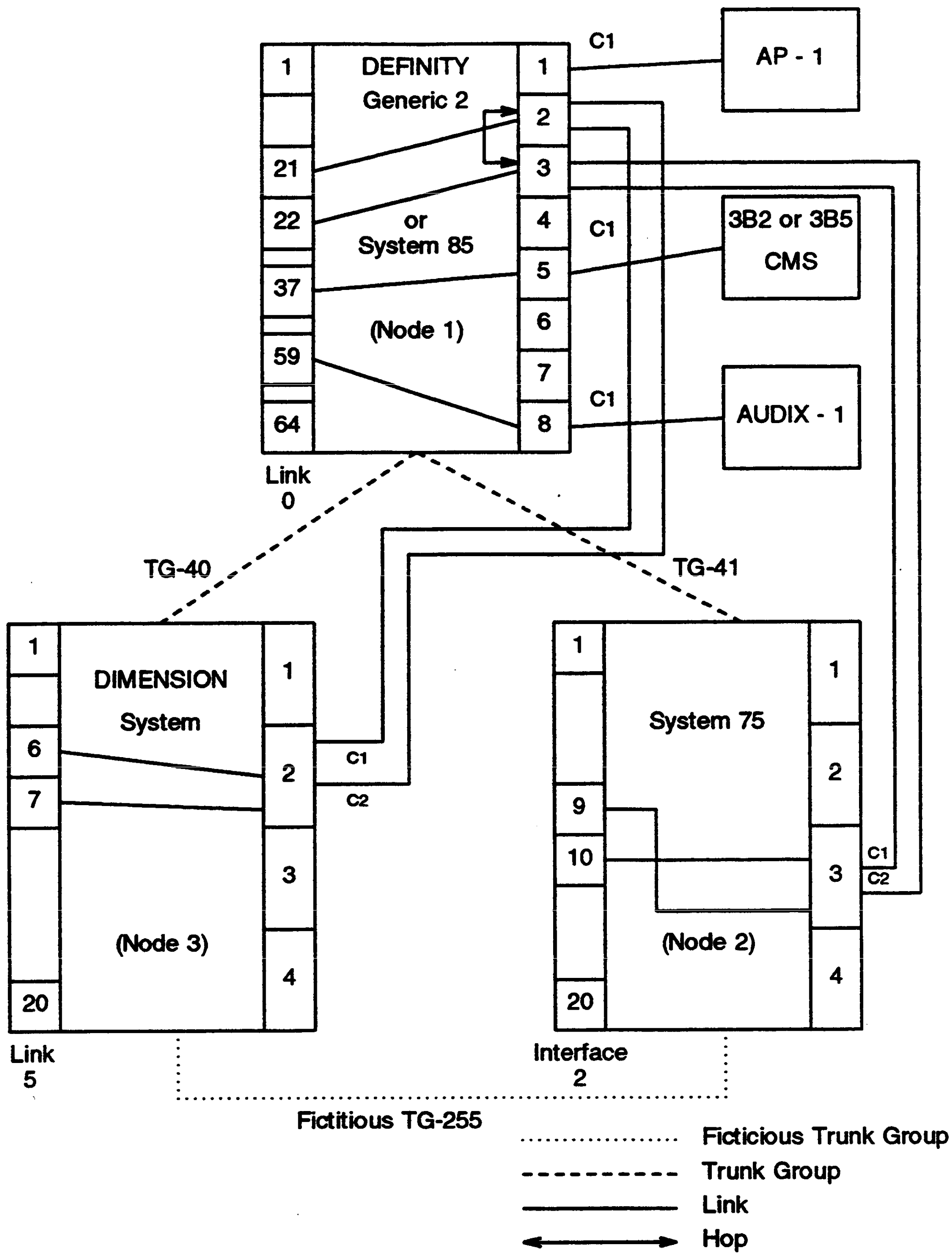


Figure H-5. Typical 3-Node DCS Administration

Basic DCIU Administration

The first example (Figure H-5) shows a 3-node DCS that uses both direct and indirect linkage. In this example, node 1 is a System 85 Release 2 or DEFINITY Generic 2 switch, node 2 is a System 75 switch, and node 3 is an Enhanced DIMENSION System (Feature Package 8, Issue 3) switch. Sample translations are for node 1 only (the System 85 or DEFINITY Generic 2 switch). They show typical ways to set up the following:

- A local AP link with Message Center Service
- A local AUDIX Adjunct
- A direct linkage DCIU path to node 3 (Enhanced DIMENSION System) for DCS messaging traffic
- A direct linkage DCIU path to node 2 (System 75) for DCS messaging traffic
- The Network Channel for the data path to node 2
- The Network Channel for the data path to node 3
- The Network Channel that supports the data path hop from node 2 (System 75) to node 3 (Enhanced DIMENSION System).

Administering the AP Link With Message Center Service

Procedure 256 — DCIU Administration — Link Assignment

Word 1, Link Assignment

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight.

Field 2: This field shows if the link has been assigned or not. A "1" means that the link is assigned, a "0" shows the link is unassigned.

Field 3: This sets the ***Baud Rate*** or data transmission rate for this link. For the AP 16, the baud rate should be set to "6" (9600 bps).

Field 4: This field identifies the transmission characteristics of the local end of the link as either DTE (Data Terminal Equipment) with encode "0," or DCE (Data Communications Equipment) with encode "1." In most cases it doesn't matter which end of a DCIU link is DTE and which end is DCE; however, the two ends of any DCIU link must not match. That is, one end must be DTE, and the other end must be DCE.

Field 5: This field identifies the link as being fixed or no dial up (encode 0) or dial up (encode 1). For most DCIU link application, this will be fixed (encode 0).

Field 6: This field identifies the protocol to be used on the link. For DCS applications, this is always be set to "1" for the BX.25 protocol.

Field 7: This field identifies the type of machine that is located at the far end of the data path being set up. For direct linkage connections, this is also the far end of the specific link. For an AP 16 attached to a System 85 switch, the appropriate encode is "1." Encodes are listed in the administration manual for the switch being administered.

Field 8: This is the DCS node number or machine number of the machine at the far end of this link.

Field 9: This field selects the memory table values to be used. If a dash or "0" is entered, the values given are scratch pad values. If a "1" is entered, the values shown are the machine table (translated) values.

Field 10: This is a display only field. If a "0" is displayed, the scratch pad and machine tables do not agree (changes have been made). If a "1" is displayed, the scratch pad and machine tables agree (no change).

BX.25 Link Characteristics

Procedure 256 — DCIU Administrtion — Link Assignment
Word 2, Level 2 Link Characteristics

Except in Field 1, typical values for link level 2 and 3 characteristics are given in the Administration Manual, Procedures (555-105-507). These typical values are generally used in both Word 2 and Word 3 of Procedure 256. If these should fail to give satisfactory service, problems must be addressed on a case-by-case basis. There are no generally recommended values other than the typical values given.

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. Default values do not apply in this field.

Field 2: This field sets the retransmission timer.

Field 3: This field sets the idle timer.

Field 4: This field sets the maximum number of retransmission attempts that will be made.

Field 5: This field sets the maximum number of unacknowledged frames allowed before retransmission.

Field 6: This field selects the memory table values used. If the value entered is "0," the displayed values are scratch pad values. If the value entered is "1," the values shown are machine values.

Field 7: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 256 — DCIU Administration — Link Assignment
Word 3, Level 3 Link Characteristics

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. This is not a default value.

Field *2:* This field sets the BX.25 activity level timer.

Field 3: This field sets the acknowledgment timer, the length of time the DCIU will wait for an acknowledgment for a transmitted message.

Field 4: This field sets the interrupt timer.

Field 5: This field sets the reset timer.

Field 6: This field sets the restart timer.

Field 7: This field shows the retransmission counter.

Field 8: This field shows the reset counter.

Field 9: This field shows the restart counter.

Field 10: This field establishes the maximum number of allowable unacknowledged packets.

Field 11: This field selects the memory table values used. If the value entered is "0," the displayed values are scratch pad values. If the value entered is "1" the values shown are machine values.

Field 12: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics

Word 1, Network Channels

Component A

This is the source component for a Network Channel.

Field 1: This is the link identified with Component A.

0 = local switch port
1 through 8 = remote switch.

Field 2: This identifies the logical channel for Component A. If Field 1 is set to "0," this is a local switch port. These ports have preassigned applications and must be used as assigned in all System 85, Release 2, Version 3 and earlier switches. Port "1" is preassigned for use with Message Center Service on AP number 1. See Table H-A for a list of all applicable port assignments. If Component A is from a distant switch (link 1 through 8), the range for Field 2 is from 1 to 64, unless the other end (Component B) is an Enhanced DIMENSION System (encode 7). If this is the case, the range is 1 to 20. Logical channels on DCIU links are not preassigned.

Procedure 257 — DCIU Administration — Data Path Characteristics Word 1, Network Channels (Contd)

Component B

This is the destination component for a Network Channel.

Field 3: This is the link identified with Component B.

0 = local switch port
1 through 8 = remote switch.

Field 4: This identifies the logical channel for Component B. If Field 3 is set to "0," this is a local switch port. These ports have preassigned applications and must be used as assigned in all System 85, Release 2, Version 3 and earlier switches. See Table H-A for a list of applicable port assignments. If Component B is from a distant processor (link 1 through 8), the range for Field 4 depends on the type of the distant processor. For a distant processor that is an AP 16, the range is from 1 to 11. The ranges applicable to different types of processors are given in the Administration Manual, Procedures. Logical channels on DCIU links are not preassigned.

Field 5: This field sets the priority of the network channel being established. A "0" sets up low priority, and a "1" sets up high priority. In the event of network channel contention, all high priority network channel messages will be served before low priority messages are served.

Field 6: This field sets the alternate routing flag. A "0" sets up fixed (Permanent Virtual Circuit) routing. This type of routing requires that Component B be identified in Fields 3 and 4. A "1" establishes this as an Alternate Routing network channel. This type of routing requires that Component B is unspecified (dashed) in Fields 3 and 4.

Field 7: This field selects the memory table used. If the value entered is "0," the displayed values are scratch pad values. If the value entered is "1," the values are machine values.

Field 8: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 DCIU Administration, Data Path Characteristics
Word 2, Port Characteristics

Field 1: This field identifies the number of the local (switch) port.

Field 2: This field shows the port number on the remote processor. The range of possible remote port numbers depends on the type of remote processor (i.e., the range of possibilities for an AP 16 is from 1 to 11). The applicable ranges for different types of remote processors are given in the Administration Manual, Procedures.

Field 3: This field identifies the destination routing code for an alternate routing network channel. If this is not an alternate routing network channel, this field is dashed.

Field 4: This field assigns the postage value for alternate routing network channels. If this is not an alternate network channel, this field is dashed.

Field 5: This is a display only field. A value of "0" shows that the port is not assigned to a network channel. A value of "1" shows that the port is assigned to a network channel.

Field 6: This is a display only field. This field shows the assigned priority of the network channel. A "0" shows low priority, and a "1" shows a high priority channel.

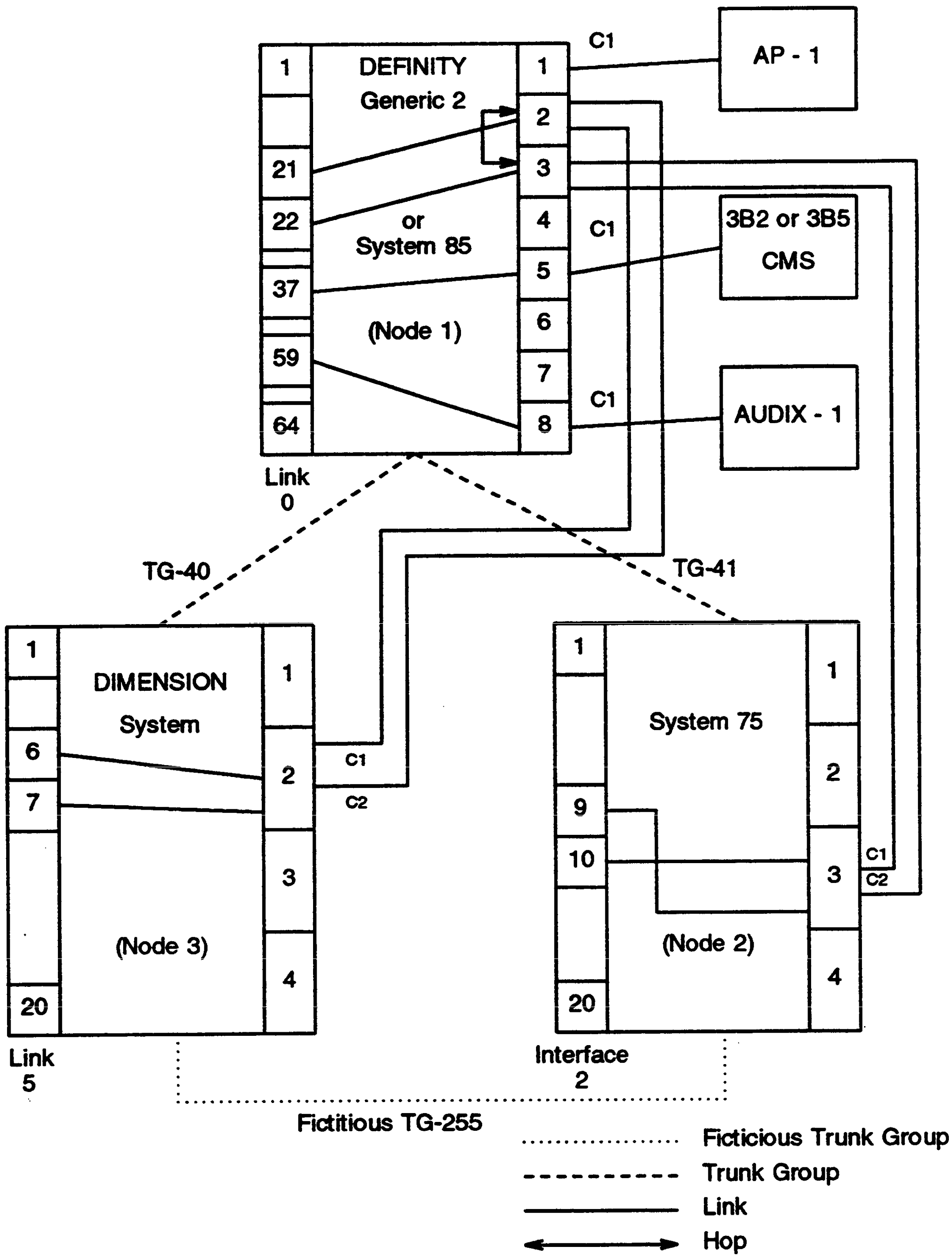
Field 7: This is a display only field. A "0" shows this is not an alternate routing network channel port, and a "1" shows alternate routing is assigned.

Field 8: This field selects the memory table values used. If the value entered is dash or "0," the displayed values are scratch pad values. If the value entered is "1," the values shown are machine values.

Field 9: This is a display only field If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Words 3 and 4

Words 3 and 4 of Procedure 257 apply to DCS link connections only and are not used for AP administration.


DEFINITY
Generic 2
or
System 85
(Node 1)
1
21
22
37
59
64
1
2
3
4
5
6
7
8
C1
AP - 1
C1
3B2 or 3B5
CMS
C1
AUDIX - 1
Link
0
TG-40
TG-41
DIMENSION
System
(Node 3)
1
6
7
20
1
2
3
4
C1
C2
System 75
(Node 2)
1
9
10
20
1
2
3
4
C1
C2
Link
5
Interface
2
Fictitious TG-255
Ficticious Trunk Group
Trunk Group
Link
Hop

Administering the AUDIX Adjunct Link

Procedure 256 — DCIU Administration — Link Characteristics
Word 1, Link Assignment

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight.

Field 2: This field shows if the link has been assigned or not. A "1" means that the link is assigned, a "2" shows the link is unassigned.

Field 3: This sets the ***Baud Rate*** or data transmission rate for this link. For the AUDIX Adjunct link, the baud rate should be set to "6" (9600 bps).

Field 4: This field identifies the transmission characteristics of the local end of the link as either DTE (Data Terminal Equipment) with encode "0," or DCE (Data Communications Equipment) with encode "1." In most cases, it doesn't matter which end of a DCIU link is DTE and which end is DCE, however, the two ends of any DCIU link must not match. That is, one end must be DTE and the other end must be DCE.

Field 5: This field identifies the link as being fixed or no dial up (encode 0), or dial up (encode 1). For most DCIU link applications, this will be fixed (encode 0).

Field 6: This field identifies the protocol to be used on the link. For DCIU applications, this is always set to "1" for the BX.25 protocol.

Field 7: This field identifies the type of machine that is located at the far end of the data path being set up. For direct linkage connections this is also the far end of the specific link. For an AUDIX Adjunct, the appropriate encode is "3." Encodes are listed in the Administration Manual, Procedures (555-105-506).

Field 8: This is the number of the processor at the far end of this link. In this case, it is AUDIX Adjunct number 1.

Field 9: This field selects the memory table value used. If a dash or "0" is entered, the values shown are scratch pad values. If "1" is entered, the values shown are the machine table (translated) values.

Field 10: This is a display only field. If a "0" is displayed, the scratch pad and machine tables do not agree (changes have been made). If a "1" is displayed, the scratch pad and machine tables agree (no change).

BX.25 Link Characteristics

Procedure 256 DCIU Administration — Link Assignment
Word 2, Level 2 Link Characteristics

Except in Field 1, typical values for link level 2 and 3 characteristics are given in the administration manual for the switch. These typical values are generally used in both Word 2 and Word 3 of Procedure 256. If these values should fail to give satisfactory service, problems must be addressed on a case-by-case basis. There are no generally recommended values other than the typical values given.

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. Typical values do not apply in this field.

Field 2: This field sets the retransmission timer.

Field 3: This field sets the idle timer.

Field 4: This field sets the maximum number of retransmission attempts that will be made.

Field 5: This field sets the maximum number of unacknowledged frames allowed before retransmission.

Field 6: This field selects the memory table values used. If the value entered is "0" or dash, the values are scratch pad values. If the value entered is "1," the values are machine values.

Field 7: This is a display only field. If a "0" is shown, the scratch pad and machine table values do not agree. If a "1" is shown, the scratch pad and machine table values agree.

Procedure 256 DCIU Administration — Link Assignment
Word 3, Level 3 Link Characteristics

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. This is a specific link associated with each application and not a typical value.

Field 2: This field sets the BX.25 activity level timer.

Field 3: This field sets the acknowledgment timer, the length of time the DCIU will wait for an acknowledgment for a transmitted message.

Field 4: This field sets the interrupt timer.

Field 5: This field sets the reset timer.

Field 6: This field sets the restart timer.

Field 7: This field shows the retransmission counter.

Field 8: This field shows the reset counter.

Field 9: This field shows the restart counter.

Field 10: This field establishes the maximum number of allowable unacknowledged packets.

Field 11: This field selects the memory table values used. If the value entered is "0" or "-," the values are scratch pad values. If the value entered is "1," the values are machine values.

Field 12: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics Word 1, Network Channels

Component A

This is the source component for a Network Channel.

Field 1: This is the link identified with Component A.

0 = local switch port
1 through 8 = remote switch.

Field 2: This identifies the logical channel for Component A. When Field 1 is set to "0," this is a local switch port. These ports have preassigned applications and must be used as assigned in all System 85, Release 2, Version 3 and earlier switches. Port "59" is preassigned for use with AUDIX Adjunct number 1. See Table H-A for a list of all applicable port assignments.

Procedure 257 — DCIU Administrtion — Data Path Characterisitcs Word 1, Network Channels (Contd)

Component B

This is the destination component for a Network Channel.

Field 3: This is the link identified with Component B.

0 = local switch port
1 through 8 = remote swtich.

Field 4: This identifies the logical channel for Component B. If Field 1 is set to "0," this is a local switch port. These ports have preassigned applications and must be used as assigned in all System 85, Release 2, Version 3 and earlier switches. See Table H-A for a list of all applicable port assignments.

Field 5: This field sets the priority of the network channel being established. A "0" sets up low priority, and a "1" sets up high priority. In the event of network channel contention, all high priority network channel messages will be served before low priority messages are served.

Field 6: This field sets the alternate muting flag. A "0" sets up fixed Permanent Virtual Circuit (PVC) routing. This type of routing requires that Component B be identified in Fields 3 and 4. A "1" establishes this as an Alternate Routing network channel. This type of routing requires that Component B is unspecified (dashed) in Fields 3 and 4.

Field 7: This field selects the memory table values used. If the value entered is "0," the values are scratch pad values. If the valued entered is "1," the values are machine values.

Field 8: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics

Word 2, Port Characteristics

Field 1: This field identifies the number of the local (switch) port.

Field 2: This field shows the port number on the remote processor. The range of possible remote port numbers depends on the type of remote processor (i.e., the range of possibilities for an AUDIX Adjunct is from 1 to 10). The applicable ranges for different types of remote processors are given in the Administration Manual.

Field 3: This field identifies the destination routing code for an alternate routing network channel. If this is not an alternate routing network channel, this field is dashed.

Field 4: This field assigns the postage value for alternate routing network channels. If this is not an alternate network channel, this field is dashed.

Field 5: This is a display only field. A value of "0" shows that the port is not assigned to a network channel. A value of "1" shows that the port is assigned to a network channel.

Field 6: This is a display only field. This field shows the assigned priority of the network channel. A "0" shows low priority, and a "1" shows a high priority channel.

Field 7: This is a display only field. A "0" shows this is not an alternate routing network channel port, and a "1" shows alternate routing is assigned.

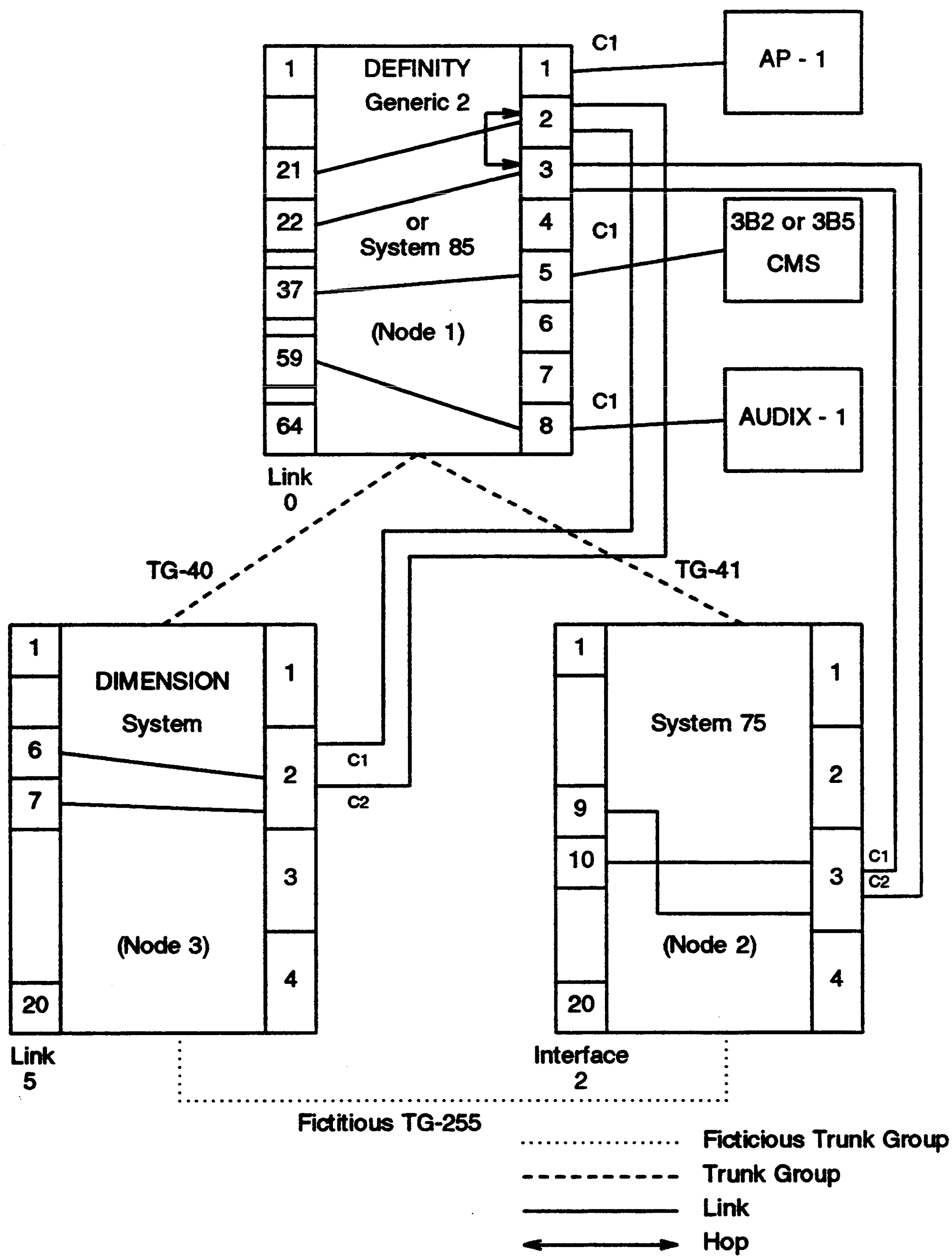
Field 8: This field selects the memory table values used. A "0" displays scratch pad values and a "1" displays machine values.

Field 9: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree (a change is being made). If a "1" is displayed, the scratch pad and machine table values agree (no change).

Words 3 and 4

Words 3 and 4 of Procedure 257 apply to DCS link connections only and are not used for AUDIX administration.

NOTES:


DEFINITY
Generic 2
or
System 85
(Node 1)
1
21
22
37
59
64
1
2
3
4
5
6
7
8
C1
AP - 1
C1
3B2 or 3B5
CMS
C1
AUDIX - 1
Link
0
TG-40
TG-41
DIMENSION
System
(Node 3)
1
6
7
20
1
2
3
4
C1
C2
System 75
(Node 2)
1
9
10
20
1
2
3
4
C1
C2
Link
5
Interface
2
Fictitious TG-255
Ficticious Trunk Group
Trunk Group
Link
Hop

Administering the CMS Adjunct Link

Procedure 256 — DCIU Administration — Link Characteristics
Word 1, Link Assignment

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight.

Field 2: This field shows if the link has been assigned or not. A "1" means that the link is assigned, a "2" shows the link is unassigned.

Field 3: This sets the ***Baud Rate*** or data transmission rate for this link. For the CMS Adjunct link, the baud rate should be set to "6" (9600 bps).

Field 4: This field identifies the transmission characteristics of the local end of the link as either DTE (Data Terminal Equipment) with encode "0," or DCE (Data Communications Equipment) with encode "1." In most cases, it doesn't matter which end of a DCIU link is DTE and which end is DCE, however, the two ends of any DCIU link must not match. That is, one end must be DTE and the other end must be DCE.

Field 5: This field identifies the link as being fixed or no dial up (encode 0), or dial up (encode 1). For most DCIU link applications, this will be fixed (encode 0).

Field 6: This field identifies the protocol to be used on the link. For DCIU applications, this is always set to "1" for the BX.25 protocol.

Field 7: This field identifies the type of machine that is located at the far end of the data path being set up. For direct linkage connections this is also the far end of the specific link.

For an ***AP 16*** CMS Adjunct, the appropriate encode is "1."

For a ***3B5*** CMS Adjunct, the appropriate encode is "2."

For a ***3B2*** CMS Adjunct, the appropriate encode is "8."

Encodes are listed in the Administration Manual, Procedures.

Field 8: This is the number of the processor at the far end of this link. In this case, it is CMS Adjunct number 1.

Field 9: This field selects the memory table value used. If a dash or "0" is entered, the values shown are scratch pad values. If "1" is entered, the values shown are the machine table (translated) values.

Field 10: This is a display only field. If a "0" is displayed, the scratch pad and machine tables do not agree (changes have been made). If a "1" is displayed, the scratch pad and machine tables agree (no change).

BX.25 Link Characteristics

Procedure 256 — DCIU Administration — Link Characteristics
Word 2, Level 2 Link Characteristics

Except in Field 1, typical values for link level 2 and 3 characteristics are given in the Administration Manual, Procedures for the switch. These typical values are generally used in both Word 2 and Word 3 of Procedure 256. If these values should fail to give satisfactory service, problems must be addressed on a case-by-case basis. There are no generally recommended values other than the typical values given.

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. Typical values do not apply in this field.

Field 2: This field sets the retransmission timer.

Field 3: This field sets the idle timer.

Field 4: This field sets the maximum number of retransmission attempts that will be made.

Field 5: This field sets the maximum number of unacknowledged frames allowed before retransmission.

Field 6: This field selects the memory table values used. If the value entered is "0" or dash, the values are scratch pad values. If the value entered is "1," the values are machine values.

Field 7: This is a display only field. If a "0" is shown, the scratch pad and machine table values do not agree. If a "1" is shown, the scratch pad and machine table values agree.

Procedure 256 — DCIU Administration — Link Assignment
Word 3, Level 3 Link Characteristics

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. This is a specific link associated with each application and not a typical value.

Field 2: This field sets the BX.25 activity level timer.

Field 3: This field sets the acknowledgment timer, the length of time the DCIU will wait for an acknowledgment for a transmitted message.

Field 4: This field sets the interrupt timer.

Field 5: This field sets the reset timer.

Field 6: This field sets the restart timer.

Field 7: This field shows the retransmission counter.

Field 8: This field shows the reset counter.

Field 9: This field shows the restart counter.

Field 10: This field establishes the maximum number of allowable unacknowledged packets.

Field 11: This field selects the memory table values used. If the value entered is "0" or "-," the values are scratch pad values. If the value entered is "1," the values are machine values.

Field 12: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics
Word 1, Network Channels

Component A

This is the source component for a Network Channel.

Field 1: This is the link identified with Component A.

0 = local switch port
1 through 8 = remote switch

Field 2: This identifies the logical channel for Component A. When Field 1 is set to "0," this is a local switch port. These ports have preassigned applications and must be used as assigned in all System 85, Release 2, Version 3 and earlier switches. Port "37" can be used assigned for CMS Adjunct number 1. See Table H-A for a list of all applicable port assignments.

Procedure 257 — DCIU Administration — Data Path Characteristics
Word 1, Network Channels (Contd)

Component B

This is the destination component for a Network Channel.

Field 3: This is the link identified with Component B.

0 = local switch port
1 through 8 = remote switch.

Field 4: This identifies the logical channel for Component B. If Field 1 is set to "0," this is a local switch port. These ports have preassigned applications and must be used as assigned in all System 85, Release 2, Version 3 and earlier switches. See Table H-A for a list of all applicable port assignments.

Field 5: This field sets the priority of the network channel being established. A "0" sets up low priority, and a "1" sets up high priority. In the event of network channel contention, all high priority network channel messages will be served before low priority messages are served.

Field 6: This field sets the alternate routing flag. A "0" sets up fixed (Permanent Virtual Circuit) routing. This type of routing requires that Component B be identified in Fields 3 and 4. A "1" establishes this as an Alternate Routing network channel. This type of routing requires that Component B is unspecified (dashed) in Fields 3 and 4.

Field 7: This field selects the memory table values used. If the value entered is "0," the values are scratch pad values. If the valued entered is "1," the valued are machine values.

Field 8: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 DCIU Administration — Data Path Characteristics

Word 2, Port Characteristics

Field 1: This field identifies the number of the local (switch) port.

Field 2: This field shows the port number on the remote processor. The range of possible remote port numbers depends on the type of remote processor The allowed entries for:

An AP 16 CMS Adjunct is from "1" to "11"

A 3B2 or 3B5 CMS Adjunct is "1."

The applicable ranges for different types of remote processors are given in the Administration Manual, Procedures.

Field 3: This field identifies the destination routing code for an alternate routing network channel. If this is not an alternate routing network channel, this field is dashed.

Field 4: This field assigns the postage value for alternate routing network channels. If this is not an alternate network channel, this field is dashed.

Field 5: This is a display only field. A value of "0" shows that the port is not assigned to a network channel. A value of "1" shows that the port is assigned to a network channel.

Field 6: This is a display only field. This field shows the assigned priority of the network channel. A "0" shows low priority, and a "1" shows a high priority channel.

Field 7: This is a display only field. A "0" shows this is not an alternate routing network channel port, and a "1" shows alternate routing is assigned.

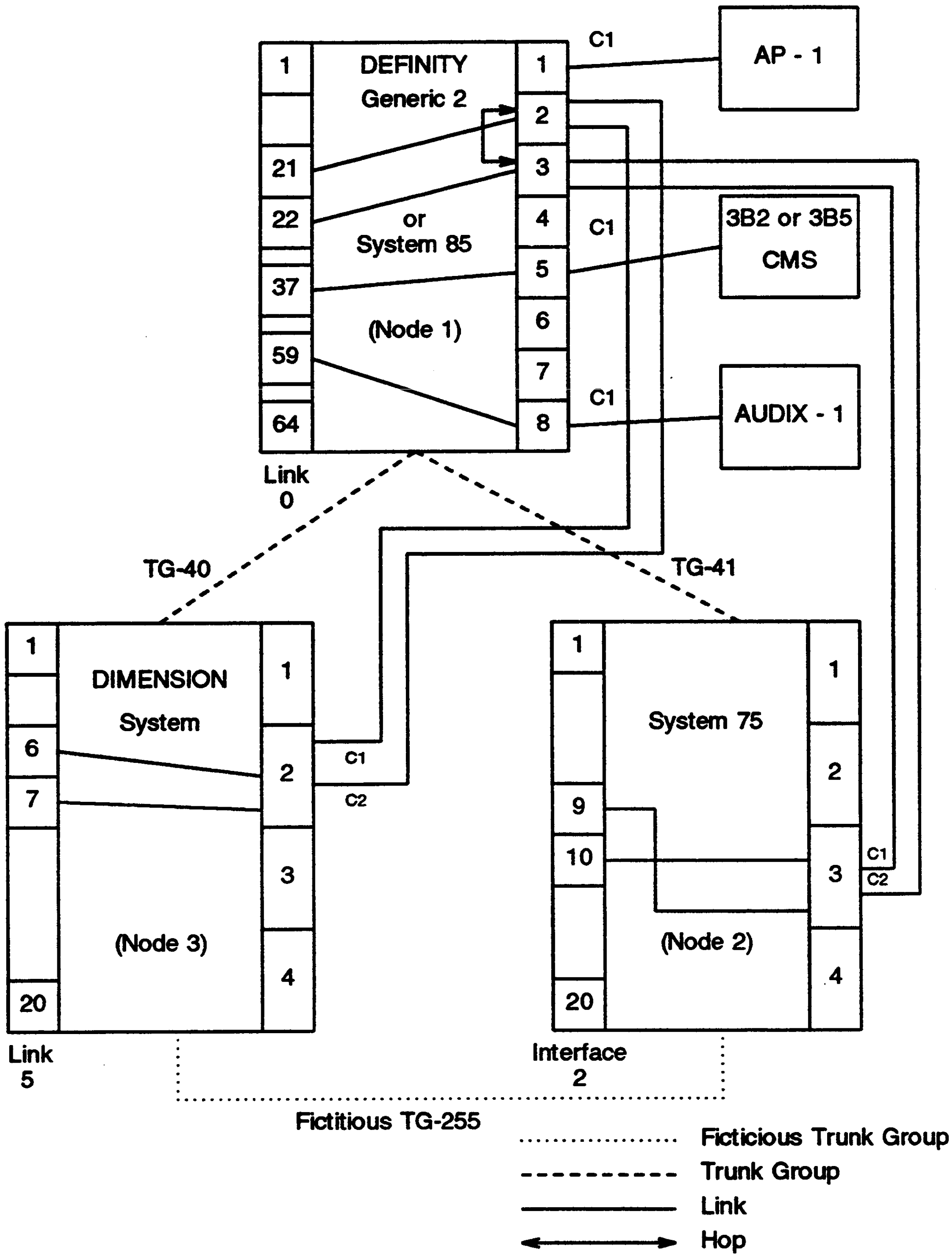
Field 8: This field selects the memory table values used A "0" displays scratch pad values and a "1" displays machine values.

Field 9: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree (a change is being made). If a "1" is displayed, the scratch pad and machine table values agree (no change).

Words 3 and 4

Words 3 and 4 of Procedure 257 apply to DCS link connections only and are not used for CMS administration.

NOTES:


DEFINITY
Generic 2
or
System 85
(Node 1)
1
21
22
37
59
64
1
2
3
4
5
6
7
8
C1
C1
C1
AP - 1
3B2 or 3B5
CMS
AUDIX - 1
Link
0
TG-40
TG-41
DIMENSION
System
(Node 3)
1
6
7
20
1
2
3
4
C1
C2
Link
5
System 75
(Node 2)
1
9
10
20
1
2
3
4
C1
C2
Interface
2
Fictitious TG-255
Ficticious Trunk Group
Trunk Group
Link
Hop

Administering the DCIU Link to an Enhanced "DIMENSION" System

Procedure 256 — DCIU Administration — Link Characteristics Word 1, Link Assignment

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight.

Field 2: This field shows if the link has been assigned or not. A "1" means that the link is assigned, a "2" shows the link is unassigned.

Field 3: This sets the ***Baud Rate*** or data transmission rate for this link. For DCS applications, the baud rate should be set to "6" (9600 bps).

Field 4: This field identifies the transmission characteristics of the local end of the link as either DTE (Data Terminal Equipment) with encode "0," or DCE (Data Communications Equipment) with encode "1." In most cases, it doesn't matter which end of a DCIU link is DTE and which end is DCE. However, the two ends of any DCIU link must not match. That is, one end must be DTE and the other end must be DCE.

Field 5: This field identifies the link as being fixed or no dial up (encode 0), or dial up (encode 1). For most DCIU link application, this will be fixed (encode 0).

Field 6: This field identifies the protocol to be used on the link. For DCS applications, this is always set to "1" for the BX.25 protocol.

Field 7: This field identifies the type of machine that is located at the distant end of the data path being set up. For direct linkage connections, this is also the far end of the specific link. For an Enhanced DIMENSION System switch, the appropriate encode is "7." Encodes are listed in the Administration Manual for the switch being administered.

Field 8: This is the DCS node number of the machine at the distant end of this link.

Field 9: This field selects the memory table values used If a "-" or "0" is entered, the values are scratch pad values. If a "1" is entered, the machine table (translated) values are used.

Field 10: This is a display only field. If a "0" is displayed, the scratch pad and machine tables do not agree (changes have been made). If a "1" is displayed, the scratch pad and machine tables agree (no change).

BX.25 Link Characteristics

Procedure 256 — DCIU Administration — Link Assignment
Word 2, Level 2 Link Characteristics

Except in Field 1, typical values for link level 2 and 3 characteristics are given in the Administration Manual, Procedures for the switch. These typical values are generally used in both Word 2 and Word 3 of Procedure 256. If these should fail to give satisfactory service, problems must be addressed on a case-by-case basis. There are no generally recommended values other than the typical values given.

Field 1: This is the link number of the DCIU(external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. Default values do not apply in this field.

Field 2: This field sets the retransmission timer.

Field 3: This field sets the idle timer.

Field 4: This field sets the maximum number of retransmission attempts that will be made.

Field 5: This field sets the maximum number of unacknowledged frames allowed before retransmission.

Field 6: This field selects the memory table values used. If the value entered is "0," the values are scratch pad values. If the value entered is "1," the values are machine values.

Field 7: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 256 — DCIU Administration — Link Assignment
Word 3, Level 3 Link Characteristics

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. This is not a default value.

Field 2: This field sets the BX.25 activity level timer.

Field 3: This field sets the acknowledgment timer, the length of time the DCIU will wait for an acknowledgment for a transmitted message.

Field 4: This field sets the interrupt timer.

Field 5: This field sets the reset timer.

Field 6: This field sets the restart timer.

Field 7: This field shows the retransmission counter.

Field 8: This field shows the reset counter.

Field 9: This field shows the restart counter.

Field 10: This field establishes the maximum number of allowable unacknowledged packets.

Field 11: This field selects the memory table values used. If a "0" is entered, scratch pad values are used. If a "1" is entered, machine values are used.

Field 12: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics Word 1, Network Channels

Component A

This is the source component for a Network Channel.

Field 1: This is the link identified with Component A.

0 = local switch port
1 through 8 = remote switch

Field 2: This identifies the logical channel for Component A. If Field 1 is set to "0," this is a local switch port. These ports have preassigned applications and must be used as assigned in all System 85, Release 2, Version 3 and earlier switches. Port "21" is one of the ports that can be used for DCS applications. See Table H-A for a list of all applicable port assignments.

Procedure 257 — DCIU Administration — Data Path Characteristics Word 1, Network Channels (Contd)

Component B

This is the destination component for a Network Channel.

Field 3: This is the link identified with Component B.

0 = local switch port
1 through 8 = remote switch.

Field 4: This identifies the logical channel for Component B. If Component B is from a distant switch, the range for Field 4 depends on the type switch. For a System 85, Release 2 switch the range is 1 to 64. If the distant switch is an Enhanced DIMENSION System (encode 7) or a System 75 (encode 4), the range is 1 to 20. Logical channels on DCIU links are not preassigned.

Field 5: This field sets the priority of the network channel being established. A "0" sets low priority, and a "1" sets high priority. In the event of network channel contention, all high priority network channel messages will be served before low priority messages are served.

Field 6: This field sets the alternate routing flag. A "0" sets fixed (Permanent Virtual Circuit) routing. This type of routing requires that Component B be identified in Fields 3 and 4. A "1" sets this as an Alternate Routing network channel. This type of routing requires that Component B is unspecified (dashed) in Fields 3 and 4.

Field 7: This field selects the memory table values used. If the value entered is "0," the values are scratch pad values. If a "1" is entered, machine values are used.

Field 8: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics Word 2, Port Characteristics

Field 1: This field identifies the number of the local (switch) port.

Field 2: This field shows the port number on the remote processor. The range of possible remote port numbers depends on the type of remote processor (i.e., the range of possibilities for an Enhanced DIMENSION System switch is from 1 to 20). The applicable ranges for different types of remote processors are given in the **DEFINITY Generic 2, Administration Manual, Procedures** (555-105-506).

Field 3: This field identifies the destination muting code for an alternate routing network channel. If this is not an alternate routing network channel, this field is dashed.

Field 4: This field assigns the postage value for alternate routing network channels. If this is not an alternate network channel, this field is dashed.

Field 5: This is a display only field. A value of "0" shows that the port is not assigned to a network channel. A value of "1" shows that the port is assigned to a network channel.

Field 6: This is a display only field. This field shows the assigned priority of the network channel. A "0" shows low priority, and a "1" shows a high priority channel.

Field 7: This is a display only field. A "0" shows this is not an alternate routing network channel port, and a "1" shows alternate routing is assigned.

Field 8: This field selects the memory table values used. If the value entered is "0," the values are scratch pad values. If a "1" is entered, the values are machine values.

Field 9: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics
Word 3, Trunk Group & DCS Node Assignment

Field 1: This field is used to designate the number of the local (switch) port being associated with a trunk group and DCS remote node.

Field 2: This field identifies the trunk group (first choice) associated with traffic from the local switch to a specific DCS remote node.

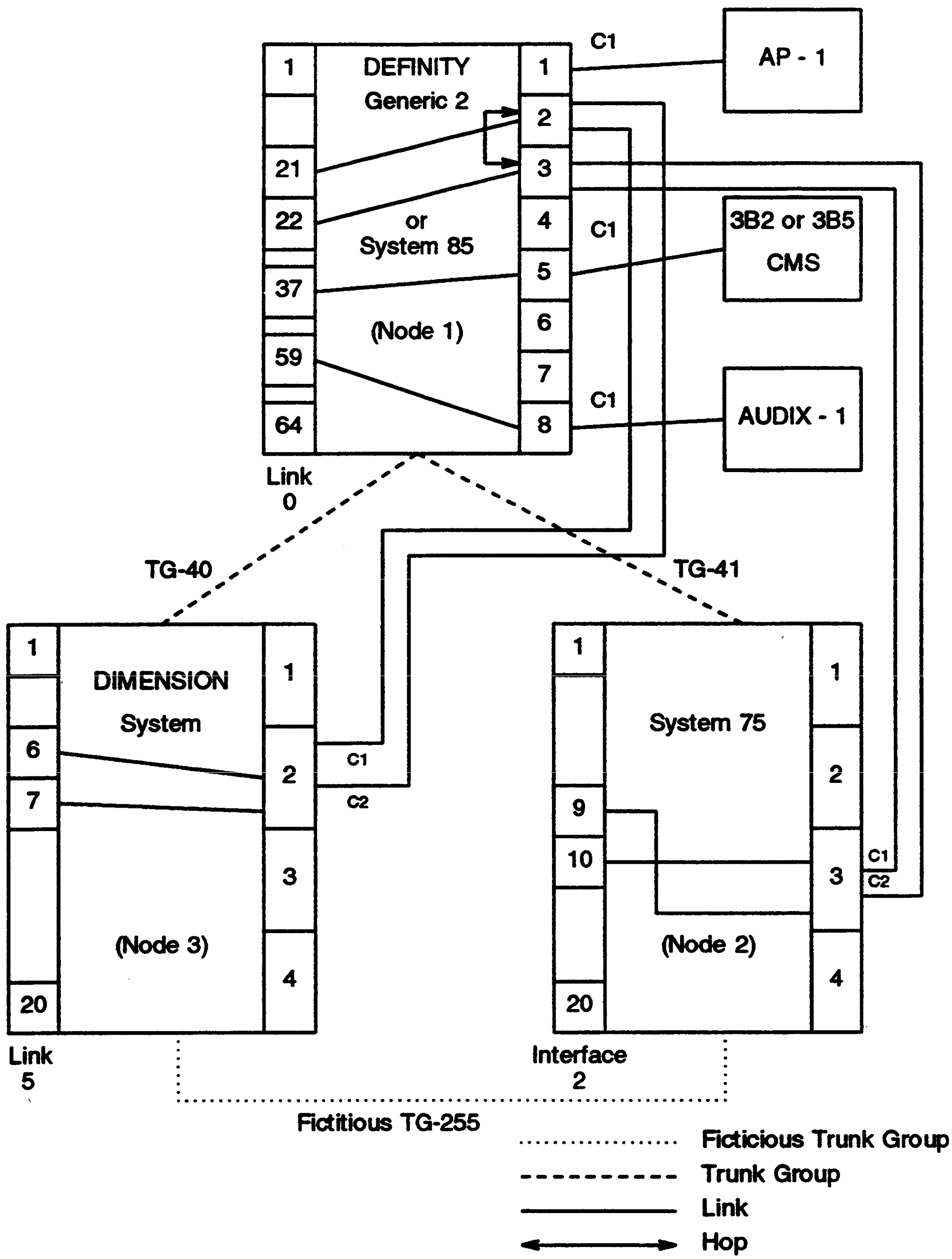
Field 3: This is the number of the DCS remote node associated with traffic served by the port in Field 1 and the trunk group in Field 2.

Field 4: This is a display only field. If the value displayed is "0," the displayed values are scratch pad values. If the display shows "1," the values shown are machine values.

Field 5: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Word 4, Alternate Routing

Word 4 of Procedure 257 is used only for Alternate Routing. As this example does not use alternate routing, Word 4 does not apply.


1
DEFINITY
Generic 2
1
2
21
3
22
or
System 85
4
C1
3B2 or 3B5
CMS
37
5
6
(Node 1)
59
7
C1
AUDIX - 1
64
8
C1
AP - 1
Link
0
TG-40
TG-41
1
DIMENSION
System
1
6
2
C1
C2
7
3
(Node 3)
4
20
1
System 75
1
2
9
10
3
C1
C2
(Node 2)
4
20
Link
5
Interface
2
Fictitious TG-255
Ficticious Trunk Group
Trunk Group
Link
Hop

Administering the DCIU Link to a System 75 Switch

Procedure 256 — DCIU Administration — Link Characteristics Word 1, Link Assignment

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight.

Field 2: This field shows if the link has been assigned or not. A "1" means that the link is assigned, a "2" shows the link is unassigned.

Field 3: This sets the ***Baud Rate*** or data transmission rate for this link. For DCS applications, the baud rate should be set to "6" (9600 bps).

Field 4: This field identifies the transmission characteristics of the local end of the link as either DTE (Data Terminal Equipment) with encode "0," or DCE (Data Communications Equipment) with encode "1." In most cases, it doesn't matter which end of a DCIU link is DTE and which end is DCE. However, the two ends of any DCIU link must not match. That is, one end must be DTE and the other end must be DCE.

Field 5: This field identifies the link as being fixed or no dial up (encode 0), or dial up (encode 1). For most DCIU link application, this will be fixed (encode 0).

Field 6: ‘This field identifies the protocol to be used on the link. For DCS applications this is always set to "1" for the BX.25 protocol.

Field 7: This field identifies the type of machine that is located at the distant end of the data path being set up. For direct linkage connections this is also the far end of the specific link. For a System 75 switch, the appropriate encode is "4." Encodes are listed in the Feature Translations Manual for the switch being administered.

Field 8: This is the DCS node number of the machine at the distant end of this link.

Field 9: This field selects the memory table values used. If a dash or "0 is entered, the values are scratch pad values. If a "1" is entered, the values are the machine table (translated) values.

Field 10: This is a display only field. If a "0" is displayed, the scratch pad and machine tables do not agree (changes have been made). If a "1" is displayed, the scratch pad and machine tables agree (no change).

BX.25 Link Characteristics

Procedure 256 — DCIU Administration — Link Assignment Word 2, Level 2, Link Characteristics

Except in Field 1, typical values for link level 2 and 3 characteristics are given in the **DEFINITY Generic 2, Administration Manual, Procedures** (555-105-506). These typical values are generally used in both Word 2 and Word 3 of Procedure 256. If these should fail to give satisfactory service, problems must be addressed on a case-by-case basis. There are no generally recommended values other than the typical values given.

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Work 1. Default values do not apply in this field.

Field 2: This field sets the retransmission timer.

Field 3: This field sets the idle timer.

Field 4: This field sets the maximum number of retransmission attempts that will be made.

Field 5: This field sets the maximum number of unacknowledged frames allowed before retransmission.

Field 6: This field selects the memory table values used. If the value entered is "0," the values are scratch pad values. If a "1" is entered, machine values are used.

Field 7: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 256 — DCIU Administration — Link Assignment
Word 3, Level 3 Link Characteristics

Field 1: This is the link number of the DCIU (external) link being assigned. The range is from one to eight. This should be the same as used in Word 1. This is not a default value.

Field 2: This field sets the BX.25 activity level timer.

Field 3: This field sets the acknowledgment timer, the length of time the DCIU will wait for an acknowledgment for a transmitted message.

Field 4: This field sets the interrupt timer.

Field 5: This field sets the reset timer.

Field 6: This field sets the restart timer.

Field 7: This field shows the retransmission counter.

Field 8: This field shows the reset counter.

Field 9: This field shows the restart counter.

Field 10: This field establishes the maximum number of allowable unacknowledged packets.

Field 11: This field selects the memory table values used. If the value entered is "0" the values are scratch pad values. If the value entered is "1," the values are machine values.

Field 12: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Link Assignment
Word 1, Network Channels

Component A

This is the source component for a Network Channel.

Field 1: This is the link identified with Component A.

0 = local switch port
1 through 8 = remote switch

Field 2: This identifies the logical channel for Component A. When Field 1 is "0," this is a local switch port. These ports have preassigned applications and must be used as assigned in all System 85, Release 2, Version 3 and earlier switches. Port "22" is one of ports that can be used for DCS applications. See Table H-A for a list of all applicable port assignments.

Procedure 257 — DCIU Administration — Data Path Characteristics
Word 1, Network Channels (Contd)

Component B

This is the destination component for a Network Channel.

Field 3: This is the link identified with Component B.

0 = local switch port
1 through 8 = remote switch

Field 4: This identifies the logical channel for Component B. When Component B is from a distant switch, the range for Field 4 depends on the type of switch. For 2 System 85, the range is from 1 to 64. For an Enhanced DIMENSION System (encode 7), the range is from 1 to 20. Logical channels on DCIU links do not have preassigned functions.

Field 5: This field sets the priority of the network channel being established. A "0" sets up low priority, and a "1" sets up high priority. In the event of network channel contention, all high priority network channel messages will be served before low priority messages are served.

Field 6: This field sets the alternate routing flag. A "0" establishes fixed (Permanent Virtual Circuit) routing. This type of routing requires that Component B be identified in Fields 3 and 4. A "1" establishes this as an Alternate Routing network channel. This type of routing requires that Component B is unspecified (dashed) in Fields 3 and 4.

Field 7: This field selects the memory table values used. If the value entered is "0," the values are scratch pad values. If a "1" is entered, the values are machine values.

Field 8: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics Word 2, Port Characteristics

Field 1: This field identifies the number of the local (switch) port.

Field 2: This field shows the port number on the remote processor. The range of possible remote port numbers depends on the type of remote processor (i.e., the range of possibilities for an Enhanced DIMENSION switch is from 1 to 20). The applicable ranges for different types of remote processors are given in the Feature Translations Service Manual.

Field 3: This field identifies the destination routing code for an alternate routing network channel. If this is not an alternate routing network channel, this field is dashed.

Field 4: This field assigns the postage value for alternate routing network channels. If this is not an alternate network channel, this field is dashed.

Field 5: This is a display only field. A value of "0" shows that the port is not assigned to a network channel. A value of "1" shows that the port is assigned to a network channel.

Field 6: This is a display only field. This field shows the assigned priority of the network channel. A "0" shows low priority, and a "1" shows a high priority channel.

Field 7: This is a display only field. A "0" shows this is not an alternate routing network channel port, and a "1" shows alternate routing is assigned.

Field 8: This field selects the memory table values used. If the value entered is "0," the values are scratch pad values. If the value entered is "1," the values are machine values.

Field 9: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Procedure 257 — DCIU Administration — Data Path Characteristics
Word 3, Trunk Group & DCS Node Assignment

Field 1: This field is used to designate the number of the local (switch) port being associated with a trunk group and DCS remote node.

Field 2: This field identifies the trunk group (first choice) associated with traffic from the local switch to a specific DCS remote node.

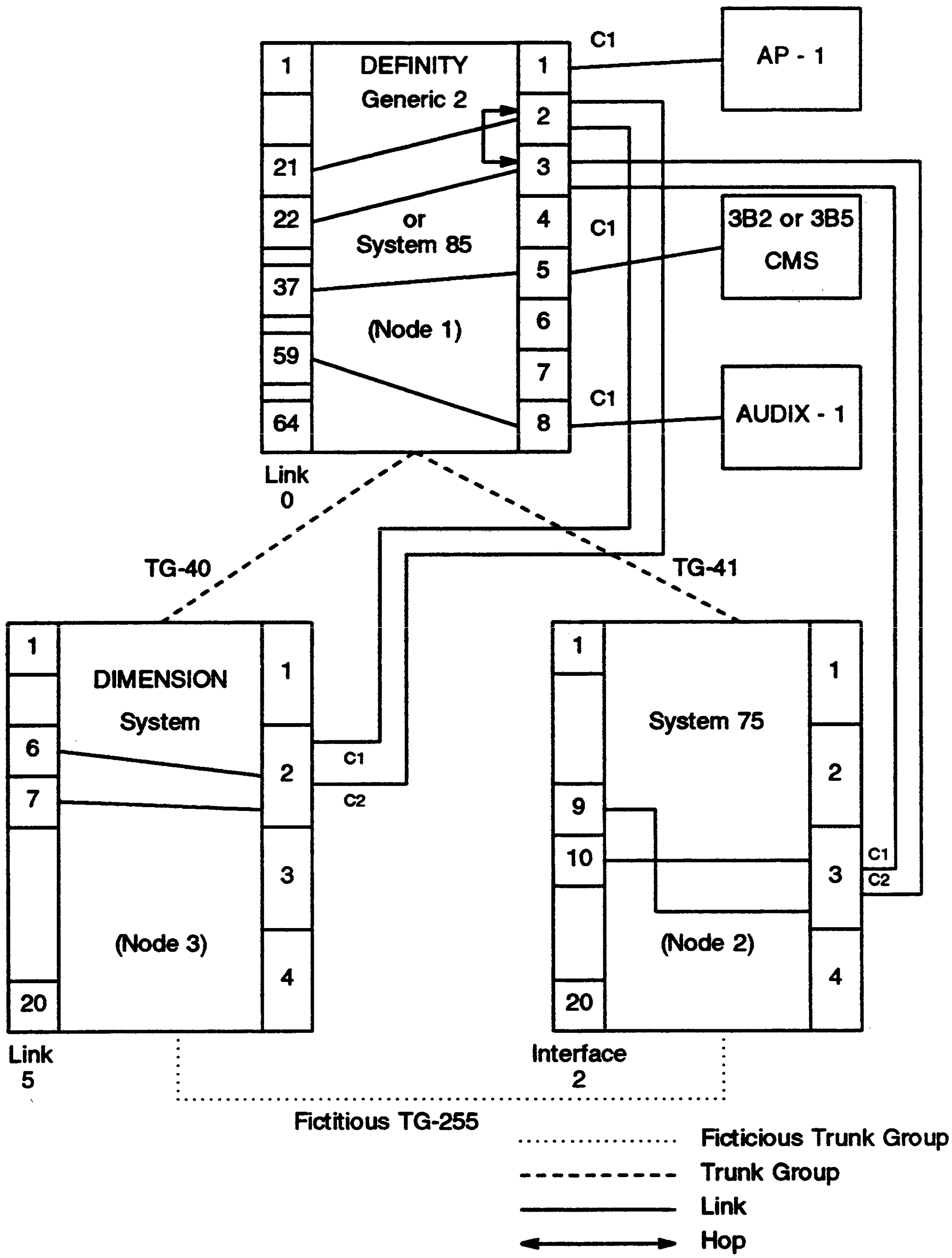
Field 3: This is the number of the DCS remote node associated with traffic served by the port in Field and the trunk group in Field 2.

Field 4: This field selects the memory table values used. If a "0" is entered, scratch pad values are used. If a "1" is entered, machine values are used.

Field 5: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

Word 4, Alternate Routing

Word 4 of Procedure 257 is only used for Alternate Routing. The example given does not use alternate routing, therefore Word 4 does not apply.


DEFINITY
Generic 2
or
System 85
(Node 1)
1
21
22
37
59
64
1
2
3
4
5
6
7
8
C1
AP - 1
C1
3B2 or 3B5
CMS
C1
AUDIX - 1
Link
0
TG-40
TG-41
DIMENSION
System
(Node 3)
1
6
7
20
1
2
3
4
C1
C2
Link
5
System 75
(Node 2)
1
9
10
20
1
2
3
4
C1
C2
Interface
2
Fictitious TG-255
Ficticious Trunk Group
Trunk Group
Link
Hop

Administering the Network Channel for a Hop

This administrative requirement is simpler than it at first appears. The links that are needed from node 1 have already been set up. These are the same links that were administered as direct connections from node 1 to node 2, and from node 1 to node 3.

The only administration still needed at node 1 is to set up the network channel that will connect logical channel 2 (C2) from node 2 to logical channel 2 (C1) from node 3 (shown in the example as a dashed line within Node 1).

Procedure 257—DCIU Administration—Data Path Characteristics Word 1, Network Channels

Component A

This is the source component for a Network Channel.

Field 1: This is the link identified with Component A.

0 = local switch port
1 through 8 = remote switch

Field 2: This identifies the logical channel for Component A. When component A is from a distant switch (link 1 through 8), the range depends on the type of switch at the far end. When Component A is an Enhanced DIMENSION System (encode 7), the range is 1 to 20. Logical channels on DCIU links are not preassigned.

Procedure 257—DCIU Administration—Data Path Characteristics Word 1, Network Channels (Contd)

Component B

This is the destination component for a Network Channel.

Field 3: This is the link identified with Component B.

0 = local switch port
1 through 8 = remote switch

Field 4: This identifies the logical channel for Component B. When Component B is from a distant switch, the range for Field 4 depends on the type of switch involved When the other end of the link is an Enhanced DIMENSION System (encode 7), the range is from 1 to 20. Logical channels on DCIU links are not preassigned.

Field 5: This field sets the priority of the network channel being established. A "0" sets up low priority, and a "1" sets up high priority. In the event of network channel contention, all high priority network channel messages will be served before low priority messages are served.

Field 6: This field sets the alternate routing flag. A "0" sets up fixed (Permanent Virtual Circuit) routing. This type of routing requires that Component B be identified in Fields 3 and 4. A "1" establishes this as an Alternate Routing network channel. This type of routing requires that Component B is unspecified (dashed) in Fields 3 and 4.

Field 7: This field selects the memory table values used If the field value is "0," the values are scratch pad values. If the field value is "1," the values are machine values.

Field 8: This is a display only field. If a "0" is displayed, the scratch pad and machine table values do not agree. If a "1" is displayed, the scratch pad and machine table values agree.

The Link Characteristics have already been established for both DCIU links. If they needed to be changed for some reason we would have to use different links because the same link cannot be assigned separate characteristics for different logical channels. The Data Path Characteristics (Procedure 257, Words 2 through 4) are setup at the originating switches (node 2 and 3) and are not reset at the node performing a hop.

Flexible DCIU Administration

In Release 2, Version 3 and earlier System 85 switches, the uses for DCIU ports are fixed. That is, each DCIU port can be used for only one application. Each DCIU port is reserved for a specific application in the factory before the system is shipped and cannot be used for any other purpose. The same ports on all DCIUs must be used for the same purpose.

For example, on a System 85, Release 2, Version 3, switch, DCIU port number 11 can only be used for Leave Word Calling on AP number 2. DCIU port number 11 has been permanently reserved for the Leave Word Calling feature on AP 2. If there is no AP number 2 on a given switch, or if AP 2 does not have a leave word calling function, this DCIU port can not be used.

Flexible DCIU Port Reservations

Port Reservation Versus Port Assignment

Flexible DCIU port reservations was introduced with System 85, Release 2, Version 4, and is continued in the DEFINITY Generic 2 switch. Flexible DCIU port reservations presents a degree of flexibility and a new administrative requirement not previously done by users or service support personal. With flexible port reservations, if there is no AP 2, ports 9 through 13 (those ports designated for AP 2) can be used for another purpose if needed. This flexibility is further enhanced by the shared use of adjuncts in a DCS.

Assignment

Assignment is done at the customer facility. Assignment amounts to ***turning on*** the function for which a DCIU port has been reserved. Assignment is done through system administration. The assignment of an application to a DCIU port is limited to the application for which that port has been reserved. This means that in Release 2, Version 3 and earlier systems, port assignment is limited by the ***"fixed reservations"*** applied at the factory. In Release 2, Version 4 and DEFINITY Generic 2 switches, port assignment is made flexible by the ***"flexible port reservation"*** system.

Flexible Port Reservations

In System 85, Release 2, Version 4, and DEFINITY Generic 2, the fixed reservation of DCIU ports is replaced by flexible DCIU port reservation. This allows a port that is not needed for a particular function to be used for a different purpose. Flexible port reservation also provides for the use of a new port type, the ES (Enhanced Services) port. The ES port is used for Centralized Messaging service. With ES ports, DCIU links can be provided not only from a switch processor to local adjunct processors but also to remote adjunct processors.

Flexible Port Assignment

With flexible DCIU port reservations, the assignment of DCIU ports is also flexible. However, once the port is *"reserved"* for a particular application, it must either be assigned to that application, left unassigned, or the reservation must be changed before a different assignment can be made. Port assignment flexibility is based on the DCIU port reservation process rather than the DCIU port assignment process.

Reserved or Unreserved Port Status

Because DCIU ports no longer have standard reservations, some new and different terms and concepts are needed. The "reserved" or "unreserved" status of a port effects administration. DCIU ports on System 85, Release 2, Version 4, and DEFINITY Generic 2 switches can be ***reserved at the factory*** before the switch is shipped, just as with Version 3 and earlier switches. This is done based on the TRACS (Translations Recovery Additions Conversions System) input. However, ports do not default to standard reservations. If not pre-reserved, the DCIU ports will be **unreserved.** Separate administration is then required to set up DCIU port reservations. Whether or not ports are pre-reserved, separate administration is still required to assign (or turn on) each DCIU port to its reserved application.

Standard Reservations and Assignments for APs

For local APs, DCIU port reservations and assignments are still fixed. If APs are to be used on a particular switch, standard DCIU port reservations and assignments must be used. However, these standards are not assigned by default. Standard reservations and assignments must be administered just like any other port reservation and assignment. If a local AP is not required, these ports can be reserved for other uses, but other ports cannot be reserved for use by a local AP. The standard ports for local APs are:

- Ports 1 through 8 for AP 1
- Ports 9 through 13 for AP 2
- Ports 14 through 18 for AP 3
- Port 19, and ports 30 through 33 for AP 4
- Ports 37 through 41 for AP 5.

The flexible DCIU port reservations enhancement is not readily apparent to the user. However, it is basic to the function of other enhancements including the IACC (Integrated Adjunct Call Control) message set and Centralized Messaging service.

With flexible DCIU port reservations and the new ES port type, changes are needed in the recommended port uses. Some **recommended port reservations** are shown in Table H-B. These are recommended port reservations and except for port "0" are not automatic or default reservations.

These changes apply **to Release 2, Version 4 switches** and to the DEFINITY Generic 2 switch. They are mandatory only for the standard local AP applications.

TABLE H-B. Flexible DCIU Recommended Port Reservations

PORT	APPL	PORT	APPL	PORT	APPL	PORT	APPL
0	ILCC	16	LWCH 3	32	LWCL 4	48	DCS 19
1	MCS 1	17	LWCL 3	33	AMWL 4	49	ES 1
2	LWCH 1	18	AMWL 3	34	DCS 10	50	ES 2
3	LWCL 1	19	CLK 4	35	DCS11	51	ES 3
4	AMWL 1	20	TEST 2	36	DCS 12	52	ES 4
5	TRAF	21	DCS 1	37	CLK 5	53	ES 5
6	TEST 1	22	DCS 2	38	MCS 5	54	ES 6
7	SMDR	23	DCS 3	39	LWCH 5	55	ES 7
8	CLK 1	24	DCS 4	40	LWCL 5	56	ES 8
9	CLK 2	25	DCS 5	41	AMWL 5	57	ES 9
10	MCS 2	26	DCS 6	42	DCS 13	58	ES 10
11	LWCH 2	27	DCS 7	43	DCS 14	59	ES 11
12	LWCL 2	28	DCS 8	44	DCS 15	60	ES 12
13	AMWL 2	29	DCS 9	45	DCS 16	61	ES 13
14	CLK 3	30	MCS 4	46	DCS 17	62	ES 14
15	MCS 3	31	LWCH 4	47	DCS 18	63	ES 15
						64	ES 16

LEGEND:

ILCC= Interlevel Communications Channel (Local switch Port)
MCS = Message Center Service
LWCH= Leave Word Calling High Priority
LWCL= Leave Word Calling Low Priority
AMWL= Automatic Message Waiting Lamp
TRAF=Traffic
SMDR= Station Message Detail Recording
CLK = AP Clock Synchronization
DCS = Distributed Communications System
TEST= DIP/DCIU Test Port
ES = Enhanced Services Port

Numbers on the right of the APPL column show the instance number of the application. For example, MCS 1 refers to the Message Center Service application on AP number 1.

Appendix I: Compatibility Matrix

The following tables provide an abbreviated listing of features and adjuncts that have been added or enhanced since the introduction of the System 85, Release 2 switch. It shows their compatibility with different versions of the System 85, Release 2 and DEFINITY Generic 2 switches. It also shows some relationships between adjunct releases and features and hardware items, such as terminals and interfaces.

Feature	System 85 Release and Version				DEFINITY	Manager IV		3B5 AP Release		
	R2 V1	R2 V2	R2 V3	R2 V4	G2	SM-1	SM-2	R1	R2	R3
ACCUNET Interface	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
AAR/ARS Enhancements:										
640 AAR Patterns	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
16 Preferences	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
160 ARS 6 Digit Translators	No	No	Yes	Yes	Yes	Yes	Yea	—	—	—
10 Patterns Per Traslator	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
Pattern Queuing	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
IXC Access	No	No	Yes	Yes	Yes	Yes	Yea	—	—	—
Generalized Routing	No	No	Yes	Yes	Yes	Yes	Yes	-	-	-
ATMS	No	Yes	Yes	Yes	Yes	Yes	Yes	—	—	—
AUDIX										
Basic	No	Yes	Yes	Yes	Yes	Yes	Yes	—	—	—
Coverage	No	Yes	Yes	Yes	Yes	—	—	—	—	—
Call Transfer Out	No	No	No	Yes	Yes	—	—	—	—	—
CDR Enhancements										
Multiple LSUs	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
Variable Call Completion Threshold Timing	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
90,000 Auth Codes	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
Account Codes:										
15-digit Codes	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
Forced Entry	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
Processor Communication Circuit	No	No	No	Yes	No	Yes	Yes	—	—	—
3B2 LSUs	No	No	No	Yes	Yes	Yes	Yes	—	—	—

Feature	System 85 Release and Version				DEFINITY	Manager IV		3B5 AP Release		
	R2 V1	R2 V2	R2 V3	R2 V4	G2	SM-1	SM-2	R1	R2	R3
Data Interfaces:										
DCP Data Module	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
DCP (to AP)	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
BRI Data Module*	No	No	No	No	Yes	—	—	—	—	—
EIA (4-Port Board)	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
3270 Conversion	*	*	Yes	Yes	Yes	Yes	Yes	—	—	—
ISN Interface	No	†	Yes	Yes	Yes	Yes	Yes	—	—	—
Default Terminal Dialing	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
DS1 (Tie Trunks)	Yes	Yes	Yes	Yes	Yes	Yes	Yes	—	—	—
Dual Path coverage (Call Coverage)	No	Yes	Yes	Yes	Yes	Yes	Yes	—	—	—
Unrestricted 5-Digit Dialing	No	No	Yes	Yes	Yes	Yes	Yes	No	Yes	Yes
Extension Number Portability	No	No	Yes	Yes	Yes	Yes	Yes	No	Yes	Yes
LWC w/out AP in a DCS Environment	No	Yes	Yes	Yes	Yes	Yes		—	—	—
Paging With Music Option	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—
Remote Port Carrier	No	†	Yes	Yes	Yes	Yes	Yes	—	—	—
Remote Module with Fiber Optic Connections	Yes	Yes	Yes	Yes	Yes	Yes	Yes	—	—	—
Symbolic Dialing	No	No	Yes	Yes	Yes	Yes	Yes	—	—	—

* Available as a line appearance only, thus not recordable on CDR.
† Available but with limited or reduced functionality.

Feature	System 85 Release and Version				DEFINITY	Manager IV		365 AP Release		
	R2 V1	R2 V2	R2 V3	R2 V4	G2	SM-1	SM-2	R1	R2	R3
Terminals:										
VDS (7404D)	†	Yes	Yes	Yes	Yes	Yes	Yes	Yes	**	**
IDT (7407D)	†	Yes	Yes	Yes	Yes	Yes	**	**	**	
7406D's	†	†	†	Yes	Yes	Yes	Yes	**	**	**
BCT 510D PT	†	Yes	Yes	Yes	Yes	Yes	Yes	‡	‡	†
7303 and 05S	No	Yes	Yes	Yes	Yes	Yes	Yes	—	—	—
7505 *	No	No	No	No	Yes	Yes	Yes	**	**	**
7506 *	No	No	No	No	Yes	Yes	Yes	**	**	**
7507 *	No	No	No	No	Yes	Yes	Yes	**	**	**
Unified Messaging:										
LWC on Switch	Yes	Yes	Yes	Yes	Yes	—	—	—	—	—
LWC on AP	No	Yes	Yes	Yes	Yes	—	—	—	—	—
MCS and AUDIX	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
MCS, UNIX System, and AUDIX	No	Yes	Yes	Yes	Yes	Yes	Yes	No	Yes	Yes

* Available as a line appearance only, thus not recordable on CDR.

† Available but with limited or reduced functionality.

** MCS messages and mail can be received using EIA terminals attached to DCP Mode 2 interfaces that are available with these terminals.

‡Functions as an enhanced EIA terminal with screen labeled keys.

Notes:

Appendix J: Acronyms and Abbreviations

The following are the specialized acronyms and abbreviations and their associated terms. Explanations of these terms are provided within the text or in the glossary.

- ***AAR:*** Automatic Alternate Routing
- ***ACD:*** Automatic Call Distribution
- ***ACTGA:*** Attendant Control of Trunk Group Access
- ***ACU:*** Automatic Calling Unit
- ***ADFTC:*** Analog Digital Facilities Test Circuit
- ***ADM-T:*** Asynchronous Data Module - T Interface
- ***ADU:*** Asynchronous Data Unit, ***or*** Automatic Dialing Unit
- ***AIM:*** Asynchronous Interface Module
- ***AIOD:*** Automatic Identification of Outward Dialing
- ***AMA:*** Automatic Message Accounting
- ***ANI:*** Automatic Number Identification
- ***AP:*** Applications Processor
- ***ARS:*** Automatic Route Selection
- ***ASAI:*** Adjunct/Switch Application Interface
- ***ASCII:*** American Standard Code for Information Interchange
- ***ATMS:*** Automatic Transmission Measurement System
- ***AUDIX:*** Audio Information Exchange
- ***AUTOVON:*** Automatic Voice Network
- ***AVD:*** Alternate Voice Data
- ***AWG:*** American Wire Gauge
- ***BCC:*** Bearer Capability Class
- ***BCCOS:*** Bearer Capability Class of Service
- ***BCT:*** Business Communications Terminal
- ***Bisync:*** Binary Synchronous Communications
- ***BLF:*** Busy Lamp Field
- ***BOS:*** Bit-Oriented Signaling
- ***bps:*** Bits Per Second
- ***BRI:*** Basic Rate Interface

- ***BTC:*** Better Transmission Checking
- ***CAAVT:*** Call Answer From Any Voice Terminal
- ***CAS:*** Centralized Attendant Service
  or
  Call Accounting System
- ***CCIS:*** Common-Channel Interoffice Signaling
- ***CCITT:*** International Telephone and Telegraph Consultation Committee
- ***CCS:*** Hundred Call Seconds
- ***CCSA:*** Common Control Switching Arrangement
- ***CDM:*** Channel Division Multiplexer
- ***CDR:*** Call Detail Recording
- ***CDRP:*** Call Detail Record Poller
- ***CDRR:*** Call Detail Recording and Reporting
- ***CDRU:*** Call Detail Recording Utility
- ***CEM:*** Channel Expansion Multiplexer
- ***CIC:*** Carrier Identification Code
- ***CMDR:*** Centralized Message Detail Recording
- ***CM:*** Call Management
- ***CMS:*** Call Management System
- ***COS:*** Class of Service
- ***CPE:*** Customer Premises Equipment, ***or*** Customer Provided Equipment
- ***CRC:*** Cyclic Redundencey Check
- ***CRT:*** Cathode Ray Tube
- ***CSM:*** Centralized System Management
- ***CSMDR:*** Centralized Station Message Detail Recording
- ***CSU:*** Customer Service Unit
- ***CU:*** Control Unit
- ***DAC:*** Dial Access Code
- ***DACS:*** Digital Access Cross-Connect System
- ***dB:*** Decibels
- ***DCA:*** Data Communications Access
- ***DCE:*** Data Communications Equipment

- ***DCIU:*** Data Communications Interface Unit
- ***DCP:*** Digital Communications Protocol
- ***DCS:*** Distributed Communications Service
- ***DDC:*** Direct Department calling
- ***DDCMP:*** Digital Data Communications Message Protocol
- ***DDD:*** Direct Distance Dialing
- ***DID:*** Direct Inward Dialing
- ***DMA:*** Direct Memory Access
- ***DMI:*** Digital Multiplexed Interface
- ***DNHR:*** Dynamic Nonhierarchical Routing
- ***DNIS:*** Dialed Number Identification Service
- ***DOD:*** Direct Outward Dialing
- ***DSA:*** Dial Service Assistance
- ***DSC:*** Dedicated Switch Connections
- ***DSU:*** Data Service Unit
- ***DS1:*** Digital Service - 1
- ***DTDM:*** Digital Terminal Data Module
- ***DTGS:*** Direct Trunk Group Selection
- ***DTE:*** Data Terminal Equipment
- ***DTMF:*** Dual Tone Multi-Frequency
- ***DTS:*** Disk/Tape System
- ***DXS:*** Direct Extension Selection
- ***EAS:*** Expert Agent Selection
- ***EBCDIC:*** Extended Binary Coded Decimal Interchange Code
- ***EDC:*** Electronic Document Communications
- ***EFC:*** Electronic File Cabinet
- ***EIA:*** Electronics Industry Association
- ***ELL:*** Equipment Line Location
- ***EPSCS:*** Enhanced Private Switched Communications Service
- ***ERAS/AES:*** Enhanced Remote Access Security/Adjunct Enhanced Security
- ***ES:*** Enhanced Services
- ***ESS:*** Electronic Switching System

- ***ETA:*** Extended Trunk Access
- ***ETN:*** Electronic Tandem Network
- ***EUCD:*** Enhanced Uniform Call Distribution
- ***FADS:*** Force Administration Data System
- ***FAX:*** Facsimile
- ***FEAC:*** Forced Entry of Account Codes
- ***FM:*** Facilities Management
- ***FP:*** Feature Package
- ***FRL:*** Facilities Restriction Level
- ***FX:*** Foreign Exchange
- ***GPP:*** General Purpose Port
- ***GRS:*** Generalized Route Selection
- ***GT:*** General Trade ***GTA:*** General Terminal Administration
- ***HCA:*** Host Computer Access
- ***HCMR:*** High Capacity Minirecorder
- ***HNPA:*** Home Numbering Plan Area
- ***IACC:*** Integrated Adjunct Call Control
- ***IBM:*** International Business Machine Corporation
- ***ICI:*** Incoming Call Identification
- ***IDDD:*** International Direct Distance Dialing
- ***IDT:*** Integrated Display Terminal (7407D)
- ***IE:*** Information Element
- ***IMT:*** Intermachine Trunk
- ***IPA:*** Interpartition Access
- ***ISDN:*** Integrated Services Digital Network
- ***ISN:*** Information Systems Network
- ***ISO:*** International Standards Organization
- ***ITGI:*** *(Obsolete)* Integrated Telemarketing Gateway Interface. Former name for CallVisor™ ASAI Gateway Interface.
- ***IXC:*** Interexchange Carrier
- ***I/O:*** Input/Output
- ***Kbps:*** Kilobits per second

- ***KSR:*** Keyboard Send and Receive
- ***LADS:*** Local Area Data Set
- ***LAN:*** Local Area Network
- ***LATA:*** Local Access and Transport Area
- ***LDN:*** Listed Directory Number
- ***LND:*** Last Number Dialed
- ***LSU:*** Local Storage Unit
- ***LUDM:*** Linked Universal Data Module
- ***LWC:*** Leave Word Calling
- ***LXD:*** Last Extension Dialed
- ***MAAP:*** Maintenance and Administration Panel
- ***MADU:*** Multiple Asynchronous Data Unit
- ***Mbps:*** Megabits per second
- ***MCS:*** Message Center Service
- ***MCT:*** Malicious Call Trace
- ***MDM:*** Modular Data Module
- ***MET:*** Multibutton Electronic Telephone (DIMENSION System Voice Terminal)
- ***MFT:*** Multifunction Telephone (Multiappearance Voice Terminal)
- ***MFAT:*** Multifunction Analog Telephone (73-Series)
- ***MFDT:*** Multifunction Digital Telephone (74-Series, 510D, 515 BCT)
- ***MFET:*** Multifunction Electronic Telephone (72-Series)
- ***MHz:*** Megahertz (or cycles) per second
- ***MIA:*** Most Idle Agent
- ***MIM:*** Management Information Message
- ***MIS:*** Management Information System
- ***MLDN:*** Multiple Listed Directory Number
- ***MMWR:*** Manual Message Waiting Receive
- ***MMWS:*** Manual Message Waiting Send
- ***MOS:*** Message-Oriented Signaling
- ***MPDM:*** Modular Processor Data Module
- ***MTCP:*** Maintenance Test Circuit Pack
- ***MTDM:*** Modular Trunk Data Module

- ***Mux:*** Multiplex
- ***NCOSS:*** Network Control Operations Support System
- ***NCP:*** National Control Point
- ***NFAS:*** Non-facility Associated Signaling
- ***NPA:*** Numbering Plan Area
- ***NSF:*** Network-Specific Facilities
- ***NSO:*** Network Service Office
- ***ORPI:*** Optically Remoted Peripheral Interface
- ***OTL:*** Originating Test Line
- ***PC:*** Personal Computer
- ***PCM:*** Pulse Code Modulation
- ***PDM:*** Processor Data Module
- ***PRI:*** Primary Rate Interface
- ***PT:*** Personal Terminal
- ***PVC:*** Permanent Virtual Circuit
- ***QDN:*** Queue Directory Number
- ***RAM:*** Random Access Memory
- ***RF:*** Radio Frequency
- ***RJE:*** Remote Job Entry
- ***RLT:*** Release Link Trunk
- ***RMATS:*** Remote Maintenance, Administration, and Traffic System
- ***ROM:*** Read-Only-Memory
- ***RS:*** Recommended Standard
- ***SCI:*** Switch Communication Interface
- ***SCS:*** System Clock Synchronizer
- ***SCSI:*** Small Computer System Interface
- ***SDCPI:*** Switched Digital Communications Protocol Interface
- ***SDLI:*** Synchronous Data Link Interface
- ***SDN:*** Software Defined Network
- ***SID:*** Station Identification
- ***SLS:*** Straight Line Set
- ***SMDR:*** Station Message Detail Recording

- ***SMT:*** System Management Terminal
- ***SNA:*** Systems Network Architecture
- ***SPID:*** Service Profile Identifier
- ***SSI:*** Standard Serial Interface
- ***Service SPID:*** Service Service Profile Identifier
- ***TAA:*** Tandem Access Avoidance
- ***TEL:*** Terminal Equipment Location
- ***TCM:*** Terminal Change Management ***or*** Traveling Class Mark
- ***TDM:*** Trunk Data Module
- ***TEMPEST:*** Text Editing, Management, and Processing in an Enhanced Synchronized Terminal
- ***TMS:*** Time Multiplexed Switch
- ***TNS:*** Transit Network Selection
- ***TRACS:*** Translation Recovery, Additions, and Conversions System
- ***TRE:*** Tandem Routing Efficiency
- ***TRIB:*** Transfer Rate of Information Bits
- ***TRL:*** Trunk Reservation Limit
- ***TSI:*** Time Slot Interchanger
- ***TTL:*** Terminating Test Line
- ***TTTN:*** Tandem Tie Trunk Network
- ***TTY:*** Teletype
- ***TVA:*** Trunk Verification by Attendant
- ***TVVT:*** Trunk Verification by Voice Terminal
- ***UCD:*** Uniform Call Distribution
- ***UDM:*** Universal Data Module
- ***UDP:*** Uniform Dialing Plan
- ***UUCP:*** UNIX System-to-UNIX System Copy
- ***UUI:*** User-to-User Information
- ***VBR:*** Variable Bit-Robbed
- ***VDN:*** Vector Directory Number
- ***VDS:*** Voice Data Station
- ***VFCDR:*** Variable Format Call Detail Recording

- ***VMAAP:*** Visual Maintenance and Administration Panel
- ***VNI:*** Virtual Nodepoint Identifier
- ***VOM-T:*** Voice Only Module - T Interface
- ***VRU:*** Voice-Response Unit
- ***WATS:*** Wide Area Telecommunications Service
- ***WCR:*** World Class Routing
- ***WS:*** Work Station.

Glossary

10- to 7-Digit Conversion

A process used with the ARS (Automatic Route Selection) feature that allows the switch to convert the dialed number of a call to conform to the needs of a new (different) network. This process is used specifically, for public network numbers that can be routed over private network facilities (converts the 10-digit public network number to a 7-digit private network number). See also **Digit Modification**

AAR

See **Automatic Alternate Routing.**

Abandoned Call

A call during which the calling party "hangs up" ("abandons" the call) before the call can be connected to the called party. Often, calls are abandoned when the calling party is waiting in queue for an appropriate answering position to become available.

Access Code

A 1-, 2-, 3-, or 4-digit dial code used to activate or cancel a feature. The star (*) and pound (#) are sometimes used as the first digit of these codes.

Account Code

A dialed code (used with the CDR feature) that allows a call to be charged to a specific department's (or project's) account.

ACD Call

A call directed toward an ACD split using an associated extension number (or a published number). Since these calls enter the split's queue, ACD calls differ from calls to an agent's individual extension number which do not enter the queue.

ACD Console

The CALLMASTER voice terminal. This digital multiappearance voice terminal is primarily designed for ACD agents. The special attributes of this terminal for ACD agents include built-in alphanumeric display, two direct Starset-headset connectors, horizontal button layout, a fixed MUTE button, and a moderate price. (This voice terminal can also serve as a cost-reduced attendant console for the System 75.)

ACI (AT&T Communication Interface)

The name given to ISDN—PRI on the 4 ESS switching system. The 4 ESS is a Class 4 switch used for toll switching.

Action Control Point (ACP)

A switch in the AT&T Switched Network that recognizes an SDN (Software Defined Network) call and queries the NCP (Network Control Point) for routing instructions. The NCP returns specific routing instructions to the ACP.

Active on a Loop

An attendant is answering an incoming call or originating a call by pressing one of six appearance buttons.

Adjunct/Switch Application Interface (ASAI)

A protocol through which adjunct processors and switches coorperate to provide services that permit adjunct-based software applications to initiate, receive, and control calls or make use of switch features.

Administer

To add, remove, or change software programs or stored data that provide services or features.

Administrable Recall Button

A feature button that can be assigned as a RECALL button to those multiappearance voice terminals that do not have a fixed RECALL button. These RECALL buttons can be assigned to voice terminals on System 85s beginning with R2 V2. The multiappearance voice terminals without a fixed RECALL button include Models 7401D, 7404D, 7406D, 7407D, 7410D, 7505D, 7506D, 7507D, 510D, and CALLMASTER.

Advanced 800 Service

The second generation of the AT&T 800 Service. This collection of enhanced 800 Service offerings provides additional capabilities, flexibility, and control to 800 Service customers. Some of the Advanced 800 Service offerings include: Single Number Service, Area Code Routing, Time Manager, Day Manager, Call Allocator, Call Prompter, Command Routing and Courtesy Response.

Agent

A member of an ACD (EUCD in Release 2, Version 2) split.

Agent Hold

See **Multiple Call Handling.**

Agent Queue

A ordered sequence of agents who are available for and are waiting to receive an ACD call.

AIOD (Automatic Identification of Outward Dialing)

The AIOD feature identifies, translates, and transmits the calling party's extension number and trunk-group access code to the serving switch (i.e., CO, CCSA, or EPSCS switching office). In turn, the serving switch augments, compiles, and records this call-detail information and periodically provides the information to the customer.

Alerting

Audible (ringing) and/or visible signals indicating incoming calls.

American Standards Code for Information Interchange (ASCII)

An 8-bit binary code adopted by the American Standards Association to achieve compatibility between data services. Only seven of the eight bits are used for character identification. The eighth bit is left available for use as a parity bit.

Analog

Continuous in form. Analog signals contrast with digital signals which are discontinuous, or discrete, in form.

Analog Voice Terminals

An analog voice terminal (telephone) receives acoustic voice signals and sends analog electrical signals along the line. These voice terminals are served by a single wire pair (tip and ring). The analog voice terminals include Models 2500, 7101A, 7102A, and 7103A.

Answer-Back Channel

A group of dedicated circuits which a paged party can use to answer a page.

ANI (Automatic Number Identification)

See **AIOD.**

Answer Supervision

A signal sent by System 85 to the serving CO (Central Office) indicating that an incoming call has been answered. Upon receiving this signal, the originating CO (generally) begins tracking toll charges for the call (if charges apply).

Appearance

A point of access to an extension (also known as an occurrence). Using a multiappearance voice terminal, an extension can be manually accessed by pressing a button labeled with the extension number. Indicator (status) lamps next to the button light when a terminal user places a call, receives a call, or puts a call on hold.

AP (Applications Processor)

An adjunct minicomputer to System 85.

Area Code

See **Numbering Plan Area.**

ARS

See **Automatic Route Selection**

ASAI

See **Adjunct/Switch Application Interface.**

ASAI Gateway

A hardware and software package that provides a gateway between a DEFINITY Generic 2 and call-center software residing on a host computer. The ASAI Gateway enables the call-center software to monitor and control certain incoming, outgoing, and internal calls on the DEFINITY Generic 2.

ASCII

See **American Standards Code for Information Interchange.**

Associated Extension Number

An extension number that is assigned to and correlated with another extension number in Procedure 001.

Calls to an ACD (previously, EUCD, UCD, or DDC) split are routed to the split's queue with an "associated extension number." The system then distributes the calls to an available agent from the queue. Since these calls enter the split's queue, they are considered ACD (or EUCD / UCD / DDC) calls.

Asynchronous Data Transmission

Transmission in which time intervals between transmitted characters may be of unequal length. Transmission is controlled by start and stop elements at the beginning and end of each character. For comparison see ***Synchronous Data Transmission.***

Attendant

The operator of an attendant console.

Attendant Console

An electronic switchboard, with pushbutton control, used by the attendant to manage calls.

Attendant Diversion to Recorded Announcement

A function of the Intercept Treatment feature. Using this function, a System 85 attendant can divert ***all*** attendant-seeking calls to a recorded announcement by dialing an access code.

Attendant-Extended Call

A call (usually from the public network) that routes to an attendant, and is in turn relayed ("extended") by the attendant to an appropriate destination (e.g., a voice terminal or an ACD split). Attendant-extended calls differ from ***direct attendant calls*** that are ***initiated*** by an attendant.

Attendant Overflow

A function of the Tenant Services feature. Attendant overflow allows calls directed to an attendant partition to overflow and terminate to that partition's overflow partition. This function operates during periods of heavy calling to attendants, or when attendant partitions are lightly staffed.

Attendant Queue

The ordered sequence of calls waiting to be answered by an available attendant from within the group of attendants. (Using a System 85, there can be as many as 40 attendants in the group.)

ATMS

See **Automatic Transmission Measurement System.**

Audible Alerting

See **Ringback Tone.**

Audit Trail

A record of Automatic Circuit Assurance referrals. This system-generated record includes the date, time of day, trunk-group access code, and trunk number. The record indicates whether the referral was for a long call or a series of short calls. Also, the record indicates whether the trunk was tested after a referral or not. The audit trail record can be retrieved using the Facilities Management feature.

Attedant-Completing Trunk

A member of an attendant-completing trunk group.

Attendant-Copleting Trunk Group

Incoming (or 2-way) groups that terminate calls to preassigned answering positions within the switch. Since the answering positions are preassigned, digits are not passed to the local switch. These preassigned answering positions include: the attendant queue (by default), an ACD queue (Procedure 115), or a VDN (Procedure 031, Word 2). (The term "attendant-completing" usually refers to CO, 800 Service, or FX trunk groups.)

Authorization Code

A dialed code that can override the FRL of the facility being used to place an outgoing call. Authorization Codes can also be used (in preference to the Barrier Code) to protect against unauthorized entry to System 85 on Remote Access trunks.

Automatic Alternate Routing (AAR)

A feature on System 85 and DEFINITY Generic 2.1 switches, that provides least-cost routing for private network calls by selecting, in descending order of desirability, the best route available.

Automatic Calling Units (ACU)

An automatic dialing device that permits a machine to place calls over the communications system.

Automatic Dialing

A method of using Abbreviated Dialing to place a call. With Automatic Dialing, the calling party accesses a stored number with a single button press.

Automatic Incoming Trunk

A member of an automatic incoming trunk group.

Automatic Incoming (Automatic-in) Trunk Groups

Incoming (or 2-way) trunk groups that terminate calls to preassigned answering positions within the switch. Since the answering positions are preassigned, digits are not passed to the local switch. These preassigned answering positions include the attendant queue (by default), an ACD queue (Procedure 115), or a VDN (Procedure 031, Word 2). (The term "automatic-in" usually refers to the various kinds of tie trunks.)

Automatic Route Selection (ARS)

A feature on System 85 and DEFINITY Generic 2.1 swiches, that provides least-cost routing for public network. calls by selecting, in descending order of desirability, the best route available.

Automatic Transmission Measurement System (ATMS)

A System Management feature that automatically measures the transmission characteristics of private and public network trunk facilities.

Backup Answering Position

The second or third termination point in a coverage path.

Backup Terminal

A voice terminal used with Centralized Attendant Service (CAS) to answer calls at a branch location when the attendant at the main location is not available.

Balance Field

Field 2 of Procedure 101, Word 1. When this field is assigned to "1" for a trunk group, the switch inserts an extra 2 dB of loss to each trunk-to-trunk connection involving an analog CO (Central Office), FX (Foreign Exchange), WATS (Wide Area Telecommunications Service), or DID (Direct Inward Dialing) trunk This field is set to reduce reflections (including echoes) on analog trunks where the serving CO is not providing enough impedance compensation (i.e., balance) for the trunk group. (A trunk is considered balanced with an Echo Return Loss of at least 16 dB.)

Band Width

The range of frequencies assigned to a channel or system; the difference expressed in hertz between the highest and lowest frequencies of a band.

Barrier Code

The security code that allows a remote user to access the services of the System 85 and prevents unauthorized access to the system.

Basic Rate Interface (BRI)

One of two standard ISDN frame formats. The BRI is the station direct interface to an ISDN. It consists of a 192 Kbps carrier interface organized into two 64 Kbps B (bearer) channels and one 16 Kbps D (data) channel.

Bay

Another term for "split" that is used by some other ACD vendors. A split (or bay) is a group of agents (group members) organized to receive calls in an efficient and cost-effective manner. Throughout this manual, the term "split" is used in preference to "bay."

BCT

See **Business Communications Terminals.**

Bearer Capability

A term used with ISDNs (Integrated Services Digital Networks) that identifies the types of traffic that can be supported by a given facility.

Bearer Capability Class (BCC)

A term used with ISDN that identifies the type of carrier service required for a caller. Part of the class of service for the calling party and identifies the bearer capability required.

Bearer Capability Class of Service (BCCOS)

An expansion of the ISDN BCC concept used with System 85, R2 V4, and DEFINITY Generic 2 switches. The BCCOS provides additional granularity to the BCC concept and applies to both ISDN and non-ISDN calls and facilities.

Binary Synchronous Communications (BSC or Bisync)

A data communications protocol, developed by IBM, that interfaces the AP to an IBM host computer. This protocol is used with the Terminal Emulation feature.

Bisync

See **Binary Synchronous Communications.**

Bit (Binary Digit)

Smallest unit of information in binary notation (one of two possible states or values).

Bits Per Second (bps)

The number of units of information in binary notation that are transmitted or received per second.

Bit-Oriented Signaling (BOS)

A form of call control signaling that transmits information in the form of single digital bits, most commonly A & B bit signaling.

Blocking

The inability to connect the calling party with the called party. Blocking occurs when either (1) all suitable trunk paths are busy or (2) a path between a given inlet and a suitable outlet of the switching system is unavailable.

Branch Locations

Telecommunication systems served by attendants at a centralized location.

BRI Voice Terminal

A voice terminal designed to use the "2 B + D" interface specified in the ISDN—BRI standard. The AT&T BRI voice terminals available with DEFINITY Generic 2, include the 7500D series models 7505D, 7506D, and 7507D. Because ISDN—BRI is an open standard, other manufacturer's voice terminals may also be used once certified.

Bridged Appearance

An appearance on a voice terminal matching an appearance on another voice terminal. For every appearance on a "home" terminal, there can be a bridged appearance on as many as 15 other voice terminals. (The concept of is similar to the concept of "images.")

Bridging

Connecting one circuit in parallel with another without interrupting the continuity of the first.

BSC

See **Binary Synchronous Communications.**

Business Communications Terminals (BCT)

A family of terminals that provide the user with the ability to enter data and message information via a keyboard. The terminals are 500 BCT, 513 BCT, and 515 BCT.

Bypass Access

A method of connecting to public telecommunications facilities whereby a customer (terminal or PBX) is provided with direct physical connections to a long distance carrier service that physically ***circumvent*** the LEC (local exchange carrier).

Bypass Access Tie Trunks

One-way tie trunks from an ETN (Electronic Tandem Network) tandem switch to an ETN main location that is homed on a ***different*** tandem within the ETN configuration.

Call Appearance

A point of access to an extension (also known as an occurrence). Using a multiappearance voice terminal, an extension can be manually accessed by pressing a button on the voice terminal labeled with an extension number. Indicator (status) lamps next to the button light when a terminal user places calls, receives calls, or puts a call on hold.

Call-by-Call Service Selection

An attribute of the ISDN—PRI feature. Call-by-Call Service Selection allows ISDN trunks to be more efficient than traditional trunk facilities. By specifying a calling service the ISDN—PRI call SETUP message, Call-by-Call Service Selection allows different network services to be applied to the same trunks on a per-call basis. Therefore, ISDN—PRI trunk groups reduce the need to provide separate trunk groups for services such as WATS (Wide Area Telecommunications Service), DID (Direct Inward Dialing), and DOD (Direct Outward Dialing). All of these services can be provided over the same set of trunks.

The ISDN—PRI feature also allows the same digital trunk facilities to handle various types of calls. The SETUP message contains information to specify the type of call including voice calls and various types of data calls. This ISDN—PRI capability, provided on the Dynzmic Trunk Type (Trunk Type 120), is provided by System 85 and DEFINITY Generic 2 switches using the Generalized Routing function of the AAR, ARS, and WCR features. Also known as Network Specific Facilities (NSF) selection.

Call Category

The classification of a call based on selected factors that can influence the selection of a network routing pattern. The call category is used by the generalized route selection function of the AAR, ARS, and WCR features. The factors used in determining call category vary depending on the type and vintage of switch. On System 85 and DEFINITY Generic 2.1 switches, call category is determined by the conditional routing or satellite hop control count. On Generic 2.2 switches, call category definition expands to include not only the conditional routing count, but also the time-of-day plan, and the partition of origin on switches with Tenant Services active.

Call Detail Recording and Reporting (CDRR)

The CDRR feature collects, formats, and reports switch generated call detail information using a local Applications Processor. The CDRR feature surpasses SMDR as a comprehensive and customized way of reporting call details.

Call Detail Recording Utility (CDRU)

A series of software programs that run on the 3B2 LSU to process the CDR records generated by the switch.

Call Forwarding—Off Net

A function of the Call Forwarding—Follow Me feature. Using this function, a user can forward all calls to a telephone in the nontoll public network.

Call Forwarding Override

A function of the Call Forwarding—Follow Me feature. With Call Forwarding—Follow Me active, most calls directed to the forwarding terminal forward to a specified destination terminal. Call Forwarding Override allows the user of the forwarded-to terminal to complete calls to the forwarding terminal.

Call Hold

The name of the access code that allows a single-appearance voice terminal user to access the hard hold function of the Hold feature. See **Hold.**

Call Management System (CMS)

Software routines on an adjunct processor that assist in the management of ACD activity. The CMS system generates reports on ACD activity, provides real-time displays of current agent activity, and provides the customer with the ability to partially administer the ACD feature for the switch.

Call-Progress Tone

One of a set of tones (e.g., ringback tone, busy tone, or reorder tone) that a calling party can receive from a switch. Call-progress tones indicate the specific status of a call or the facilities used by the call.

Call Reference Value

A number between 0 and 32767, used to identify a specific ISDN call in progress. The call reference value is used to associate ISDN D-channel messages with the related ISDN call.

Called Party

The person who receives a call.

Caller Response Interval

The administrable period of time for internal callers to respond to a call that has been redirected to coverage. During this system-wide interval (from 0 to 10 seconds) that begins with coverage tone, an internal caller could hang up, activate Leave Word Calling, or wait for the covering user to answer.

Calling Party

The person who places a call.

Carrier Ready Tone

See **Ready Tone.**

CCITT (Committee Consultative for Inernational Telegraph and Telephone)

An agency of the United Nations (under the auspices of the International Telecommunication Union). The CCITT is also known in english as the International Telegraph and Telephone Consultive Committee. It is a multinational organization that sets international telecommunications standards.

CCS (Hundred Call Seconds)

The unit with which telephone call traffic is measured. The equivalent of one call lasting for 100 seconds.

CCSA

See **Common Control Switching Arrangement.**

CDRR

See **Call Detail Recording and Reporting.**

CDRU

See **Call Detail Recording Utility.**

Cell

A 4-bit (1/2 byte) portion of a CDR record. Each CDR record contains four adjacent cells.

Central Office

A place where public telephone switching equipment is housed.

Central Office Trunk

A telecommunications channel on the public network between the central office and the system.

Central Processor

The part of a computer system that contains the main storage, arithmetic logic unit, and special registers.

Channel

A communications path over which voice or data is transmitted.

Channel Negotiation

A capability of ISDN connections that allows the called (terminating) switch to request, on a call-by-call basis, a different channel than the channel selected by the calling switch. For an ISDN—BRI connection, channel negotiation is between a terminal device and a switch, rather than between switches. The benefit of channel negotiation is that the only other alternative, if the originally selected channel is not acceptable for some reason, is to deny the call.

Chime Sidetone

See **Chimeback Tones.**

Chime Signals

Tone bursts sent over the loudspeaker system to page persons with the Code Calling Access feature. These 892-hertz tone bursts are combined in a coded manner to signal a specific person out of a larger group of persons.

Chimeback Tones

Tones (resembling chime signals) sent to the paging party's voice terminal by the System 85 switch. These tones are used to confirm that chime signals are being sent over the loudspeaker system.

Class of Service (COS)

A numeric code that specifies a group of features and calling privileges that together determine the calling privileges of a group of extension numbers.

Cluster

A group (2 or more) of individual nodes connected by DCIU links and tie trunks into a DCS network.

CMS

See **Call Management System.**

Codepoint

A term used in ISDN terminology to identify a point within a "codeset" that defines an IE (Information Element). See also **Codeset** and **Information Element.**

Codeset

One of eight possible standard groupings, consisting of 133 ISDN IEs (Information Elements), used in the ISDN messaging structure. Codeset 0 contains the IEs defined by the CCITT. Codesets 1 through 5 are reserved for future standards expansion. Codeset 6 is for IEs specific to the local serving network (switch), and codeset 7 is for user-specific (user-to-user) IEs. See also **Information Element.**

Codeset Conversion

An administrative process, used with ISDN messaging that allows an IE (Information Element) from one codeset to be translated to an IE in a different codeset. This allows ISDN messaging to be used between two separate interfaces (switches) that use different codeset structures.

Codeset Mapping

Another name for Codeset Conversion.

Collision

See **Glare.**

Command Path

See **Path Name.**

Common Channel Signaling

A method of providing call control and signaling for multiple links (trunks) on a common or shared channel. This is the signaling form required by DS1 clear channel or AVD (Alternate Voice Data) arrangements and by ISDN (Integrated Services Digital Network) arrangements. Also known as 24th channel signaling.

Common-Channel Interoffice Signaling (CCIS)

A system of supervising the routing, setting up, and disconnecting of voice and data calls within a telecommunications network. CCIS is the primary method of signaling used within the AT&T Communications Network. AT&T CCIS consists of a network of processors and high-speed data links residing in parallel to the actual switching and transmission facilities. Some of the benefits of CCIS signaling include faster call-setups, smarter "look-ahead" routing capabilities, fewer blocked calls, lower-cost routing.

Common-Control Switching Arrangement

A private telecommunication network using dedicated trunks and a shared switching center for interconnecting company locations.

Conditional Routing

A method of controling call route selection in the System 85 and DEFINITY Generic 2 routing features (AAR, ARS, and WCR). Conditional routing selects possible routing patterns based on the number of satellite links used by a call and the number of satellite links that are included in a potential routing patterns. See also, Satellite Hop Control.

Confirmation Tone

Three short bursts of tone provided by the System 85 switch to confirm that a feature has been accepted or canceled.

Control Vectors

Another term for "Call Vectoring" that is used by some other ACD vendors. Call Vectoring (or the similar concept, Control Vectors) provides an enhanced method of defining calling-party interfaces for incoming calls (usually incoming ACD calls). Throughout this manual, the term "Call Vectoring" is used rather than "Control Vectors."

Controlling Terminal

For Abbreviated Dialing the voice terminal within a group that is used to program and change group-list items for the group.

For ACD, the terminal within an assigned ACD (previously, EUCD / UCD / DDC) group that is used to activate or deactivate functions for the group.

Conversion (Code)

See **Code Conversion.**

Conversion (Digit)

See **10- to 7-Digit Conversion** and **Digit Modification.**

Conversion Resource

The combination of a TDM (Trunk Data Module) and a modem that converts DCP (Digital Communications Protocol) data signals from digital format to analog format suitable for transmission over analog trunk facilities. The reverse conversion (analog to digital) is also provided. This term is used primarily with the Modem Pooling feature.

Coverage Call

A call that is redirected from the called extension (the principal) to another extension.

Coverage Group

The combination of a principal, the principal's coverage path(s), and the associated criteria. A dual-path coverage group contains two paths and criteria.

Coverage Module

An optional attachment to the 7205H or the 7405D voice terminal. Coverage modules provide 20 additional appearance buttons for these voice terminals, and are usually used by covering users. The C201A coverage module attaches to the 7205H, and the C401A or C401B module attaches to the 7405D or the 7434D.

Coverage Path

An ordered sequence of up to three answering positions (points) to which coverage calls are restricted.

Coverage Point

An extension designated as an alternate answering position in a coverage path.

Coverage Tone

A call-progress tone, also known as Coverage Redirect Feedback, that indicates to a calling party that the coverage process is being started for the call.

Covering User

A person authorized to answer redirected calls at a coverage point.

Criteria

The conditions under which a call to a principal is redirected to coverage. This term is used with the Call Coverage feature.

Crossover (Network)

See **Network Crossover.**

Customer-Provided Equipment (CPE)

Customer-owned equipment that is not provided as part of the system but is to be connected to it.

Cut-Through Connection

An attribute of outgoing calls that are not automatically routed (such as AAR or ARS calls). Cut-through connections are established between local and distant switches whereby the local switch sets up a talking connection to the distant switch and then relinquishes control of the call to the distant switch. At this time, the distant switch may return dial tone for subsequent dialing and call routing.

D (Data) Channel

The designation for either of two types of communications link:

DCP usage

A communications link (as opposed to signaling link) in the DCP setup.

ISDN Usage

The signaling link (as opposed to a communications link) for an ISDN span, either PRI or BRI.

DC (Digit Collect) Signal Ignore

An attribute of the AAR and ARS Subnetwork Trunking functions. The DC Signal Ignore fields are Field 11 of Procedure 309, Word 1 and Field 8 of Procedure 321, Word 1. When the DC Signal Ignore field is assigned to an AAR/ARS preference, the System 85 only sends digits to the serving switch for outgoing (or tandem) calls after the assigned pause times out. The System 85 does not send digits in response to a Wink-Start, Delay-Dial, or precise dial tone signal.

This field might be assigned to an AAR/ARS preference to compensate for excess noise on the trunk group. As an example, the trunk type of many AAR preferences allows for "universal" outgoing signaling. These trunk groups can respond to either Wink signals, Delay-Dial signals, or precise dial tone. (In this way, the System 85 switch administrator need not be overly concerned about the precise trunk type at the distant end of the preference.) However, excess noise over one of these trunks can be misinterpreted as digit-collect signal which would result in the System 85 sending digits before the receiving switch is ready to receive them.

D-Channel Backup

A technique used with ISDN NFAS (Non-Facility Associated Signaling) on PRI (Primary Rate Interface) spans to provide a second (redundant) D-channel for a D-channel group.

With D-channel backup, two signaling channels (D-1 and D-2) are assigned to a D-channel group. One D-channel is active (usually D-1) and the other is in a standby mode (active at level 2 only [of the ISO Model]). When the active D=channel is taken out of service for any reason, the standby channel automatically takes over signaling for the assigned B-channels.

D-Channel Group

A term used with ISDN NFAS (Non-Facility Associated Signaling), to identify the B-channels and D-channels that are associated in a common signaling arrangement.

Data Channel

The means of transmission and the intervening equipment involved in the transfer of information in a given direction.

Data Communications Access

The feature that provides access to on-premises host computers for data end points with an analog interface (modem). These end points include analog voice terminals and attendant consoles.

Data Communications Equipment (DCE)

The equipment that provides the functions required to establish, maintain, and terminate a connection, the signal conversion, and coding required for communication between DTE (Data Terminal Equipment) and data circuit. Any equipment that connects to a data terminal device using an EIA RS-232C interface.

Data Communications Interface Unit (DCIU)

A circuitry configuration that provides interprocessor communication between switch processors or between a switch processor and another processor.

The DCIU is used between switches in a DCS cluster, between a switch and an Applications Processor, or between a switch and an AUDIX system.

Data Link

The configuration of physical facilities enabling end terminals to communicate directly with each other.

Data Path

The complete connection (end-to-end) used for a data communications link (see also Virtual Circuit). This term is used for the combination of all elements used in an interprocessor communication in the DCS feature.

Data Module

A device that interfaces customer-provided data equipment to the System 85 equipment.

Data Port

A point of access to a computer that uses trunks or lines for sending or receiving data.

Data Protection

The feature that prevents intrusions by bridge-on features (e.g., Call Waiting, Override, and Busy Verification of Lines) into data transmissions. The bridge-on feature, if allowed, would disrupt a data transmission by inserting a warning tone into the connection. Two forms of Data Protection are available: temporary and permanent. Temporary data protection is activated with an access code and permanent data protection is assigned to a line class of service.

Date Rate

The transmission of data measured in bits per second.

Data Service Unit (DSU)

A device designed specifically to transmit digital data on transmission facilities.

Date Set

A term sometimes used for ***modem.*** A device that converts data communications between a digital (i.e., RS232C) format and an analog format. These terms (data set and modem) are used interchangeably in this manual.

Data Supportable Tie Trunk (DSTT)

A tie trunk that can support data rates greater than 300 bps.

Data Terminal

An input/output device [with a keyboard and a CRT (cathode ray tube)] that has either switched or direct access to a host computer (or to an AP). The AT&T data terminals include: The AT&T Personal Terminal 510D, and Models 500, 513, and 515 BCT (Business Communications Terminal).

Data Terminal Equipment (DTE)

1) The equipment comprising the data source, data destination, or both. A data endpoint such as a data terminal or line printer.
2) The configuration of leads and control devices (and logic) specified by a protocol that complements the ***Data Communications Equipment (DCE) for the same protocol.***

Data Tone

See **Ready Tone.**

Datagram

A data communications technique that does not involve 2-way or even direct connections between the two endpoints. It is analogous to a letter being delivered by a postal service. The data is submitted to the system and later delivered to its destination. There is no direct connection between the sender and destination.

DCIU

See **Data Communications Interface Unit.**

DCIU Link

A hardware communications link (data link) that connects two DCIUs.

DCIU Network Channel

The association between two link/logical channel pairs such that a DCIU message received on one link/logical channel pair is transmitted on the other pair. There are two types of network channels, fixed or PVC (Permanent Virtual Circuit) and alternate routing.

DCIU Port

A gateway to or from an application (DCS is an example of an application). Ports are the endpoints of a virtual circuit and look like an input/output device to the application.

DCS

See **Distributed Communications System.**

DCS Cluster

Two or more switches interconnected by DCIUs.

DCS Node

A switch within a DCS cluster.

Default FRL

A value (between 0 and 7) assigned to a specific extension class of service or trunk group. The default FRL of the originating facility is used as the initial FRL for a call for route selection purposes. *See also* Facility Restriction Level.

Default Voice Terminal

A preassigned voice terminal to which calls can be routed when the attendant console is unattended.

Delay Dial Signaling

A trunk signaling protocol that provides a waiting period for the far end of a trunk connection to respond before the near end outpulses digits. See Appendix F, Enhanced Trunking, for more information on trunk signaling protocols.

De-multiplexer

A device or circuit used to separate two or more signals that were previously combined on a single carrier medium. The de-multiplexor must be associated with a compatible multiplexer.

Designated Voice Terminal

The forwarded-to voice terminal (that is, the specific voice terminal to which calls, for a certain extension, are forwarded) in a Call Forwarding arrangement.

Dial Access Restriction

A method of preventing voice terminal users, data terminal users, and attendants from directly accessing a trunk group by dialing the trunk-group access code. Dial access restriction also prevents an attendant from accessing a restricted trunk group with a DTGS

(Direct Trunk Group Selection) button. This restriction (assigned to a trunk group in Procedure 100, Word 1), does not prevent verification of trunks in the trunk group by an attendant or the designated voice terminal user.

Dial Pulse Addressing

A means of signaling consisting of regular momentary interruptions of a current path at the sending end. The number of these interruptions corresponds to the value of a digit or character. This method of address signaling is usually associated with rotary dial terminals. However, this method of signaling can also be used between switches.

Dial Repeating Tie Trunk

A telecommunications channel between two systems. The number dialed is "repeated" (or dialed-in) at the distant end.

Digit Modification

A function of the WCR (World Class Routing) feature that allows the switch software to change the digits dialed to meet routing requirements. Also known as M to N conversion.

Digit Strings

A term used in the WCR (World Class Routing) feature to refer to the elements of a dialed number. A digit string is any grouping of digits (one or more) that constitutes a descrete element of a dialed number string. Digit strings are used within the digit analysis module of the WCR software to make call processing decisions about the call in progress. Digit strings include such elements as: account codes, interexchange carrier access codes, international dialing prefixes, toll prefixes, and address strings.

Digit-Oriented Routing

A method of routing calls into the System 85 switch. Using digit-oriented routing, dialed digits are passed through the serving switch (usually, the serving CO) and to the local System 85 in a similar manner to DID calls.

Digital Access Cross-Connect System (DACS)

A large electronic "cross-connect field" where telecommunications channels are permanently connected using software translations. AT&T Communications maintains a DACS network in parallel to the AT&T Switched Network that primarily contains 4 ESS switches. To limit the trunk load on each 4 ESS, private lines through the AT&T Communications network are usually routed through a set of DACS offices to reach their destinations.

Specific trunks in a DACS office can also interface the AT&T Switched Network using translated connections to the adjacent 4 ESS. As an economical alternative when this interface is used, the DACs can also concentrate the incoming direct-access trunks to consume fewer switched trunk facilities at the 4 ESS.

Digital

Discrete, or discontinuous, in form. Digital signals contrast with analog signals which are continuous in form. Digital signals are usually binary.

Digital Communications Protocol (DCP)

A protocol used to transmit both digitized voice and data over the same communications link. [A System 85 data link made up of two information channels and one signaling channel.]

Digital Data

Data represented in discrete, discontinuous form, usually binary. This is in contrast to continuous analog data usually sine wave form.

Digital Service-1 (DS1)

A high-speed, high-volume digital trunking facility.

Digital Terminal Data Module (DTDM)

A data module designed as a plug in module for a digital voice terminal.

Digital Trunk

A circuit in a telecommunications channel designed to handle digital data.

Digital Voice Terminal

A voice terminal (telephone) that converts acoustic voice signals (analog signals) into digital electrical signals to be sent along the line. These voice terminals are served by two pairs of wire. The ditial voice terminals include: Models 7401D, 7403D, 7404D, 7405D, 7406D, 7407D, 7410D, 7434D, 7505D, 7506D, 7507D, 510D, 515 BCT, and CALLMASTER.

Direct Access

A method of connecting to public telecommunications facilities whereby a customer (terminal or PBX) is provided with direct physical connections to a long-distance carrier that employ fixed ***translated*** connections through the LEC (local exchange carrier).

Direct Attendant Call

A call that is ***initiated*** by an attendant. Direct attendant calls differ from ***attendant-extended calls*** that are initiated elsewhere (usually from the public network), and extended by an attendant to an appropriate destination.

Direct Distance Dialing

Long distance calls completed without operator assistance.

Direct Extension Selection

An option on an attendant console which allows an attendant direct access to an idle voice terminal (inside the system) by pressing a hundreds button and a tens and units button.

Disconnect Supervision

A signal sent by the serving switch to the local System 85 indicating that an "answered call" was hung up at the distant end. Depending on the type of serving switch, a delay of from 2 to 25 seconds can precede this signal.

Display Voice Terminal

A voice terminal with display capabilities. The display voice terminals include: 7404D (with Z300B messaging cartridge and EIA terminal), 7405D (with D401A display module), 7406D With Display, 7407D, 7506, 7507, 510D, 515 BCT, and CALLMASTER.

Distant

Pertaining to, or within, the physical limits of another network node [from the perpective of the local (or serving) switch].

Distributed Communication Service (DCS)

A network of switches appearing to the user as a single switch.

Don't Answer Interval

The specified number of times that unanswered voice terminals can ring (number of alerting cycles) before a call is redirected to coverage or forwarded.

Drop Button

A fixed feature button on the AT&T multiappearance voice terminals. When a multiappearance voice terminal user is the controller of a 3-Party Conference, the user can press the DROP button to disconnect the third party in the conference. Also, when a multiappearance voice terminal user is active on a 2-party call, the user can press the DROP button to disconnect the other party and to receive new dial tone on the same appearance.

Using the DROP button to disconnect a 2-party call and receive new dial tone is faster than using the DISCONNECT button. Using the DISCONNECT button returns dial tone on the appearance specified by the Multiappearance Preselection and Preference feature.

Dual Tone Multifrequency (DTMF)

A signaling technique where specific tone pairs are used to represent the dialing characters (0 through 9, *, and #). Also known as touch-tone.

Duplicated Switch

A switching system that contains backup processing equipment. A "high-reliability" switch contains backup common control processors. A "critical-reliability" switch contains backup common control processors, module processors, and TMSs (Time Multiplexed Switches).

DXS Button

One of a set of 100 buttons, arranged as 2 groups of 50 buttons, engraved with a number (from 00 to 99) on the type 34 console. When DXS is assigned, an attendant can select an extension (such as, 1356) by pressing the appropriate hundreds group selection button (for this example, the one assigned as "13") and then pressing the appropriate DXS button (for this example, "56"). When Extended DXS is assigned, however, the attendant can select an extension by pressing two DXS buttons in sequence (for this example, "13," followed by "56"). When Extended DXS is assigned, the hundreds group selection buttons do not function.

Dynamic Nonhierarchial Routing (DNHR)

The primary algorithm used to route calls through the AT&T Switched Network. Each 4 ESS in the network has a direct route to most of the other 4 ESS switches in the network. Meanwhile, each 4 ESS can also have up to 13 indirect routes with just one intervening node. Under this algorithm, the direct route is usually the first preference, and the indirect routes are arranged in order of desirability. At the time that a preference is actually selected for a call, each preference is queried on a "look-ahead" basis by the CCIS network. After the CCIS network reserves an available path from the originating 4 ESS, through the intervening 4 ESS (if needed), to the destination 4 ESS, the CCIS network instructs the originating 4 ESS to route the call over this path.

E&M (Ear and Mouth) Supervision

A symmetric signaling scheme using dc voltage levels over "E" and "M" interface leads. The E and M leads are separate from the transmission path. The signals are used to indicate on-hook and off-hook states at each end of the connection path. See **Appendix F, Enhanced Trunking,** for more information on trunk signaling protocols.

Egress

The opposite of access; the act of going out or emerging. For example, 800 Service and/or MEGACOM WATS 800 Service can be used to egress the public network for access to a System 85. Also, the SDN (Software Defined Network) Access feature allows a single call to use special access from a System 85 or DEFINITY Generic 2 to the public network and then use special egress from the public network to another System 85 or Generic 2.

Electronic Document Communications (EDC)

A group of services provided by the AP 16 that allows users to prepare, manage, and exchange information. One of these services is a form of electronic mail.

Electronic Tandem Network (ETN)

A private telecommunications network configured with electronic tandem switches. It interconnects customer locations via dedicated intertandem tie trunks, access trunks, and bypass trunks.

Emulation

A technique using software programming that allows one computer or digital device to behave like a different device.

Enhanced Private Switched Communications Service (EPSCS)

A private network service that provides advanced voice and data communications services for companies with widespread operations.

Equipment Location

The location, or the corresponding numerical representation of the location, of a circuit within the switch. Equipment locations are represented by 7-digit numbers (in the form, 01 2 3 12 0). The first and second digits identify a module. The third digit identifies a cabinet within the module. The fourth digit identifies a carrier within the cabinet. The fifth and sixth digits identify a slot within the carrier. The seventh digit identifies a circuit of the slot's circuit pack.

Extended Digital Subscriber Line (EDSL)

The name used for the ISDN—PRI feature (network interface) on the 5ESS switching system. The 5ESS is used for COs (Central Offices) and large PBX (Private Branch Exchange) applications.

Extension Number Steering

An attribute of the Main / Satellite feature. The Main / Satellite switches use the initial digit(s) of a dialed extension number to steer the call to the appropriate switch. The call is then routed to its final destination.

Facilities Restriction Level (FRL)

An number (between 0 and 7) assigned to calls and calling facilities, used during route selection, to determine accessability of call routing facilities for specific calls. A call can access a calling facility (trunk group or routing preference) if the FRL of the call is equal to or greater than the FRL of the facility being accessed. *See* also Default FRL.

Facsimile

A process, or the result of a process, where fixed graphic material is scanned and the information converted into electrical signal waves to produce a likeness or copy.

Feature

A specifically defined function or service provided by System 85.

Feature Key Module

See **Function Key Module.**

Feature Button

A labeled button designating a specific feature.

Fiber Optics

A technology using light-guide materials and ultra-wide band electromagnetic wave forms for high-capacity carrier systems.

Final Effective Step

A "final effective (vector) step" is either the last vector step or a vector step that is followed by a "stop" step.

Flash

A momentary press of (noun) [or to momentarily press (verb)] the switchhook on an analog voice terminal. Switchhook flashes can have a duration of from 0.2 seconds to 1.2 seconds. A shorter press would not be detected, while a longer press would cause the voice terminal to disconnect.

Flash Button

Another term for "recall button" that is used by some other switch vendors. After a recall button press (or flash button press) the switch returns recall dial tone to the voice terminal user. Throughout this manual, the term "recall button" is used in preference to "flash button."

Flexible Routing Selection (FRS)

Another term for "ARS" (Automatic Route Selection) that is used by some other switch vendors. ARS (or the similar concept, FRS) provides least-cost routing of public network calls. Throughout this manual, the term "ARS" is used in preference to "FRS." On Generic 2.2 switches the ARS feature is replaced by the WCR (World Class Routing) feature.

Flow Control

See **ISDN Flow Control.**

Foreign Exchange (FX)

A central office other than the one located in the calling customer area.

Foreign Exchange Trunk

A telecommunications channel that connects a private telephone system to a central office other than its own central office.

Forwarded-To Voice Terminal

The designated voice terminal (i.e., the specific voice terminal to which calls, for a certain extension, are forwaded).

Frame

One of several segments of an analog or digital signal that has a repetitive characteristic. Corresponding elements of successive frames represent the same thing. The specific nature of a frame varies depending on the content in which the term is used.

- In a time-division system, a frame is a sequence of time slots, each containing a sample from one of the channels served by the system.
- In DS1 Signaling, a frame is the digital representation of the 24 logical channels represented by 192 bits of data. An additional bit is included for framing making a frame 193 bits long.
- In D4 signaling, the framing format includes 12 frames.
- In extended super framing (Fe), the framing format includes 24 frames.
- In packet switched data communications, a frame incloses a "packet" by adding a header and a trailer element for additional control purposes (See the "Packet" and "Packet Switching" definitions for further details).

Full Duplex

A transmission system capable of carrying signals in both directions simultaneously.

Function Key Module

An optional attachment for the 7205H or the 7405D voice terminal. Function key modules provide 24 additional feature buttons for these voice terminals.

Gate

Another term for "split" that is used by some other ACD vendors. A split (or gate) is a group of agents (group members) organized to receive calls in an efficient and cost-effective manner. Throughout this manual, the term "split" is used in preference to "gate."

General Purpose Port (GPP)

An SN270 (or TN754 for universal modules) port used for either a digital voice terminal or a data module. The GPP uses the digital communications protocol and can be used as either a line or trunk appearance.

General Trade (GT)

Commercially available materials and equipment that are not restricted to a single manufacturer or to manufacture under license.

Generalized Route Selection (GRS)

The process of selecting a routing pattern and preference for a call (in a networking invironment) that uses a series or set of administrable criteria to make the routing selection. GRS is used by the AAR, ARS, and WCR features.

Ghost Call

A queued incoming call that is directed to an idle answering position even though the calling party has already abandoned the call. Abandon call search can be used with the ACD or EUCD features to minimize the occurence of ghost calls.

Glare

A fault condition in trunk seizure operations where both ends of a trunk attempt to seize the same connection at the same time. See **Appendix F,** Enhanced Trunking, for more information on trunk signaling protocols.

Ground Start Supervision

A supervisory signal given at certain terminals and switches by connecting one side of the line or trunk to ground. Ground start signaling was introduced in trunk signaling to minimize glare. See Appendix F, Enhanced Trunking, for more information on trunk signaling protocols.

Group Number

Software packages before R2 V2 allow for as many as 28 UCD/DDC groups. In Procedure 025, Word 1, the "group number" (a number between 1 and 28) is used to uniquely identify the individual groups. These same group numbers are used in Procedure 011, Word 1 to specify a UCD/DDC group which is used as the final point in a coverage path.

GTA (General Terminal Administration)

A capability introduced with DEFINITY Generic 2 that enables a system administrator to define the characteristics of new multiappearance terminals and data modules. That is, terminal types that are not defined in the switch administration software (Procedure 051, Word 1).

Half Duplex

Transmission of signals in either direction but not both directions simultaneously.

Handshaking Logic

Logic circuits used to establish a data connection between two devices.

Hard Disk

A rigid magnetic platter used to store data.

Hard Hold

One of the two forms of hold provided by the Hold feature. Hard hold is more durable than soft hold, providing greater assurance of being able to return to the held call. Hard hold is available for multiappearance voice terminals (using the HOLD button) and for single-appearance voice terminals (using the Hold dial access code).

Hard Processor Swap

An attribute of a duplicated System 85. During a "hard swap," the switch abruptly shifts control from one processor to the other "healthier" processor. Since the current call-status information is not transferred to the new processor, hard processor swaps are usually noticed by users who are active on a call. (See also **Soft Processor Swap.**)

Hard-Wired

Permanently connected [as opposed to a temporary (switched) connection].

Head-End Hop Off

A call-routing attribute of ETN main and tandem switches. Using head-end hop off, a main or a tandem quickly routes a public-network call over a public-network trunk facility instead of routing the call part of the way over private-network trunk facilities.

Head of Queue

The first item (e.g., a call or an agent) in a queue. The item at the head of queue is usually the next item to be processed.

hertz

A unit of frequency equal to one cycle per second.

Hold

A feature provided for System 85 voice terminal uses. The Hold feature allows the user to separate from an active call, while retaining the ability to return to the call. There are two forms of hold: hard hold and soft hold. Hard hold is more ***durable*** than soft hold, providing greater assurance of being able to return to the held call. Hard hold is available for multiappearance voice terminals (using the HOLD button) and for single-appearance voice teminals (using the Hold dial access code). Soft hold is only available for single-appearance voice terminals (using the switchhook or the RECALL button).

Home Numbering Plan Area (HNPA)

The dialing plan area aarea code in the public network) within which a private network switch is located.

Homing

The connection of a subtending switch to a tandem switch based on factors such as traffic loads, geographic proximity, and community of interest.

Hookswitch

See **Switchhook** or **Recall Button.**

Hop

Nondirect communication between two DCIUs, whereby the DCIU message passes through one or more intermediate DCIUs.

Host Computer

A computer connected to the network that processes data from various data-entry devices.

Hub

The central or controlling switch in a "STAR" networking configuration.

Hundred Call-Seconds (CCS)

The unit with which telephone call traffic is measured. One call that lasts for 100 seconds constitutes 1 CCS.

Hundreds Group Selection Button

One of 18 assignable buttons (arranged horizontally as three nows of six buttons) on the type 34 console. When DXS is assigned, an attendant can select an extension by pressing the appropriate hundreds group selection button and then pressing the appropriate DXS button. When Extended DXS is assigned, the hundreds group selection buttons don't function.

Hybrid Module

See **Universal Module.**

Hybrid Voice Terminal

Hybrid voice terminals (telephones) possess characteristics of both analog and digital voice terminals. Like analog voice terminals, hybrid voice terminals send analog electrical voice signals. However, like digital voice terminals, hybrid voice terminals are controlled by digital signals. These voice terminals are served by three pairs of wire. The hybrid voice terminals include Models 7203H, 7205H, 7303S and 7305S.

I-Field

A channel used for communicating digital information for voice and data.

I-Use Lamp

An indicator lamp on a multiappearance voice terminal that indicates whether or not a particular appearance is in use.

ID/Password

A security measure that limits access to those with the correct ID and password.

Idle Loop

An idle appearance on the attendant console.

IE

See **Information Element.**

Initializing Terminal

A term used with specific ISDN—BRI terminals to indicate that the terminal interface exchanges information with the switch (similar to handshaking between data modules), when first places into service or when returned to service after an outage.

Image

A point of access to an appearance. There can be as many as 16 images of an appearance. An unshared appearance has only one image of the appearance. (The concept of an "image" is similar to the concept of a "bridged appearance.")

Immediate Start Signaling

A trunk signaling protocol in which the receiving switch (incoming end of a trunk seizure) has 0.07 seconds to respond to the incoming seizure. See Appendix F, Enhanced Trunking, for more information on trunk signaling protocols.

Individual Extension Number

Agents in an ACD (or EUCD, UCD, DDC) split have "individual extension numbers" and are allowed to receive calls on these numbers. Calls to an individual extension number do not enter the split's queue and are not considered ACD calls.

IMT (Intermachine Trunk)

See **Intertandem Tie Trunk.**

Information Element (IE)

A logical block of data in an ISDN message (control and signaling). IEs provide specific information related to terminals, lamps, ringing, data data rates, and other call related data. Information elements are specified as "codepoints" within the series of ISDN "codesets." See also **Codepoint** and **Codeset.**

INOS

See **Interactive Network Optimization System.**

Integrated Services Digital Network (ISDN)

According to the CCITT, a network "that provides end-to-end digital connectivity to support a wide range of services, including voice and nonvoice service to which users have access by a limited set of standard multipurpose user-network interfaces."

Intelligent Main

Usually, a main location in a Main / Satellite / Tributary configuration that accesses an ETN (Electronic Tandem Network) tandem switch and that also has the AAR (Automatic Alternate Routing) feature assigned. AAR is assigned to the intelligent main to give this switch local control over some of the AAR private-network routing decisions.

Intelligent Data Terminal

A data terminal containing a microprocessor to reduce the data transmitted and to expand the data received.

Interactive Network Optimization System (INOS)

A heuristic computer program that is maintained and operated by AT&T to help design cost-effective private / virtual-private networks.

Intercept Tone

An alternating high and low tone; indicates a dialing error or denial of the service requested.

Interface

A common boundary between two systems or pieces of equipment.

Interflow

The diversion of calls from an ACD (previously, EUCD) split to another node of the network (also known as Overload Balancing).

International Standard Organization (ISO)

An agency of the United Nations, given the responsibility for establishing and coordinating common standards between member nations.

Intertandem Tie Trunk

A private telecommunications channel (of Trunk Type 41 to 46) between two tandem switches in an ETN network. An intertandem tie trunk can be classified as either "high-usage," or "final."

High-usage trunk groups efficiently carry a large volume of call traffic. When an entire high-usage trunk group is busy, this trunk group can overflow to a final trunk group that carries a mixture of "first-route" and overflow traffic. Normally, final trunk groups do not overflow to alternate routes.

Intervening Switch

For Look-Ahead Interflow, a private- or public-network switch between the sending (or tandeming) and receiving switches of a Look-Ahead Interflow call. The intervening switch uses its usual routing software to continue the routing of these calls on an "ISDN Preferred" basis. However, when no ISDN facilities are available or permitted, the intervening switch continues to route the interflow call on a ***non***-Look-Ahead basis.

Interworking

The term used with ISDN to refer to the processes by which ISDN and non-ISDN services interconnect and work together.

Intraflow

The diversion of calls from an ACD (previously, EUCD) split to a destination within the local switch.

Intrusion Tone

A 440-hertz tone applied by System 85 during a call. Intrusion tones are used to alert the talking parties to the presence of an unexpected party in the connection. The features that are able to provide an intrusion tone include ACD (Automatic Call Distribution), Busy Verification of Lines, EUCD (Enhanced Uniform Call Distribution), Override, Trunk Verification—Attendant, and Trunk Verification—Voice Terminal.

ISDN Flow Control

A defensive mechanism designed to protect switch processors from overload due to an excessively hight level of ISDN D-channel messaging activity. In the DEFINITY Generic 2 time frame, Flow Control consists of monitoring and controlling the level of messaging activity at the interface (BRI or PRI) level.

ISDN Gateway

A type of ACD split administered in Procedure 026, Word 1. ISDN Gateway software runs on an adjunct processor. The ISDN Gateway software receives call-related information from the switch, converts the information into a form that a customer's application software can use, and then sends the converted information to the application software.

Least Cost Routing (LCR)

Another term for "ARS" (Automatic Route Selection) that is used by some other switch vendors. ARS (or the similar concept, LCR) provides least-cost routing of public network calls. Throughout this manual, the term "ARS" is used in preference to "LCR."

Leave Word Calling Administrator

A person authorized to access messages for anyone on the system.

Leave Word Calling Without an AP

Leave Word Calling messages can be stored on the switch, on an AUDIX system, or on an AP (Applications Processor). Leave Word Calling without an AP refers to message storage on the switch or on an AUDIX system. That is, the messages are not stored on an AP.

LEC

See **Local Exchange Carrier.**

Line Port

The actual hardware providing the access point to the system switching network for each circuit associated with an extension.

Link

The physical connection between two devices. Specifically, with the CDR feature, the CDR (or SMDR) link is the physical connection between the output port on the switch and the CDR recording adjunct. With the DCS feature link usually refers to the physical connection between the DCIU and the local switch (switch port) or between ports on separate DCIUs (located on different switches).

Link Access Procedure (LAP)

A group of standards (LAP, LAPB, LAPD, etc.) adopted by the CCITT for the link level protocols used with CCITT standards such as X.25 and ISDN. The CCITT LAP standards are generally modeled after the various ISO recommendations for HDLC (High-level Data Link control).

Listed Directory Number (LDN)

The listed number in a public directory (phone book) for a private switching system that is usually answered by an attendant.

Local

Pertaining to, or within, the physical limits of the serving switch.

Local Area Data Set (LADS)

A modem, specifically the 48230 and 48234, manufactured by Codex Corporation, These data sets support synchronous data transmission at selectable rates of 2.4, 4.8, and 9.6 Kbps. They offer good noise immunity.

Local Area Network (LAN)

A networking arrangement specifically designed to support a limited geographical area. Generally, they are limited in range to a maximum of 6.2 miles and provide high-speed carrier service with low data error rates. Common confirmation include: star (included circuit switched), ring, and bus.

Local Distribution Service Unit (LDSU)

A modem that supports synchronous data transmission at selectable rates of 2.4, 4.8, and 9.6 Kbps.

Local Exchange Carrier (LEC)

The telephone company that provides switching and transmission service for local calls and long-distance calls within the LATA (Local Access and Transport Area).

Local Storage Unit (LSU)

A device that collects and stores CDR records. Usually, an LSU temporarily stores records that are periodically requested by a central polling device.

Logical Channel (or Circuit)

A message slot on a digital communications link. One of the logically independent elements of a data stream that are multiplexed onto a single communications carrier. This term is used in connection with "packet switching" and to refer to a designated subelement of a DCIU link. In the DCIU application, a logical circuit on the "switch link" is called a "port."

Long-Holding Time

A measurement of call length on selected trunks; occurs automatically when the Automatic Circuit Assurance (ACA) feature is activated.

Look-Ahead Interflow

A System 85 and DEFINITY Generic 2 feature that is usually an ACD application of the Call Vectoring and the ISDN—PRI features. (Call Vectoring provides the decision-making capabilities, and ISDN—PRI provides the messaging capabilities between switches.) A sending switch usually uses Look-Ahead Interflow to divert ACD calls to a different switch in the ETN network after asking the receiving switch whether that switch can adequately handle each call.

If the receiving switch can handle the interflow call, the sending switch will send the call. If not, the sending switch either queries another switch or performs a predefined alternate action.

Loop

A voice circuit associated with an appearance button on the console, used by the attendant to processor originate calls.

Loop Start Supervision

An older, and in some ways antiquated, signaling protocol that is used by some older COs in rural areas. The newer ground start signaling protocol offers improved signaling capabilities (including disconnect supervision and glare resolution) over the loop start method. Loop start signaling relies on a supervisory signal initiated by a terminal or switch in response to completing the loop current path. See Appendix F, Enhanced Trunking for more information on trunk-signaling protocols.

LSU

See **Local storage Unit**

M to N Conversion

A process used by the World Class Routing feature to change the form and content of a digit string (dialed number). The M to N conversion process allow the number being processed to be changed to conform to the needs of call routing over a network other than that originally dialed. This process is used to support the network crossover function of the WCR feature. M to N conversion is similar to the 10- to 7-digit conversion process used by the earlier ARS (Automatic Route Selection) feature; however, M to N conversion is more flexable and more powerful the 10- to 7- digit conversion.

MAAP

Maintenance and Administration Panel. A device used by the vendor's service technicians to completely administer the System 85.

Main Location

A centralized area where attendants answer calls routed from branch locations.

Main Split

The split to which a call controlled by vector processing is initially directed. Vector processing always queues calls to this split unless the split is not staffed.

Main/Satellite

A Main / Satellite complex is a private network configuration. This configuration can either stand alone or access an ETN (Electronic Tandem Network). The trunk facilities that access the public network and attendant positions are usually concentrated at the main.

Main/Tandem Field

Field 4 of Procedure 103. Assigning a "1" in this field gives certain capabilities to a trunk group. For example, the Main / Tandem field must be assigned to send an FRL TCM over an outgoing trunk group. The Main / Tandem field cannot be assigned to apply AAR subnetwork trunking to an outgoing trunk group. Also, the Network Trunk field must be assigned to each DCS trunk group where AAR routes the calls. Otherwise, transparency is lost for DCS calls.

Management Information Message (MIM)

One of a set of special ISDN messages used to communicate maintenance and management information between the switch processor and an ISDN—BRI terminal or end point.

Manual Dialing

Using the touch-tone dialing pad (or rotary dial) to place a call. The calling party dials each digit of the called number individually. Manual dialing contrasts with Automatic Dialing where a single button press outpulses a set of stored digits.

Measeured Agent

A member of an ACD split whose activity is measured by CMS (Call Management system).

Measured Split

A group of ACD agents whose activity is measured by CMS (Call Management System).

Mega-

A prefix used with units to indicate a factor of 1,000,000. Typically *N* Megahertz is *N* X 1,000,000 hertz (or cycles) per second and *N* Megabits is *N* X 1,000,000 bits per second.

Message Center

A service that provides for answering of calls that would otherwise go unanswered, accepts and stores messages for later retrieval.

Message Center Agent

A person with membership in a Message Center split. Message Center agents take and retrieve messages for voice terminal users.

Message-Oriented Signaling (MOS)

Call related signaling over a D (Data) channel that takes the form of a set of standard messages rather than bit signals. Used with ISDN.

MIA Distribution

See **Most Idle Agent Distribution.**

Modem Pooling

A feature that provides shared "conversation resources" (modems plus data modules) for cost-effective access to analog facilities by data terminals. When needed, a conversion resouce is inserted into the path of a data call. This feature serves both outgoing and incoming calls.

Modular Trunk Data Module (MTDM)

A digital interface device that provides the interface for DCP (Digital Communications Protocol) connections to trunk facilities. It is used between off-premises private line trunk facilities and the System 85 switch, and for connections to DDD modems as the DCP member of a Modem Pooling conversion resource. The MTDM provides conversion between the RS-232 protocol and the DCP protocol (see also Trunk Data Module).

Most Idle Agent (MIA) Distribution

A call-distribution routine provided by the ACD (Automatic Call Distribution) feature that distributes an ACD call to the "most idle" agent. An available agent who has not been active on an ACD call for the longest time is considered the most idle agent.

Multiappearance Voice Terminal

A terminal equipped with several appearance buttons for the same extension number. This allows the user to handle more than one call, on the same extension number, at the same time. The multiappearance voice terminals include: Models 7203H, 7205H, 7303S, 7305S, 7401D, 7403D, 7404D, 7405D, 7406D, 7407D, 7410D, 7505D, 7506D, 7507D, 510D, 515 BCT, and CALLMASTER.

Multibutton Voice Terminal

A terminal equipped with multiple buttons to access and cancel features. These buttons can be either feature buttons or Abbreviated Dialing buttons (using access codes as the stored numbers). The multibutton voice terminals include Models 7103A Fixed Feature, 7103A Programmable, 7203H, 7205H, 7303S, 7305S, 7401D, 7403D, 7404D, 7405D, 7406D, 7407D, 7410D, 7505D, 7506D, 7507D, 510D, 515 BCT, and CALLMASTER.

Multidigit Steering

A form of Extension Number Steering. This form of Main / Satellite steering uses two or more digits to route a call to the appropriate switch. The call is then routed to its final destination.

Multiple Call Handling

A function of the R2 V4 ACD feature that is primarily intended for Message Center agents. Multiple call handling, when assigned to a split, allows an ACD agent to place a call on hold (using the HOLD button) and remain available to receive ACD calls.

Multiplexer

A device for simultaneous transmission of two or more signals over a common transmission medium.

Network

Two or more interconnected switching systems or telecommunications end points. A physical network. *See also* Local Area Network (LAN).

For the WCR (World Class Routing) feature; a system of digit strings and digit string attributes that provides a software representation of a "physical network."

Network Channel

See **DCIU Network Channel.**

Network Control Point (NCF)

The remote computer (usually a 3B20) that, in response to a routing query from an ACP (Action Control Point), returns routing instructions to the ACP for SDN calls. The NCP contains the customized database for an SDN customer that specifies routing options and the correct point of egress from the AT&T Switched Network.

Network Crossover

A function of the World Class Routing feature that allows a call to "crossover" from one (software) network to another during call processing. See also **M to N Conversion.**

Network Specific Facilities (NSF) IE

An ISDN information element used to request features or service or to obtain information from the transit network The NSF IE is part of the ISDN call setup message. *See also,* **Call-by-Call Service Selection.**

Network Trunk Field

Field 3 of Procedure 103. Assigning a "1" in this field gives certain capabilities to a trunk group. For example, the Network Trunk field must be assigned to apply ARS subnetwork trunking to an outgoing trunk group. The Network Trunk field must be assigned to each outgoing DCS trunk group. Otherwise, DCS calls will fail. Also, the Network Trunk field must be assigned to an incoming ETN trunk group so that, when necessary, the AAR feature can tandem calls.

Node

A switching or control point for a network. Nodes are either tandem (receive signals and pass them on) or terminal (originate or terminate a transmission path).

Nodal Main

A private switching system with direct or bypass access to one or more of the AT&T Communications services through a service node in the AT&T Switched Network [for example, ACCUNET Service, MEGACOM WATS Service, MEGACOM WATS 800 Service, and SDN (Software Defined Network)]. Usually, in this manual, nodal mains are given more specific names according to the specific service accessed (for example, ACCUNET main or SDN main).

Nonanalog Voice Terminals

Hybrid and digital voice terminals.

Noncircuit Call

Refers to ISDN D-channel message traffic that is not call related. These messages carry maintenance and management type information between switches.

Nonfacility Associated Signaling (NFAS)

The ISDN—PRI signaling mode where a D-channel can be assigned to provide call control and signaling support B-channels that are physically located on a different ISDN circuit board.

Numbering Plan Area (NPA)

A geographic division within which telephone numbers are assigned to a common public network are code. Another name for NPA is area code.

Observer

A person who is allowed to monitor the call-handling activity of ACD (or EUCD) agents. The service observing and/or agent override functions provide this capability. Often, split supervisors observe an agent's performance for training purposes. During training, inexperienced agents can also benefit by observing experienced agents handle difficult calls. To maintain courteous and efficient handling of calls, observers (outside the split) can also be appointed to evaluate the ongoing performance of agents.

Occurrence

A point of access to an extension (also known as an appearance). Using a multiappearance voice terminal, an extension can be manually accessed by pressing a button on the voice terminal labeled with an extension number. Indicator (status) lamps next to the button light when a terminal user places calls, receives calls, or places a call on hold.

Octet

The international (CCITT) term for an 8-bit data element. Equivalent to the term "byte" more commonly used in the United States.

Off-Hook

A term signifying that the voice terminal handset has been lifted.

On-Hook

A term signifying that the voice terminal handset has been replaced on the switchhook (hung up).

Opcode (Operation Code)

In some CDR record formats, a 4-bit (1-cell) code placed in the left-most cell position of each word. Certain CDR peripherals use the opcode as an end-of-record identifier.

Originating (Only) Appearance

One of the two kinds of appearances assigned in Procedure 052, Word 1: terminating/originating and originating (only). An originating (only) appearance of (or, point of access to) an extension is reserved for placing calls.

Normally, calls placed to an extension are not allowed to terminate to an originating (only) appearance. For most of these calls (without some type of redirection active), the switch returns busy tone to the calling party when every terminating/originating appearance is busy. However, priority calls and override calls ***can terminate*** to originating (only) appearances. With Priority Calling, this operation is considered preferable to returning busy tone for an important call. With Override, this operation is considered preferable to an unnecessary entry into an active appearance.

Originating Register

A temporary software record that contains necessary information for System 85 call processing.

Overflow Partition

An attendant partition that is assigned in Procedure 270, Word 2 to receive excess incoming calls from another attendant partition. Any attendant partition (from 0 to 40) can be assigned as an overflow partition in a partitioned System 85.

Overload Balancing

The diversion of calls from an ACD (previously, EUCD) split to another node of the network (also known as Interflow).

Packet

As used in data communications, a packet is a group of bits including a message element (data) and control information element (header) that is used in packet switching and transmitted as a discrete unit. Within each packet, the message elements and control information elements are arranged in a specified format. In many systems, the packet is further encapsulated with additional header and trailer elements to form what is called a "frame." (See the following "Packet Switching" definition and the BX.25 Protocol discussion in the DCS feature for further details).

Packet Switching

A digital transmission technique that sends addressed packets over logical circuits or channels multiplexed onto a common carrier medium. Packet switching is designed to make more efficient use of high speed carriers by multiplexing or sharing these facilities between multiple users and applications. When properly implemented, packets or frames travel on the shared carrier, with each packet being separate and independent from the packet that precedes it or follows it.

A protocol is used that ensures proper sequencing of delivered packets at a common destination, and addressing is used in a header element to distinguish between multiple destinations on the common carrier. The international standard for packet switching is the CCITT X.25 protocol. The BX.25 protocol is the AT&T implementation of X.25. AT&T commonly uses the BX.25 protocol for internal communications between digital control elements of AT&T systems.

Paging Trunk

A telecommunications channel used for accessing an amplifier (loudspeaker paging).

Parallel Data

Data bits that are processed or transmitted simultaneously.

Parallel Trunk Groups

Two or more trunk groups (each with a different trunk type) that interconnect the same two switches. In a parallel trunk-group arrangement, each trunk group is used for a different feature application or for a different call-routing strategy. A common example of parallel trunk groups might be DDD, WATS, and Personal CO Line trunk groups between a System 85 and the same CO. A less common example might be an intertandem tie-trunk group and a Main/Satellite trunk group connecting the same two System 85s in a private network.

Partition

To separate or divide a switch into discrete functional parts (verb). One of the discrete functional parts of a "partitioned switch" (noun).

Party Test

An electrical test most often used by some older COs (Central Offices) to determine which party on a party line is placing a toll call. Since the CO's party test would look like "disconnect supervision" to the System 85, trunk groups residing in System 85 traditional modules can be assigned to ignore the party test. This is done by assigning a "party test" trunk type to a trunk group that accesses the LEC (Local Exchange Carrier). These trunk types include Trunk Types 18, 20, 23, 25, and 28.

Since the party-test signaling process consumes call-processing time on an outgoing call, these trunk types should only be used when the serving CO ***requires*** that they be used.

Trunk groups in System 85 ***universal*** modules cannot be assigned to ignore the party test. If the serving CO requires that these signaling types be used, these CO trunk groups ***must reside*** in traditional modules.

Path Name

A "command path," formed as a specific series of entries, which is used to access the desired Manager IV, FM, or TCM screen (form). Throughout this manual, path names are listed in their complete, unabbreviated forms. However, for easy access to the screens, unique abbreviated versions of the path names may be entered.

Permanent Seizure

In trunk signaling, a trunk state in which the distant switch seizes a trunk but does not respond to normal signaling (disconnect supervision) or is maintenance busied out.

Permanent Virtual Circuit (PVC)

A fixed network channel assigned for communications between two DCIUs. A permanently dedicated (administered) circuit communicating between two DCIU ports.

Pickup Group

A group of individuals authorized to answer any call directed to a voice terminal extension number within the group.

Ping Ring

See **Ring Ping.**

Port

A point of access to the system or to a computer that uses trunks or lines for transmitting or receiving voice or data.

Precedence Ringback Tone

An attribute of the Precedence Calling feature. This ringback tone has the same component frequencies as standard ringback tone (440 + 480-hertz). However, the cycle of this ringback tone (approximately 1.65 seconds on and 0.35 seconds off) is considerably faster than the standard-ringback cycle.

Precedence Warning Tone

An attribute of the Precedence Calling feature. The System 85 switch applies this 440 + 620-hertz tone to a connection that is about to be preempted.

Precursor Tone

A special function tone used with the Remote Access feature on selected long distance trunks to perform echo suppressor control. The precursosr tone is a high pitched (2220-hertz) tone that lasts for 0.4 seconds and precedes standard dial tone. It is used during the call-setup dialing sequence on certain Remote Access trunk connections where transmission distances are likely to exceed 1200 miles.

Preference

Within the context of network routing features, a specific trunk group within a destination routing pattern. Preferences are arranged in order and with the earlier routing features (AAR and ARS) this order is assumed to be preferential (that is, the preferred trunk group first with subsequent trunk groups in descending order of preference) hence the name preference.

Prefix

A digit (other than a dial access code) dialed (or inserted) before the destination address of a call. Dialed prefixes are used to place a destination address in proper context, to indicate service options, or both (for example, a toll prefix). Also, a prefix can be assigned to an incoming trunk group. In this context, the prefix is inserted infront of the incoming digits and becomes part of the digit stream. This type of prefixing is used to provide additional call processing instructions (such as a dial access code) for the incoming call.

Primary Covering User

The first answering position in a coverage path when a call is redirected to coverage.

Primary Rate Interface (PRI)

The switch interface to an ISDN, as specified by the CCITT. A 1.544 Mbps interface organized into 24 time slots of 64 Kbps each (23 bearer channels and 1 data channel). A specific adaptation of the DS1 signaling format.

Principal (User)

A person whose calls are redirected to a covering user.

Private Network

A network accessible only by the owner for private communications needs.

Processor Data Module (PDM)

A device providing a DCE interface for connecting to data terminals, Application Processors, and host computers. It also provides a DCP interface for connection to the switch. It supports the downloading function of Terminal Emulation between the AP and the 515 BCT. It provides protocol conversion between the RS-232 protocol and DCP.

Protocol

A set of conventions or rules mutually understood and accepted by the parties involved. Protocols are the rules governing format and timing of message exchanges to control data transfer and correction of errors.

Public Network

A network which is commonly accessible for local or long distance calling.

Q.921

See **Recommendation Q.921.**

Q.931

See **Recommendation Q.931.**

Queue

An ordered sequence of items (e.g., outgoing trunk calls, incoming ACD calls, or ACD agent positions) waiting to be processed.

Queue Directory Number (QDN)

The first associated extension number of an ACD (previously, EUCD) split. A split's QDN is assigned in Procedure 026, Word 2. The QDN is also associated with the split supervisor's individual extension number in Procedure 001. For systems that have the Call Vectoring feature, QDNs are replaced by VDNs (Vector Directory Numbers).

Queue-Status Display

A function of the ACD and Call Vectoring features. When an ACD agent has a display terminal and the class-of-service assignment, information regarding the split's queue is provided to the agent on an automatic or on-demand basis. This information includes the number of calls currently in the split's queue (ACD and Call Vectoring) and the amount of time the oldest queued call has waited (call vectoring).

Queuing

The process of placing calls in an ordered sequence. The Queuing feature orders outgoing calls that are waiting for an idle trunk. Whereas, the ACD feature places incoming calls in queue to wait for an idle agent.

Radio Paging Trunk

A telecommunications channel used to access paging transmitter equipment.

Random Access Memory (RAM)

A storage arrangement that permits information elements to be addressed directly, independent of the location of the information in storage. Also, commonly used as a misnomer for Read Write Memory that offers the same form of direct addressing.

Read Only Memory (ROM)

An electronic memory device that is permanently recorded during the manufacturing process. It provides protected memory storage with no provisions for erasing or writing over its contents.

Read Operation

The process of retrieving information from memory.

Ready Tone

A steady tone received when a data connection is ready for completion. Ready tone is also known as "carrier ready tone" or "data tone."

Recall Button

A standard button provided with many System 85 voice terminals. The functionality of the Recall button differs between single-appearance and multiappearance voice terminals. When the Recall button is provided on singleappearance terminals (such as, 2500 and 7103A), the switch provides Recall Signaling (returns recall dial tone) when the Recall button is pressed during an active call. The Recall button on single-appearance terminals is also used with the Attendant Recall, Call Park, Conference—Attendant Six Party, Loudspeaker Paging Access, and Serial Calls features.

The Recall button has limited functionality on multiappearance terminals. This button is used with the Attendant Recall, Call Park Conference—Attendant Six Party, Loudspeaker Paging Access, and Serial Calls features. The Recall Signaling operation is not provided for multiappearance terminals. Instead, the Hold, Conference, and Transfer operations are performed using fixed feature buttons provided with the voice terminal.

Recall Dial Tone

Confirmation tone followed by dial tone. The confirmation tone confirms acceptance of a requested feature and the dial tone allows subsequent dialing to proceed.

Recommendation Q.921

The CCITT recommended standard of the ISDN Levels 1 and 2 (Physical and Link Level) protocol.

Recommendation Q.931

The CCITT recommended standard of the ISDN Level 3 (Network Level) protocol.

Recursive

In software programming, a function is "recursive" if the function contains a call on itself. Intraflow—All is a recursive function of the ACD and the EUCD features.

Redundant Switch

See **Duplicated Switch**

Register

A short-term storage circuit having a capacity (usually) of one computer word.

Release Link Trunk (RLT)

A telecommunications channel used with Centralized Attendant Service to connect attendant-seeking calls from a branch location to the main location.

Remote Access Trunk

A telecommunications channel used by an authorized user to gain access to the system.

Remote Maintenance, Administration, and Traffic System (RMATS)

A centrally located system that can remotely provide maintenance, administration, and traffic (calls) measurement for the System 85 or other similar equipment

Reorder Tone

Fast busy signal; indicates that the necessary facilities are busy. This tone also serves as the default form of Intercept Treatment for incoming calls from the public network.

Reversed Battery Supervision

A supervisory signal used by 1-way trunks that uses open and close signals from the originating end and reversals of battery and ground from the terminating end. See Appendix F, Enhanced Trunking, for more information on trunk-signaling protocols.

Ring Ping

As an attribute of the Call Forwarding—Follow Me feature. Just before forwarding a call, System 85 applies ring ping, a quick burst of ringing (0.1 seconds), to remind the forwarding party that unconditional forwarding is in effect.

As an attribute of the Send All Calls function of the Call Coverage feature. Just before unconditionally redirecting a call, System 85 can apply ring ping, a quick burst of ringing (0.1 seconds), to remind the principal that Send All Calls is in effect.

Ringback Queuing

The process by which a caller is placed in queue, hangs up, and is called back when an outgoing trunk becomes available.

Ringback Tone

Tone provided by the System 85 switch to confirm that the terminal on the other end of the line is ringing. The component frequencies of this tone are 440 and 480-hertz.

Ringing Cycle

Voice terminal ringing followed by the silence before the next ring. The duration of a ringing cycle is approximately 5.2 seconds (1.2 seconds of ringing and 4.0 seconds of silence). Before redirecting calls that are placed to an unanswered voice terminal, System 85 counts these ringing cycles as a way of timing the don't answer condition.

RNX

An expression that represents the location or office code for a private network switch (in a dialing string), where R equals any digit 2 through 9 except the CDR account-code prefix, N equals any digit 2 through 9, and X equals any digit 0 through 9.

Route Advance

A switch software routine (feature) that routes outgoing calls over alternate trunk groups when the dialed trunk group is busy.

Routing Tables

Another term for "Call Vectoring" that is used by some other ACD vendors. Call Vectoring (or the similar concept, Routing Tables) provides an enhanced method of defining calling-party interfaces for incoming calls (usually incoming ACD calls). Throughout this manual, the term "Call Vectoring" is used in preference to "Routing Tables."

S-Channel Data Transmission

A channel used for communicating control signals between the System 85 and a data module.

Satellite

An orbiting telecommunications transceiver that provides long-distance transmission (usually using the microwave transmission medium).

Also, in a Main / Satellite networking arrangement, a subtending switch that operates through the main switch and appears (from an external perspective) to be part of the main. In this arrangement, the main provides attendant services for the satellite, including a common LDN (Listed Directory Number) system.

Satellite Hop Count

The number of editing transceivers that have already been used in connecting a call to its destination. This information is sent with the call as a TCM (Traveling Class Mark).

Satellite Location

An unattended location in a Main / Satellite / Tributary complex that has the same listed directory number as the main switch and accesses the public and private network through the main.

Satellite Partition

An extension partition in a partitioned switch that also serves as a satellite location in a Main / Satellite configuration. This capability allows a partitioned System 85 to act as an endpoint in a tenant's private network.

SDN

See **Software Defined Network.**

Serial Data

Data bits that are transmitted or processed sequentially, one bit after the other.

Service Node

A gateway to the services and capabilities of the AT&T Telecommunications Network. A service node usually consists of: a 4 ESS switch and a cross-connect device [such as a DACS (Digital Access Cross-Connect System) or DSX (Digital System Cross-Connect)].

Shared Extension

An extension with more than one image (bridged appearance) of an appearance. System 85 recognizes two functional types of shared extensions shared extensions with ***only one*** appearance and shared extensions with ***more than one*** appearance. Straight line sets can participate in both types of shared extensions.

Short-Holding Time

A time limit set for internal calls when the Automatic Circuit Assurance feature is activated so that faulty lines can be detected.

Sidetone

The portion of a voice signal that is extracted from a telephone user's voice transmission and is returned to the receiver of the same telephone. Sidetone assures the telephone user that the telephone is working properly and helps the user adjust the volume of his/her speech.

SID (Station Identification)/ANI (Automatic Number Identification)

An attribute of the ISDN feature that delivers a calling party's number to the called party for an ISDN call. A SID is the calling party's number provided by the calling party and contained in the Calling-Party Number IE of a SETUP message (ISDN). Whereas, an ANI is the calling party's billing number either stored at the calling party's serving switch or stored within an Interexchange Carrier's network.

At service provisioning, an ISDN customer can choose whether to subscribe to SID/ANI. For those who do subscribe, there are two service alternatives.

1. SID/ANI provided to called party whenever available
2. SID/ANI provided to called party upon request.

Within each of the previous two categories, an ISDN customer also specifies which number (either SID or ANI) is preferred. This decision can be made in one of four ways.

1. First SID, then ANI (if SID not available)
2. First ANI, then SID (if ANI not available)

3. SID only
4. ANI only.

For the initial implementation of ISDN—PRI, the network does not specify which number the called party is receiving. This can produce ambiguity when the calling party has specified that ***either*** the SID or ANI number can be transmitted.

Simplex

A data transmission system that allows data to travel in only one direction at a time.

Simulated Bridged Appearance

See **Temporary Bridged Appearance.**

Single-Appearance Voice Terminal

A voice terminal that can support only one appearance of its extension. The single-appearance voice terminnals include: Models 2500, 7101A, 7102A, and 7103A.

SLS

See **Straight Line Set.**

Soft Hold

One of the two forms of hold. Because soft held calls can be lost in certain situations, soft hold is less durable than hard hold. Soft hold is only available on single-appearance voice terminals.

Soft Processor Swap

An attribute of a duplicated System 85. During a "soft swap," the switch gracefully shifts control from one processor to the other "healthier" processor. Since the current call-status information is transferred to the new processor, soft processor swaps are usually not noticed by users. (See also **Hard Processor Swap.**)

Software

A set of computer instructions that accomplish one or more tasks.

Software Defined Network

An AT&T Communications service that provides ***virtual*** private-network routing through the AT&T Switched Network. Using this service, an ETN tandem (or an intelligent ETN main) in a customer's private network delivers seven private-network digits (RNX + XXXX) to a serving 4 ESS over a direct- or bypass-access trunk group. In turn, the serving 4 ESS [under the control of an NCP (Network Control Point)] routes the call to a specific receiving 4 ESS. The receiving 4 ESS then delivers (again with direct or bypass access) the seven digits to another private-network switch (with AAR or WCR) where the rest of the routing proceeds.

Special Access

See **Bypass Access, Direct Access,** or **Switched Access.**

Special Dial Tone

See **Recall Dial Tone.**

Special Loop Start Trunk Types

The System 85 trunk types assigned as Type 316 to 350 in Procedure 100, Word 1. This set of System 85 DS1 trunk types is primarily intended for disaster preparedness. However, these trunk types can also be used by a System 85 to communicate with some older COs [equipped with D4 channel bank(s)] that do not use ground start signaling. Since the ground start signaling protocol offers enhanced capabilities over loop start, ground start is the signaling protocol of choice whenever this protocol is supported by the serving CO.

Split

A group of agents (group members) organized to receive calls in an efficient and cost-effective manner. This terminology is generally used in relation with the ACD (previously, EUCD) feature.

Instead of "split," the terms "gate" and "bay" are used by other ACD vendors to refer to a functional grouping of agents.

Split Member

A member of an ACD split (usually referred to as an "agent").

Split Number

Sixty (thirty, before R2 V4) ACD splits can be assigned to the system. In Procedure 026, Word 1, the "split number" (a number between 1 and 60) is used to uniquely identify the individual splits. These same split numbers are used in Procedure 011, Word 1 to specify an ACD split that is used as the final point in a coverage path.

Split Supervisor

The first member of an ACD (or, EUCD) split. Usually an experienced agent who trains new agents, serves as a consultant to the split, and regulates the operation of the split.

Splitting

Separating a caller from an existing connection. Used for attendant call handling.

Stable Call

A call that has already been "set up" by call processing. Stable voice calls are in a steady talking condition. Stable data calls have finished the "hand-shaking" process.

Stage One Dialing

One Stage Dialing used with the ISN Interface feature when placing a call from the ISN to (or through) the System 85. This is done from an ISN terminal (Keyboard Dialing) by entering both the interface access code and the destination address on a single line.

Stage Two Dialing

Two Stage Dialing used with the ISN Interface feature when placing a call from the ISN to (or through) the System 85. Similar to one stage dialing (above) except that the interface access code and the destination code are each entered separately, on separate lines, in response to separate dial prompts (one from the ISN and the second from the System 85).

Standard Serial Interface (SSI)

A communications protocol that operates at 56 Kbps and interfaces the Applications Processor to 400 Series printers and the 500 BCT.

Station Message Detail Recording (SMDR)

A service that records detailed call information on incoming and outgoing calls, and associates these calls with account codes.

Statistical Multiplexing

A form of time-division multiplexing.

Status Information

Data that defines the current state of call processing within a switching system. In a duplicated switch, status information is transferred to the healthier processor during a soft processor swap. However, during a hard processor swap, status information is lost.

Status Lamp

An indicator lamp showing the status of an appearance by the state of the lamp (lighted, flashing, fluttering, or dark).

Stored Program Control

Software programs controlling system operation.

Straight Line Set (SLS)

A single-appearance voice terminal that ***appears*** to System 85 as a multiappearance voice terminal. Whenever single-appearance voice terminals are used as ***signaling*** terminals, the Terminal Busy Indication feature requires that these voice terminals are administered as straight line sets. Also, whenever single-appearance voice terminals are given bridging capabilities, the Bridged Call feature requires that these voice terminals are administered as straight line sets.

String

See **Digit String.**

String Identifier

A term used with the WCR (World Class Routing) feature. String identifiers are the elements of a digit string that uniquely identify that digit string. A string identifier can be from 1 to 18 digits long and specifies string length (of the digit string), string type, and call processing actions to be taken when the digit string is identified.

Subscriber

A voice terminal user who is authorized to access the AUDIX system to send and recover messages.

Subnetwork Trunking

Also referred to as Subnet Trunking. This is a technique used with intelligent switches that routes network calls based on the destination rather than a fixed path. This process is usually dynamic in that call routing can be changed after the call has left its point of origin. For example, if a call is processed using a 7-digit private network address and then during the route selection process it is determined that a public network route (requiring a 10-digit address) must be used, the address must be changed to conform to the route selected. The process of restructuring the address digits to conform to the route selected is subnetwork trunking. Features that use this technique include AAR, ARS, WCR and Extension Number Portability.

Subtending

A subtending switch is a lower-order switching system in the network hierarchy.

Switch

The System 85 or Generic 2 switching system; usually used to refer to the System 85 or Generic 2 as distinguished from adjuncts (such as, APs and AUDIX adjuncts).

Switch Administrator

An executive responsible for specifying features and / or services available to users of the switch.

Switched Access

A method of connecting to public telecommunications facilities whereby a customer (terminal or PBX) is provided with indirect physical connections to a long-distance carrier that employ dynamically ***switched*** connections through the LEC (local exchange carrier).

Switched Digital Communications Protocol Interface (SDCPI)

An interface that physically connects between the PDM and the SDCPI interface ports on the AP. Its primary function is to provide the system with a 64 Kbps link for the downloading of AP software into the 515 BCT.

Switched Loop Operation

An automatic system in which an incoming call is switched to an idle loop on an available attendant console.

Switchhook

The button(s) on a voice terminal that are located under the handset

Synchronous Data Transmission

Transmission in which the data characters and bits are transmitted at a fixed rate with the transmitter and receiver synchronized (same clock). This method of transmission eliminates the need for start-stop elements and provides greater efficiency. For comparison see ***Asynchronous Data Transmission.***

System Administrator

An executive responsible for specifying features and / or services available to system users.

System Management Terminal (SMT)

An administration device which is similar to the MAAP. The SMT provides limited administration capability to the customer.

System Manager

The administrator of a partitioned switch. The system manager of a System 85 is responsible for sharing the services of the switch among the various tenants.

System Status Indicator

A lamp on an SSI panel that indicates the busy / idle condition of a Release Link Trunk. System Status Indicators can also monitor the queue warning status for ACD (previously, EUCD, UCD, or DDC) splits.

Tail-End Hop Off

A service provided by ETNs (Electronic Tandem Networks) and TTNs (Tandem Tietrunk Networks) which allows calls destined for an end point on the public network to route part way over private network facilities. With tail-end hop-off, a public network call is routed over private network trunks to a node on the private network that is close to the final destination of the call. At the final private network node, the call "hops off" the private network and enters the public network for final routing. Tail-end hop off is provided by the ARS (Automatic Route Selection) feature and the WCR (World Class Routing) feature.

Tandem Access Avoidance (TAA)

See **Tandem Routing Efficiency**

Tandem Node

An intervening switch between the origin and destination switch in a network. In a network, a tandem node receives signals from one switch and passes these signals on to another switch.

Tandem Tie Trunk Network (TTTN)

A private network where several customer switching systems are interconnected by dial repeating tie trunks.

Tandeming

The act or process of passing a signal through a network node.

Tandeming Switch

For Look-Ahead Interflow, a private-network switch (with Look-Ahead Interflow assigned) that first accepts and receives a Look-Ahead Interflow call from the sending switch. Then, based on the vector programming within its receiving vector, this switch decides to interflow the same call on a Look-Ahead basis to another receiving switch.

Tap Button

Another term for "recall button" that is used by some other switch vendors. After a recall button press (or tap button press) the switch returns recall dial tone to the voice terminal user. Throughout this manual, the term "recall button" is used in preference to "tap button."

Terminal Change Management (TCM)

An administration tool based on the AP 16, available to switch administration via a business communications terminal (BCT). This tool is used for feature assignment and terminal rearrangement.

Terminal Equipment Location (TEL)

The hardware coordinates for the physical location of circuits on the switch that support a specific terminal; most specifically a voice terminal. ***Synonym:*** Equipment Line Location (ELL).

TCM

See **Traveling Class Mark** or **Terminal Change Management.**

Temporary Bridged Appearance

A function of the Call Coverage feature. This function allows a principal with a multiappearance terminal or a straight line set to bridge onto a coverage call by going off-hook on the appearance that was redirected to coverage. (A temporary bridged appearance is also known as a "simulated bridged appearance.")

Terminating/Originating Appearance

One of the two kinds of appearances assigned in Procedure 052, Word 1: terminating/originating and originating (only). When a terminating/originating appearance is idle, calls can either be received or placed using the appearance.

Text Editing, Management and Processing in an Enhanced Synchronized Terminal (TEMPEST)

A subset of AP's Function support Programs that is transported, via a downloading process, from the hard disk located in the AP into the BCT memory.

Tie Trunk

A telecommunications channel connecting two switching systems within the ***private*** network. This term is used in this manual for private network trunks only. Public network trunks are referred to by the public network function (such as, CO trunks, FX trunks, toll trunks, etc).

Time-Division Multiplexing

A type of multiplexing that divides the transmission channel into successive time slots.

Time Multiplexed Switch (TMS)

A time-shared digital space switch that interconnects modules in a multimodule System 85.

Time Slot Interchanger (TSI)

A processor that operates under the control of the Module Processor. The TSI performs the switching, digital attenuation, and conferencing functions for the switching network. A maximum of 256 simultaneous port-to-port connections can be supported by the TSI.

Timed Reminder Tone

A 1950-hertz tone provided by System 85 as part of the Timed Reminder feature. This tone is provided to remind an attendant that a call has been waiting or held on the console for longer than 30 seconds.

Toll Table

An attribute of the ARS and WCR features. A toll table is a software array (assigned in Procedure 309, Word 2 for ARS or in Procedure 319, Word 1 for WCR) that identifies the local and toll office codes (for ARS) or digit strings (for WCR) corresponding to the serving CO or other non-toll office for a particular routing preference. (By default, unidentified office codes are considered ***toll*** office codes.)

Tone Detector

An SN255 (or TN748C for universal modules) circuit (identified as Trunk Type 100) that detects call-progress tones for and provides signals to the System 85 during Modem Pooling calls and DS1 AVD calls.

TouchTone

A signaling system where specific tone pairs are used to represent dialing characters (0 through 9, *, and #). Also, known as DTMF (Dual Tone Multifrequency) signaling.

Traffic

The flow of voice and data communications through a switching system.

Transfer Rate of Information Bits (TRIB)

The number of intelligence (information, not control and signaling) bits passing a given point in the data stream per second. If 64,000 bits per second are transmitted and 8,000 of those bits are control and signaling bits, the TRIB rate is 56,000 bps.

Transient Condition

A constituent part of call processing on System 85 that quickly passes into and out of existence (also referred to as a "transient state"). The following are examples of transient conditions a line is ringing; a line is in a dialing state (dialing, hearing dial tone, etc.); the line is receiving Call Waiting, Attendant Call Waiting, or Priority Calling tones; a line is receiving busy, intercept, reorder, or ringback tone.

Translation Information

Tabular data that defines durable (but changeable) criteria for the switch. These criteria include feature options, trunk and line addresses, button assignments, extension numbers, dial access codes, and some feature activations within a switching system. In a duplicated switch, translation information is stored and updated in both switch processors.

Traveling Class Mark (TCM)

Information sent to a distant switch with the outpulsed digits. There are two separate types of traveling class marks. The first type of TCM provides the FRL (Facility Restriction Level) to the distant (receiving) switch to determine if access to a particular route or facility is permitted. If one TCM is present, it will be this type. If two TCMs are present, this will always be the first TCM. The second TCM (if used provides the Satellite Hop Count to the distant switch.

TRE

See **Tandem Routing Efficiency.**

Tributary Location

An attended location in a Main / Satellite / Tributary complex that has an unique listed directory number and accesses the public and private network directly or through the main switch.

Trunk

A communications channel between two switching systems.

Trunk Data Module (TDM)

A digital interface device that provides the interface for DCP (Digital Communications Protocol) connections to trunk facilities. It is used between off-premises private line trunk facilities and the local switch, and for connections to DDD modems as the DCP member of a Modem Pooling conversion resource. The TDM provides conversion between the RS-232 protocol and the DCP protocol.

Trunk Group

Telecommunications channels assigned as a group for certain functions.

Trunk Group-Oriented Routing

A method of routing calls into a PBX. Using trunk group-oriented routing, a call is recognized by the serving switch (usually, the serving CO) as a call that is routed to the local PBX over a specific trunk group. In turn, the local switch accepts the call from over the trunk group, and recognizes that the call completes to a specific destination (such as, the attendant queue, an ACD split, or a VDN).

Trunk Port

The hardware providing an access point to the switch network for each circuit associated with a trunk.

Trunk Vectoring

Another term for "Call Vectoring" that is used by some other ACD vendors. Call Vectoring (or the similar concept, Trunk Vectoring) provides an enhanced method of defining calling-party interfaces for incoming calls (usually to an ACD). In this manual, the term "Call Vectoring" is used in preference to "Trunk Vectoring."

UNIX Operating System

A versatile time-sharing software operating system for data processing equipment.

Universal Attendant Code

The digits "0111" used for routing calls to an attendant serving a different ETN switch. An ETN tandem recognizes the digit string "RNX + 0111" as an attendant-seeking call destined for the ETN switch identified by the dialed RNX. To continue routing the call, the ETN tandem's AAR pattern can either select a private-network trunk group or, with proper subnetwork trunking assignments, overflow the attendant-seeking call to the public network.

Universal Data Module (UDM)

A generic term for at data module that can function as both a DCE and a DTE.

Universal Module

A 5-carrier, single cabinet call processing module available with Generic 2 that replaces the traditional multicabinet module used with System 85, Release 2. The universal module uses high density circuit technology to provide increased circuitry in an single cabinet. The universal module is sometimes referred to as the hybrid module.

Unmeasured Agent

A member of an ACD split whose activity is not measured by CMS (Call Management system).

Unmeasured Split

A group of ACD agents whose activity is not measured by CMS (Call Management system).

Unstable Condition

See **Transient Condition.**

Vector

An attribute of the Call Vectoring feature. A vector is comprised of a discrete set of predefined call-processing steps that specifies a way of treating certain incoming calls.

Vector Directory Number (VDN)

A "soft" extension number that is assigned an internal line number (but not assigned to an equipment location). A vector is assigned to a VDN (or group of VDNs) to specify the call-processing sequence according to the dialed number. Each VDN can be included as part of a published number for public access to a vector's call-processing sequence.

Virtual Circuit

The entire path between two end processors, which may consist of more than one communications link. In the DCIU application, the term "data path" is used instead of virtual circuit.

Virtual Nodepoint Identifier (VNI)

An index number that represents the destination of a call, used with the World Class Routing feature. A VNI is assigned to a call during network digit analysis, based on administered values, and is not a part of the dialed number (even if it is the same as the office code that is part of the dialed number).

Voice Coupler

An electrical isolation device used for passing audio signals into a system.

Voice Terminal

A single-appearance or multiappearance telephone.

Warning Tone

A tone applied by the local switch (PBX) to advise users that a change in call routing or connected party presence is about to take place. System 85 and DEFINITY Generic 2.1 switches can be administered to return warning tone when a call would unexpectedly route over higher-cost toll facilities. On Generic 2.2 switches, warning tone can be administered to apply to a wider variety of call routing cases. The features where the switch can provide this type of warning tone include: AAR (Automatic Alternate Routing), ARS (Automatic Route Selection), and WCR (World Class Routing).

Another type of warning tone, also called "intrusion tone," is applied to alert talking parties to the presence of an unexpected party within the connection. The features where the switch is able to provide this type of warning tone include ACD (Automatic Call Distribution), Busy Verification of Lines, EUCD (Enhanced Uniform Call Distribution), Override, Trunk Verification—Attendant, and Trunk Verification—Voice Terminal.

WATS (Wide Area Telecommunications Service)

A service that allows users to place toll calls for a flat monthly charge. Specific charges are based on the size of the calling area (WATS band) and on the amount of expected usage.

WATS Trunk

A telecommunications channel used to place a WATS call.

Wild Card Digit

A term used with the WCR (World Class Routing) feature to refer to a leading digit or group of digits in a dialed number that are not used for string identification purposes. That is, a wild card digit represents any possible digit in its digit position. A wild card digit is represented by the asterisk character (*) and must be one or more leading digits. Wild card digits cannot be intervening digits (that is, they cannot occur between groups of specific digits), nor can they be trailing digits.

Wink Start Signaling

A trunk-signaling protocol that uses a "wink" (on-hook to off-hook and return to on-hook) that lasts from 0.14 to 0.29 seconds to indicate that the receiving (far end) switch is ready to receive digits. See Appendix F, Enhanced Trunking, for more information on trunk-signaling protocols.

Work-Related Call

Calling activity by an ACD agent that serves to remove the agent from the agent queue. When outgoing calling is deemed to be work related, removal of the agent from the queue can be provided for outgoing calling.

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