AT&T

DEFINITY® Communications System Generic 1

and

System 75 and System 75 XE

Feature Description

555-200-201
Issue 6, June 1990

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Contents

Figures

Tables

CHAPTER 1. INTRODUCTION

CHAPTER 1. INTRODUCTION

This manual provides a technical description of the system features and parameters for the following systems:

- DEFINITY® Communications System, Generic 1
- System 75 (V1, V2, and V3)
- System 75 XE (V2 and V3).

Information in this document applies to all versions and types of the listed systems unless **specifically noted otherwise** in parentheses.

Also, unless otherwise noted, any information noted as V2 or V3 also applies to System 75 XE V2 or System 75 XE V3, respectively.

The terms "DEFINITY Generic 1" and "G1" refer to the DEFINITY Communications System Generic 1.

Purpose

This manual, along with *AT&T System 75—System Description,* 555-200-200, the *AT&T System* 75 *XE—System Description,* 555-201-200, and the *DEFINITY Communications System Generic 1—System Description,* 555-204-200, is intended to serve as an overall reference for the planning, operation, and administration stages of the system. It is also intended to be used with *AT&T System 75—Implementation,* 555-200-650 for V1, 555-200-651 for V2, 555-200-652 for V3, and *DEFINITY Communications System—Implementation,* 555-204-654 for DEFINITY Generic 1, for software initialization and subsequent changes in feature assignments.

This issue replaces all previous issues of this document. The reason for reissue is to include enhancements to the DEFINITY Communications System Generic 1, as well as other miscellaneous information, including a description of the Ringer Cutoff feature.

The new features added to DEFINITY Generic 1 include the following:

- Basic Call Management System
- Loudspeaker Paging Access—Deluxe
- Names Registration
- Outbound Call Management
- Report Scheduler and System Printer.

Other important additions include enhancements to the Property Management System Interface feature and enhancements to the Attendant Direct Extension Selection With Busy Lamp Field feature.

Organization

The rest of this manual is divided into seven chapters as follows:

- Chapter 2—Functional Description—Provides a general description of the functions and services provided with the system. These functions and services are divided into six groups. The six groups are Voice Management, Data Management, Network Services, System Management, Hospitality Services, and Call Management. Each group of functions and services is described separately and includes a list of associated features. The listed features are fully described in Chapter 3 of this manual.
- Chapter 3—Feature Descriptions—Provides a detailed description of the features associated with Voice Management, Data Management, Network Services, System Management, Hospitality Services, and Call Management. These feature descriptions are arranged in alphabetical order, regardless of function area.
- Chapter 4—System Parameters—Provides information relating to overall system characteristics and capacities. This chapter includes items that must be considered when planning for system implementation.
- Chapter 5—References—Provides a list of reference documentation. A brief description of each document is included.
- Chapter 6—Glossary—Provides a glossary for the entire manual.
- Chapter 7—Abbreviations and Acronyms—Provides a list of abbreviations and acronyms used in this manual.
- Chapter 8—Index—Provides a permuted index for the entire manual.

An individual Table of Contents is provided for Chapters 2 through 4 of this manual.

CHAPTER 2. FUNCTIONAL DESCRIPTION

Figures

Tables

CHAPTER 2. FUNCTIONAL DESCRIPTION

Overview

This chapter describes the features, functions, and services provided with the system. These features, functions, and services are divided into six groups: Voice Management, Data Management, Network Services, System Management, Hospitality Services, and Call Management. Each group of functions and services is described separately and the description includes a list of associated features. The listed features are fully described in Chapter 3 of this manual.

Voice Management

Overview

The many Voice Management features available with the system allow the individual needs of everyone in the system to be met. As the individual's needs change, the assigned features can also be changed. The Voice Management features provide important services with benefits such as saving time and making calling more convenient.

Voice Management Features

The following features are associated with Voice Management.

Abbreviated Dialing
AP Demand Print
Attendant Auto-Manual Splitting
Attendant Call Waiting
Attendant Control of Trunk Group Access
Attendant Direct Extension Selection With Busy Lamp Field
Attendant Direct Trunk Group Selection
Attendant Display
Attendant Recall
Attendant Release Loop Operation
Audio Information Exchange (AUDIX) Interface (V3 or G1)
Authorization Codes (V3 or G1)
Automatic Callback
Automatic Incoming Call Display (V2, V3, or G1)
Bridged Call Appearance—Multi-Appearance Voice Terminal
Bridged Call Appearance—Single-Line Voice Terminal (G1)
Busy Verification of Terminals and Trunks (V2, V3, or G1)
Call By Call Service Selection (G1)
Call Coverage
Call Forwarding All Calls (V1)
Call Forwarding All Calls (V2, V3, or G1)

Call Park
Call Pickup
Call Waiting Termination
Centralized Attendant Service (V2, V3, or G1)
Class of Restriction
Class of Service
Code Calling Access
Conference—Attendant
Conference—Terminal
Consult
Coverage Callback
Coverage Incoming Call Identification
Dial Access to Attendant
Dial Plan
Direct Department Calling and Uniform Call Distribution
Direct Inward Dialing
Direct Outward Dialing
Distinctive Ringing
Emergency Access to the Attendant (V3 or G1)
Facility Busy Indication
Forced Entry of Account Codes (V2, V3, or G1)
Go To Cover
Hold
Hot Line Service
Hunting
Individual Attendant Access (V2, V3, or G1)
Integrated Directory
Intercept Treatment
Intercom—Automatic
Intercom—Dial
Inter-PBX Attendant Calls (V2, V3, or G1)
Last Number Dialed
Leave Word Calling
Line Lockout
Loudspeaker Paging Access
Loudspeaker Paging Access—Deluxe (G1)
Manual Message Waiting
Manual Originating Line Service
Manual Signaling
Multi-Appearance Preselection and Preference
Multiple Listed Directory Numbers
Music-on-Hold Access
Night Service—Hunt Group (V3 or G1)
Night Service—Night Console Service
Night Service—Night Station Service
Night Service—Trunk Answer From Any Station
Night Service—Trunk Group (V3 or G1)
Outbound Call Management (G1)
Personal Central Office Line
Personalized Ringing (V2, V3, or G1)

Power Failure Transfer
Priority Calling
Privacy—Attendant Lockout
Privacy—Manual Exclusion
Recall Signaling
Recorded Announcement
Recorded Telephone Dictation Access
Remote Access
Restriction—Controlled
Restriction—Miscellaneous Terminal
Restriction—Miscellaneous Trunk
Restriction—Toll/Code
Restriction—Voice Terminal—Inward
Restriction—Voice Terminal—Manual Terminating Line
Restriction—Voice Terminal—Origination
Restriction—Voice Terminal—Outward
Restriction—Voice Terminal—Termination
Ringback Queuing
Ringer Cutoff
Rotary Dialing (V2, V3, or G1)
Send All Calls
Senderized Operation
Single-Digit Dialing and Mixed Station Numbering (V3 or G1)
SMDR Account Code Dialing
Straightforward Outward Completion
Temporary Bridged Appearance
Terminating Extension Group
Through Dialing
Timed Reminder
Touch-Tone Dialing
Transfer
Trunk Group Busy/Warning Indicators to Attendant
Trunk Identification by Attendant (V2, V3, or G1)
Trunk-to-Trunk Transfer
Voice Message Retrieval (V2, V3, or G1)
Voice Terminal Display.

Data Management

Overview

DEFINITY Generic 1 and System 75 are private digital switching systems that permit connections with a variety of data equipment. Data terminals, printers, graphics, facsimile equipment, and computers can be connected to the switch through various protocols or interfaces.

The physical connection can be through a digital data module or analog modem.

The system provides the customer with options for selecting data modules (or data-like devices such as a Data Line Circuit [DLC] [V2, V3, G1]) for Terminal Dialing. Also, customers can use data modules without Terminal Dialing with host computers, printers, or other such applications. Computer file transfer at a speed rate of 64 kbps is possible with the Modular Processor Data Module (MPDM) and the Modular Trunk Data Module (MTDM).

The family of data modules also includes a Processor Data Module (PDM), a Digital Terminal Data Module (DTDM), a Modular Data Module (MDM), a Trunk Data Module (TDM), and a 3270 Data Module. The data modules are generally more versatile than modems, operate at faster data rates, and provide additional features.

The 7404D voice terminal has a built-in data module that allows the voice terminal to control and be connected to a data terminal. Data calls can be originated or disconnected using the key pad of the attached data terminal. Voice calls can be made or received while a data call is in progress. The 7406D and 7407D voice terminals have the same data feature functions as the 7404D, through the use of an optional base containing a data module.

The AT&T Personal Terminal 510D, which operates in alphanumeric and graphics character set mode, provides the equivalent of a Model 7405D voice terminal equipped with a DTDM, a 513 Business Communications Terminal (BCT), and a Digital Display Module.

The 515 BCT has the same video display and keyboard features as the 513 BCT. In addition, it provides voice terminal functions and the functional equivalent of a Digital Display Module.

The 510D terminal or 515 BCT provides an all-digital interface with the system. Through its built-in Electronic Industries Association (EIA) RS-232C interface, the 515 BCT can connect to other data equipment.

The DLC, which provides eight ports to connect user’s asynchronous EIA RS-232C interface to Data Terminal Equipment (DTE), can be used as an alternative to DTDM or PDM.

All data modules except the MPDM and 3270 provide a modified RS-232C interface. The MPDM provides either RS-232C V.35 or RS-449 interface. The MPDM can also emulate an automatic calling unit (ACU) and supports the RS-366 interface. The ACU emulation and RS-366 interface are required for Keyboard Dialing, which is discussed in the Data Call Setup feature description in Chapter 3. The 3270 Data Module provides a Category A coaxial Digital Communications Equipment (DCE) interface for connection to 3270-type data terminals or a cluster controller. It also provides a Digital Communications Protocol (DCP) interface for connection to the digital switch.

The 3270 Data Module is available in the following three models:

- 3270T (Terminal)—Connects to a Category A 3270-type terminal, such as the IBM* 3278 Information Delivery System. The 3270T Data Module must connect through the switch to a 3270C (Controller) Data Module.
- 3270A (Asynchronous)—Provides the same function as the 3270T Data Module. It also allows the 3270-type terminal to emulate a Digital Equipment Corporation VT100† or an AT&T asynchronous terminal.
- 3270C (Controller)—Connects an IBM 3274 or 3276 cluster controller to the switch. A 3270C Data Module can contain as many as eight ports.

The system supports digital-to-digital, digital-to-analog, analog-to-digital, and analog-to analog data calls. For data calls, the user can access the system through these digital or analog data endpoints. Digital data endpoints are data modules and associated data equipment, 510D terminals or 515 BCTs, data channels (used for remote System Access Terminals [SATs] [V1, V2, and V3] or DEFINITY Manager I terminals [G1], and Station Message Detail Recording [SMDR]), and/or the Applications Processor (AP) interface (inside the system). Analog data endpoints are modems (or acoustic coupled modems) and associated data equipment connected to the system through analog lines or trunks.

The system supports a Digital Communications Protocol (DCP). This protocol provides framing, control, and signaling for each of two information channels. Only one channel is used for voice-only or data-only applications. Both channels are used for simultaneous voice and data transmission. Simultaneous voice and data information can be transmitted on calls to or from a 510D terminal or 515 BCT, a 7403D or 7405D voice terminal with a DTDM, a 7404D with its built-in data module, and a 7406D or 7407D with an optional data module base. Calls to or from other equipment are either voice-only or data-only.

Data Networking

Data networking connects two or more data endpoints. The system is a highly reliable, centralized switch that provides switched access between endpoints. Typical data communications configurations for the system are shown in Figure 2-1.

Switched access allows one terminal to connect to any number of devices. Therefore, more effective use of data equipment is obtained than with dedicated (hard-wired) links. Switched access also reduces the need for duplicated (dedicated) equipment.

The system uses twisted-pair standard building wiring and 8-pin modular wall jacks. Each wall jack is a single outlet that can handle simultaneous voice and data information.

* Registered trademark of International Business Machines Corporation

† Trademark of Digital Equipment Corporation

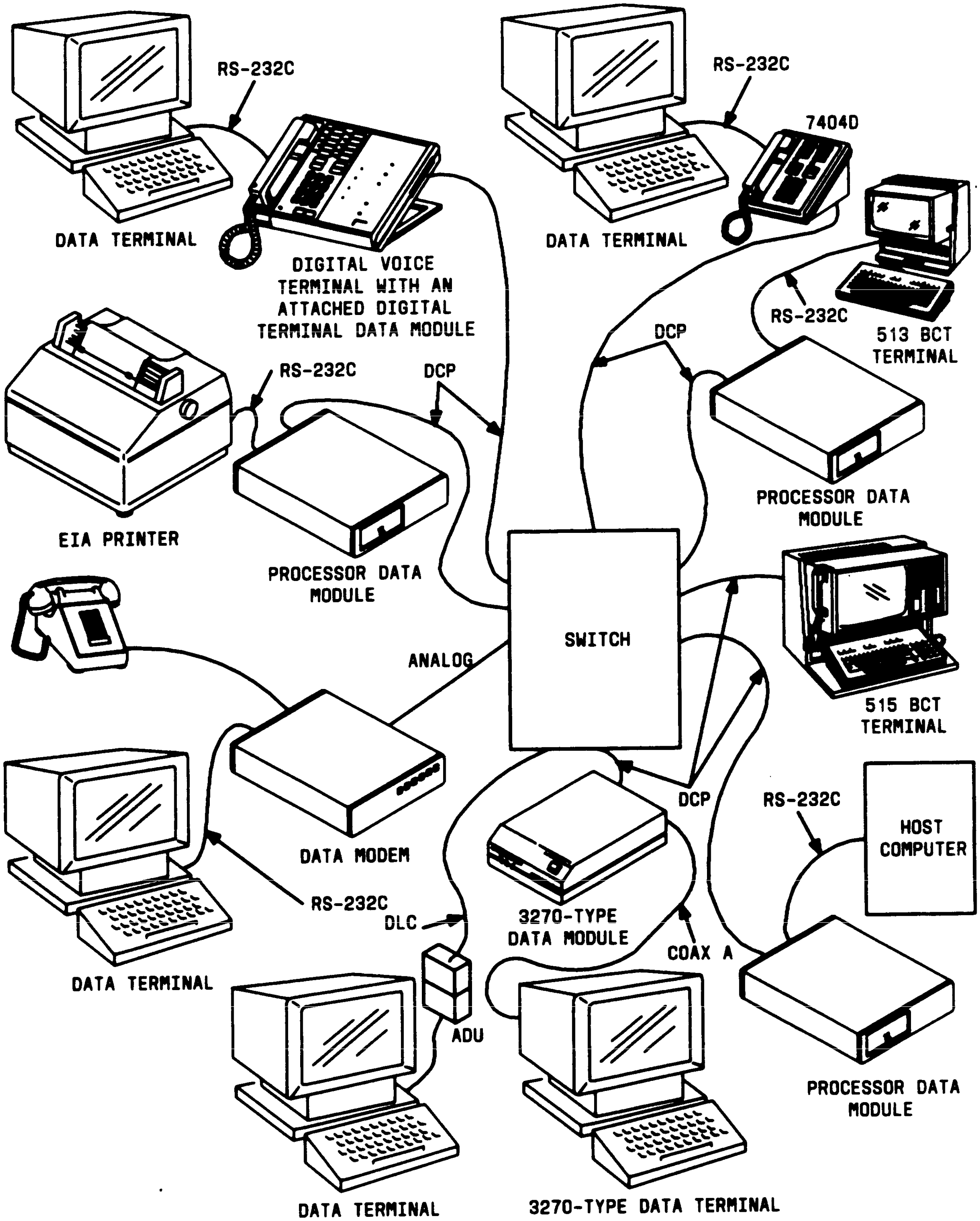


Figure 2-1. System Data Communications Configuration

The digital switch, data modules, DCP, twisted-pair wiring, modular wall jacks, and switched data features give the system its unique capabilities. These capabilities merge the business office data processing and telecommunications functions into a single system.

Generally, data networks are either local area networks, extended networks, or combinations of both. The two networks and their implementation within the system are defined as follows:

- **Local Area Networks**

  The system provides this capability by connecting communication devices that are physically located within a local-area or campus-like environment. These include conventional, semi-intelligent, and intelligent data terminals, personal computers, host computers, and virtually any device with the proper communications interface.

  The centralized network provides circuit switched paths using twisted-pair building cable that extends to the endpoints. Since the business office equipment can access multiple data systems, the data equipment and applications can be used more productively. The system also provides several data-related features that are easy to use and that contribute toward expedient use of the system and its networking capabilities.

- **Extended Networks**

  Extended networks mainly provide connections between the system and other distant switches, including remote access facilities. Through use of remote access facilities, a local terminal can access remote host computers. Also, remote terminals can access either local computer facilities or other remote computer facilities. Extended networks are constructed of analog or digital facilities and can be either public or private. Typical networking configurations are shown in Figure 2-2.

  Public networks include:

  — Local central office (CO) switching extended through direct distance dialing

  — Foreign exchange (FX) central office trunking

  — Wide Area Telecommunications Service (WATS)

  — MEGACOM™ Telecommunications Service

  — MEGACOM 800 Telecommunications Service

  — Software Defined Network (SDN)

  — ACCUNET® Digital Service.

  Private networks include:

  — AT&T DATAPHONE® Data Communications Service

- Distributed Communications System (DCS)
- Electronic Tandem Network (ETN)
- Enhanced Private Switched Communications Service (EPSCS)
- Private line (PL)
- Tandem tie trunk.

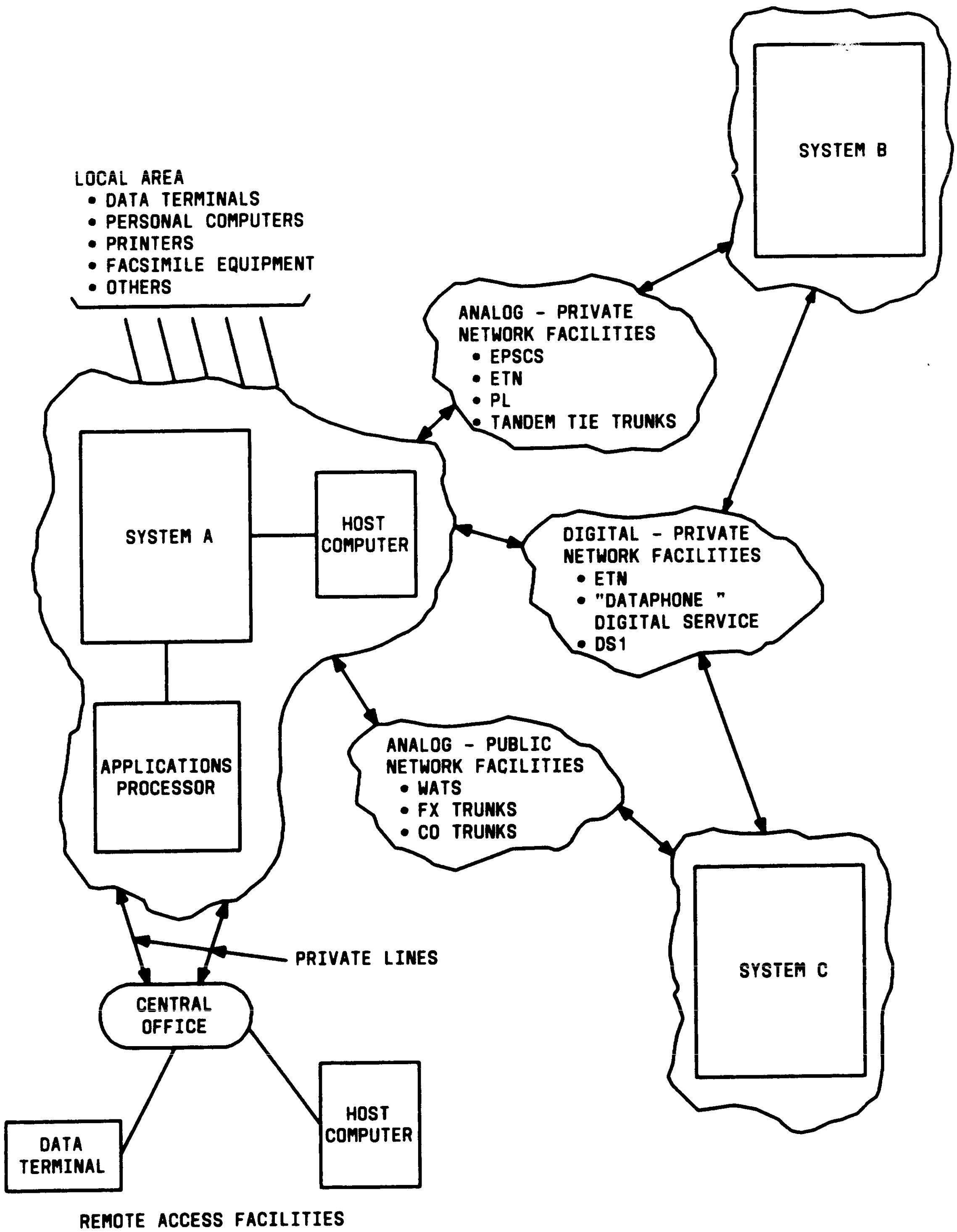


Figure 2-2. System Networking Configurations

Data Management Features

The following features are associated with Data Management:

Data Call Setup
Data Hot Line (V2, V3, or G1)
Data-Only Off-Premises Extensions
Data Privacy
Data Restriction
Digital Multiplexed Interface (V2, V3, or G1)
DS1 Tie Trunk Service (V2, V3, or G1)
EIA Interface (V2, V3, or G1)
Information System Network (ISN) Interface
Modem Pooling
PC/PBX Connection
Permanent Switched Calls (V2, V3, or G1)
Uniform Call Distribution (UCD).

Data Communications Protocols and Interfaces

Overview

A protocol is a set of conventions or rules that governs how data is transmitted and received. The rules generally cover such subjects as the following:

- Physical interface
- Mechanical interface
- Electrical interface
- Framing
- Error detection and control.

Communications protocols are designed to meet the transmission requirements for specific data exchange and data communications equipment. These communications protocols are sponsored by a national or international organization or a major corporation. The system equipment, Applications Processor (AP), and communications processing software provide the following protocols:

- RS-232C
- RS-449
- RS-366
- Standard Serial Interface (SSI)

- Teletypewriter (TTY) Modes
- Digital Communications Protocol (DCP)
- BX.25 Packet Switching
- International Telegraph and Telephone Consultative Committee (CCITT) V.35
- Binary Synchronous Communications (Bisync).

Electronic Industries Association (EIA)

RS-232C

This protocol is widely used for short distance and low-speed applications such as data terminals and modems connecting data terminals. The data link consists of a 25-conductor cable. The conductors are used for data-link control and timing, as well as for transmitting and receiving signals. Data-link control is accomplished by handshake signaling between the transmit and receive devices. Data speeds are limited to 19.2 kbps or less.

The RS-232C protocol provides two interface connectors. The female side connector is known as data communications equipment (DCE). The male side connector is known as data terminal equipment (DTE). Data equipment manufacturers design either the DCE or DTE interface into their products. Products such as modems, data service units (DSUs), Digital Terminal Data Modules (DTDMs), and Processor Data Modules (PDMs) have a built-in DCE interface. Products such as some types of multiplexers, data terminals, printers, computer ports, and Trunk Data Modules (TDMs) have a built-in DTE interface. Modular Data Modules (MDMs) can be configured as either DCE or DTE.

The maximum cable length recommended by EIA for the RS-232C protocol is 25 feet (15 meters). However, practical applications have shown that the cable length can be much greater. Factors limiting cable length include transmission speed, cable capacitance, and nearness of noise sources such as fluorescent lights or electric generators. Each application should be considered separately.

RS-449

This protocol allows longer cables than the RS-232C. Maximum cable lengths for various data speeds are as follows:

- 19.2 kbps—200 feet (61 meters)
- 9.6 kbps—400 feet (122 meters)
- 4.8 kbps—800 feet (244 meters)
- 2.4 kbps—1600 feet (488 meters).

The RS-449 protocol is provided as a communications link interface on the AP. This standard uses a 37-conductor cable. The AP RS-449 interface contains unbalanced driver/receivers that also permit interconnection to the RS-232C interface when used with a 37- to 25-pin cable adapter. Since the AP RS-449 interface is compatible with the RS-232C protocol, it also is limited to the same maximum 19.2 kbps data rate.

RS-366

The RS-366 communications protocol specifies the standards for interfacing computers to automatic calling units (ACUs). This permits a computer to originate data calls over a switched telephone network. The AP provides one RS-366 interface for each six RS-232C interface ports.

AT&T

Standard Serial Interface (SSI)

The SSI communications protocol is used with the 500-series Business Communications Terminals (BCTs) and 400-series printers. The interface operates full-duplex, in synchronous mode, at 56 kbps, and over 24-gauge standard building cable at distances up to 5000 feet (1524 meters). Cable connections are made through the 8-pin modular-type connectors.

Teletypewriter (TTY) Modes

The AP EIA RS-232C interface ports support the TTY protocol. This protocol is implemented as software within the AP's EIA terminal or port subsystem. The protocol permits each port to operate in either the transparent or TTY mode.

- Transparent Mode

  When operating in the transparent mode, the ports pass American Standard Code for Information Interchange (ASCII) characters between the AP and terminal device unchanged. Incoming characters can be echoed back to the terminal device as they are received. However, no recognition of control characters is provided. The BREAK character is the only special character that can be recognized. The following options are available:

  — Parity (enable and disable, even and odd)

  — Data rate—less than 300 up through 19,200 bps

  — Stop bits—1, 1-1/2, or 2 bits

  — Local—assume line with or without modem control

  — Character size—5, 6, 7, or 8 bits plus parity bit

  — Echo—on and off.

Some of these options are pre-coded by the applications software and cannot be changed by the voice terminal user.

- TTY Mode

When operating in the TTY mode, the EIA interface port acts as both the pre-processor and post-processor between the terminal and the AP applications software. In addition to all options listed under the transparent mode, several ASCII control characters are recognized.

A variety of control (delay) options are available to interface with different types of EIA-compatible printers. The ASCII characters DEL and NUL are used for fill (delay) characters. Termination options are provided for line control of modems and ACUs. The TTY mode also provides several mapping options.

The AP applications software determines the mode (transparent or TTY) and the options within each mode that are implemented per EIA channel. The methods for selecting EIA channel parameters are provided through option designation display forms or by default. When display forms are provided, they are an integral part of the applications software.

Digital Communications Protocol (DCP)

The DCP is used by the system's digital switch, digital voice terminals, data modules, the 510D terminal, and the 515 BCT. This protocol permits simultaneous voice and data over the same communications link to the switch.

The DCP consists of a 160-kbps, 4-wire serial data link that operates full-duplex over standard twisted-pair building cable. For data-only transmission, the maximum cable length is 5000 feet (1524 meters). When voice and data transmission is carried over the same data link, as when a 510D terminal, 515 BCT, or a DTDM is used, the cable length is limited by the voice transmission distance.

The DCP sends digitized voice and digital data in frames. Each frame consists of four fields or channels (see Figure 2-3). The first field is a unique 3-bit framing pattern that defines the frame boundary. The second field is a 1-bit control or signaling channel between the digital switch and digital data endpoint. The third and fourth fields are two independent information (I) channels. The information channels are 8 bits each and are used to send digitized voice or digital data.

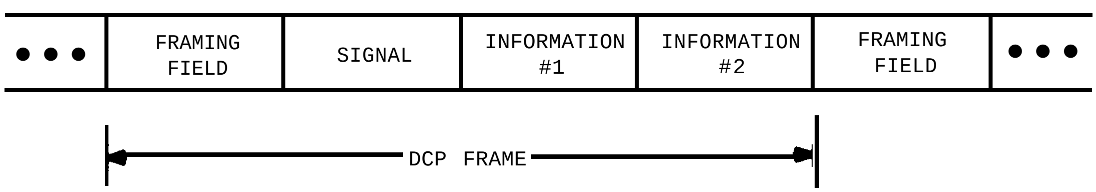


Figure 2-3. Digital Communications Protocol Frame Structure

There are 8000 frames per second. Therefore, the bit rate available is 8 kbps for the signaling channel and 64 kbps for the information channel. The digital switch routes each information channel independently so that simultaneous voice and data can be completed to different destinations.

The full capacity of the information channels (64 kbps) is available for digitized voice. Data terminals typically operate at speeds from below 300 bps up to 19.2 kbps, asynchronous or synchronous. The DCP uses data modules to map the data terminal data into a 64-kbps information channel.

The framing rate of 8000 per second and 8 bits per information channel is consistent with other telecommunication systems such as the T1 carrier. This minimizes potential conversion problems when interfacing to different digital facilities.

BX.25 Packet Switching Protocol

The BX.25 protocol implements the international standard for packet switching. It is a multilayered protocol. (Layering is a structuring of specific protocol functions [for example, error detection and correction] that are grouped together as a unique layer or level.)

The BX.25 protocol is similar to the CCITT X.25 protocol and, from a user perspective, is compatible with the standard. The BX.25 protocol has three layers which are not specified for the X.25 protocol. These layers are the Application, Presentation, and Session layers. The Application and Presentation layers (see Table 2-A) are defined in the Transaction Oriented Protocol (TOP) of the BX.25.

The TOP is a high-level protocol, intended to standardize communications between transaction-oriented systems. Transaction-oriented communications involve communication of small messages or requests describing a single unit of work, which may result in a reply being sent back to the originating system. The Session layer is intended to establish, manage, and terminate sessions for use by higher level protocols or, in some cases, by user applications directly. Other differences between X.25 and BX.25 are as follows:

- The X.25 protocol specifies network standards only; the BX.25 protocol places requirements on the user interface as well.
- The X.25 protocol provides for datagram services while the BX.25 protocol does not. Datagram service has not been implemented within the continental United States.
- The X.25 protocol leaves the users in a point-to-point environment to develop their own solutions to the following areas of potential conflict, while the BX.25 protocol provides solutions:
  - Link layer addressing
  - Logical channel selection
  - Call collision.

Basic elements of the Application and Presentation layers must be user-defined under both protocols. Table 2-A shows the relationship and similarity between the BX.25 and X.25 protocols.

The BX.25 protocol is used in the system to provide communications between the switch and the AP when the AP provides the switch-related features. The BX.25 protocol is also used in the system to provide communications between the switch and the Audio Information Exchange (AUDIX) and to provide communication between DCS switches.

Table 2-A. Packet Switching Protocols

<table>
<tr><td>USER</td><td colspan="2">USERS</td><td>USER</td><td>USERS</td></tr>
<tr><td>APPLICATION LAYER</td><td rowspan="2">T O P</td><td rowspan="2">*</td><td>APPLICATION LAYER</td><td rowspan="3">Not Specified by Protocol</td></tr>
<tr><td>PRESENTATION LAYER</td><td>PRESENTATION LAYER</td></tr>
<tr><td>SESSION LAYER</td><td colspan="2">SESSION LAYER</td><td>SESSION LAYER</td></tr>
<tr><td>TRANSPORT LAYER</td><td colspan="2" rowspan="2">PACKET LAYER</td><td>TRANSPORT LAYER</td><td rowspan="2">PACKET LAYER</td></tr>
<tr><td>NETWORK LAYER</td><td>NETWORK LAYER</td></tr>
<tr><td>DATA LINK LAYER</td><td colspan="2">LINK LAYER</td><td>DATA LINK LAYER</td><td>LINK LAYER</td></tr>
<tr><td>PHYSICAL LAYER</td><td colspan="2">PHYSICAL LAYER</td><td>PHYSICAL LAYER</td><td>PHYSICAL LAYER</td></tr>
</table>

BX.25 Protocol Layers

X.25 Protocol Layers

International Telegraph and Telephone Consulative Committee (CCITT)

X.25 Packet Switching Protocol

The CCITT is one of three divisions of the International Telecommunications Union, an agency of the United Nations. The standards set by the CCITT generally deal with public networks. Two series of standards or recommendations specifically deal with data transmission:

- The V-series provides recommendations for data transmission over analog or voice telephone networks.
- The X-series provides recommendations for data transmission over digital networks.

The V-series includes the V.10, V.11, V.24, V.28, and V.35. Also, V.26, V.27, and V.28 are modem recommendations for 2400, 4800, and 9600 bps, respectively.

V.10 and V.11 are the equivalent to the EIA RS-423 and RS-422.

V.24 provides the definitions for all interchange circuits that cross the DTE/DCE interface.

V.28 defines a set of electrical characteristics that are compatible with RS-232C.

V.35 provides the constant current interface for 48-kbps operation.

The X.25 protocol is the CCITT recommendation for implementing the International Standards Organizations Reference Model of Open Systems Interconnection which is the international model for packet switching networks. This is a bit-oriented, layered-type protocol. The transport, network, data link, and physical layers (levels) are defined functionally by the CCITT.

The X.25 protocol specifies network requirements and procedures to provide the user interface for a packet switching network. Typically, users generate low-speed asynchronous data. The X.25 software segments this data into packets, adds framing and routing information, and queues the packets into a buffer memory. User data packets, along with the added framing bits, are then transmitted over high-speed carriers. This permits efficient and dynamic sharing of these high-speed data links.

The X.25 protocol provides the communications links between multiple APs.

International Business Machines

Binary Synchronous Communications (Bisync)

Bisync is a character-oriented protocol that provides data transfer, error detection, and error correction. It is widely used for interactive data communications networks.

This is a multilayered protocol. Layering is a structuring of specific protocol functions (for example, error detection and correction) that are grouped together as a unique layer or level.

The Bisync protocol is implemented partly in hardware and partly in software. The physical or hardware level consists of the AP and its associated communications line controller. The line controller has an RS-232C and an RS-449 communications port. Both ports can be used for connection to Bisync-type networks.

The Bisync protocol can be used in either point-to-point or multipoint data link configurations. These network configurations can be either switched or dedicated lines. Generally, the data link operates in half-duplex, synchronous mode at 2.4, 4.8, or 9.6 kbps. Either the ASCII or the Extended Binary Coded Decimal Interchange Code (EBCDIC) can be used.

The AP uses the Bisync protocol in providing 2780/3780 and 3270 terminal emulation features.

Network Services

Overview

Network Services allows a group of switches (consisting of DEFINITY Generic 1, System 75, and/or other systems) to be configured to meet the communications needs of a medium- to large-size corporation. Possible arrangements include an Electronic Tandem Network (ETN), Distributed Communications System (DCS), and Main/Satellite/Tributary. Each is briefly described in this chapter.

Do not assume that the system has any capabilities other than those explicitly stated herein. Refer to *Network and Data Services—Reference,* 555-025-201, for differences between this system and other AT&T systems. (Check *DEFINITY Communications System Generic 1 and System 75 Documentation Guide,* 555-200-010, for the availability of this document.)

Network Services Features

The following features are associated with Network Services:

AAR/ARS Partitioning (V3 or G1)
Automatic Alternate Routing (V2, V3, or G1)
Automatic Circuit Assurance (V2, V3, or G1)
Automatic Route Selection (V1)
Automatic Route Selection (V2, V3, or G1)
DCS Alphanumeric Display for Terminals (V2, V3, or G1)
DCS Attendant Control of Trunk Group Access (V2, V3, or G1)
DCS Attendant Direct Trunk Group Selection (V2, V3, or G1)
DCS Attendant Display (V2, V3, or G1)
DCS Automatic Callback (V2, V3, or G1)
DCS Automatic Circuit Assurance (V2, V3, or G1)
DCS Busy Verification of Terminals and Trunks (V2, V3, or G1)
DCS Call Forwarding All Calls (V2, V3, or G1)
DCS Call Waiting (V2, V3, or G1)
DCS Distinctive Ringing (V2, V3, or G1)
DCS Leave Word Calling (V2, V3, or G1)

DCS Multi-Appearance Conference/Transfer (V2, V3, or G1)
DCS Trunk Group Busy/Warning Indication (V2, V3, or G1)
Facility Restriction Levels and Traveling Class Marks (V2, V3, or G1)
Generalized Route Selection (G1)
Integrated Services Digital Network—Primary Rate Interface (G1)
Network Access—Private
Network Access—Public
Off-Premises Station
Subnet Trunking (V2, V3, or G1)
Ten-Digit to Seven-Digit Conversion (G1)
Time of Day Routing (G1)
Uniform Dial Plan (V2, V3, or G1).

Private Network Configurations

A private network is a configuration of trunk and switching facilities dedicated to the use of a business or organization. It may have as few as two switches or it may have hundreds of switches located throughout the country. (A DEFINITY Generic 1 or System 75 DCS, however, is limited to 64 switches.) Although they normally serve moderate to heavy calling between locations, the following configurations make it possible for organizations of all sizes to realize the benefits of a private network.

- ETN—Serves the needs of customers with many locations in a large geographic area. This configuration provides for calling between locations without accessing toll facilities.

- DCS—Serves the needs of customers with several locations in a small or large geographic area. A Distributed Communications System appears as a single switch with respect to certain features.

- Main/Satellite/Tributary—Serves the needs of customers with a few locations in a small geographic area.

The system also can be used within a Tandem Tie Trunk Network (TTTN). A TTTN is a non-hierarchical network of tie trunks interconnecting three or more switches. User dialing into each switch in the call's path is required. That is, the user at one switch dials the trunk access code for a tie trunk group to another switch, receives dial tone from that switch, and then dials another trunk access code to reach another switch. When dial tone is received from the final (desired) switch, the user dials the desired extension number.

Electronic Tandem Network (ETN)

An ETN is a hierarchical network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations. An ETN consists of tandem switches, the intertandem tie trunks that interconnect them, the access or bypass tie trunks from a tandem switch to a main switch, and the capability to control call routing over these facilities. Figure 2-4 shows a typical ETN configuration. As shown in the figure, a Main/Satellite/Tributary configuration can be served by an ETN. Although not shown in the figure, a DCS can also be part of an ETN.

The system can serve as an ETN tandem switch. However, it's capabilities as an ETN tandem switch are limited.

Within an ETN, each location is identified by a unique private network office code, called an RNX. An RNX never matches an Area Code, so 640 possible RNXs are available for each ETN. After accessing the ETN, the user simply dials the RNX plus the desired extension number, for a total of seven digits.

Public network office codes (NXXs) are unique within an Area Code, whereas RNXs are unique within an ETN. RNXs are assigned when the ETN is established and, for convenience, may match NXXs (although this is not always possible). When Direct Inward Dialing (DID) is provided by the local central office, the extension numbers (last four digits of the number) will match. Network Inward Dialing (NID) is the ETN equivalent of DID and can be provided without DID.

The software program that uses the RNXs to control call routing over an ETN is called Automatic Alternate Routing (AAR). AAR not only determines the route for a call, but, through the Facility Restriction Level (FRL) function, defines up to eight levels of calling privileges for users of the ETN. Another function of AAR, Subnet Trunking, can convert an on-network number to a public network or international number. This function is useful when all on-network routes are busy or are not provided. Details of Automatic Alternate Routing, Facility Restriction Level, and Subnet Trunking are given in Chapter 3.

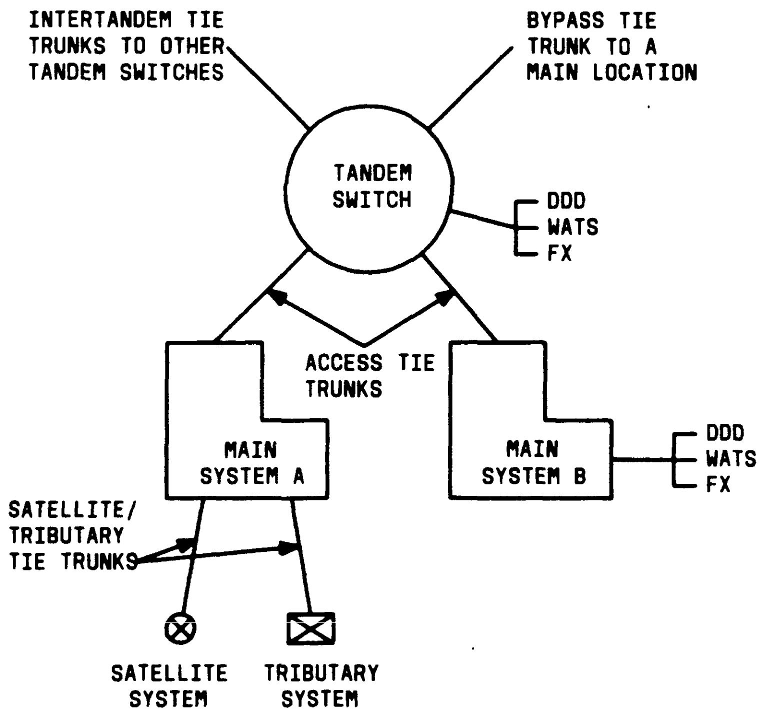


Figure 2-4. Typical ETN Configuration

Distributed Communications System (DCS) (V2, V3, or G1)

A DCS is a cluster of private communications switches (nodes), interconnected among several geographic locations. These switches can be either a System 75, System 85, DEFINITY Generic 1, DEFINITY Generic 2.1, or DIMENSION® PBX. If all nodes are System 75s or DEFINITY Generic 1s, the DCS can have as many as 64 nodes. This maximum number of nodes decreases slightly if the DCS includes a System 85 or DIMENSION PBX. An attribute of a DCS configuration that distinguishes it from other networks is that it appears as a single switch with respect to certain features. This provides simplified dialing procedures between locations, as well as the convenience of using some of the system's features between locations. DCS is particularly attractive if there is frequent interlocation calling.

Each DCS node is connected with every other DCS node by tie trunks for voice communications and data links that send and receive control and feature information. However, each DCS node does not have to be directly connected to every other node. Communication may be through a DCS tandem node. The data links and voice channels may be directly between nodes or may pass through a tandem node. Nodes that cannot serve as a tandem node (that is, those that cannot receive information from one node and pass it on to another node) are called endpoints (or endpoint nodes). Nodes that can pass information are simply referred to as nodes. Within a DCS, a V2 System 75 can only serve as an endpoint. A V3 System 75 or DEFINITY Generic 1 can serve as either an endpoint node or a regular (tandem) node. Figure 2-5 shows a typical DCS configuration.

A DCS can consist of all endpoints. That is, each node in the DCS may be directly connected by data links and voice channels with every other node in the DCS. In this case, System 75 or DEFINITY Generic 1 can serve as all nodes.

Some of the applications of the DCS configuration are as follows:

- In a "campus environment" that has two or more separate buildings and the nodes are connected by local cable
- In a larger area such as a city, several states, or even the entire country, where the nodes are separated by distances too great for local cable and may be connected to different central offices.

A DCS has the property of "transparency" with respect to inside calling and some features. Transparency is the ability of the system, from the user's standpoint, to operate across several nodes in the same way it does at the local node. This allows users to dial from any terminal to any other terminal within the DCS without regard for which nodes are involved. Likewise, transparency allows certain voice features to be used across nodes.

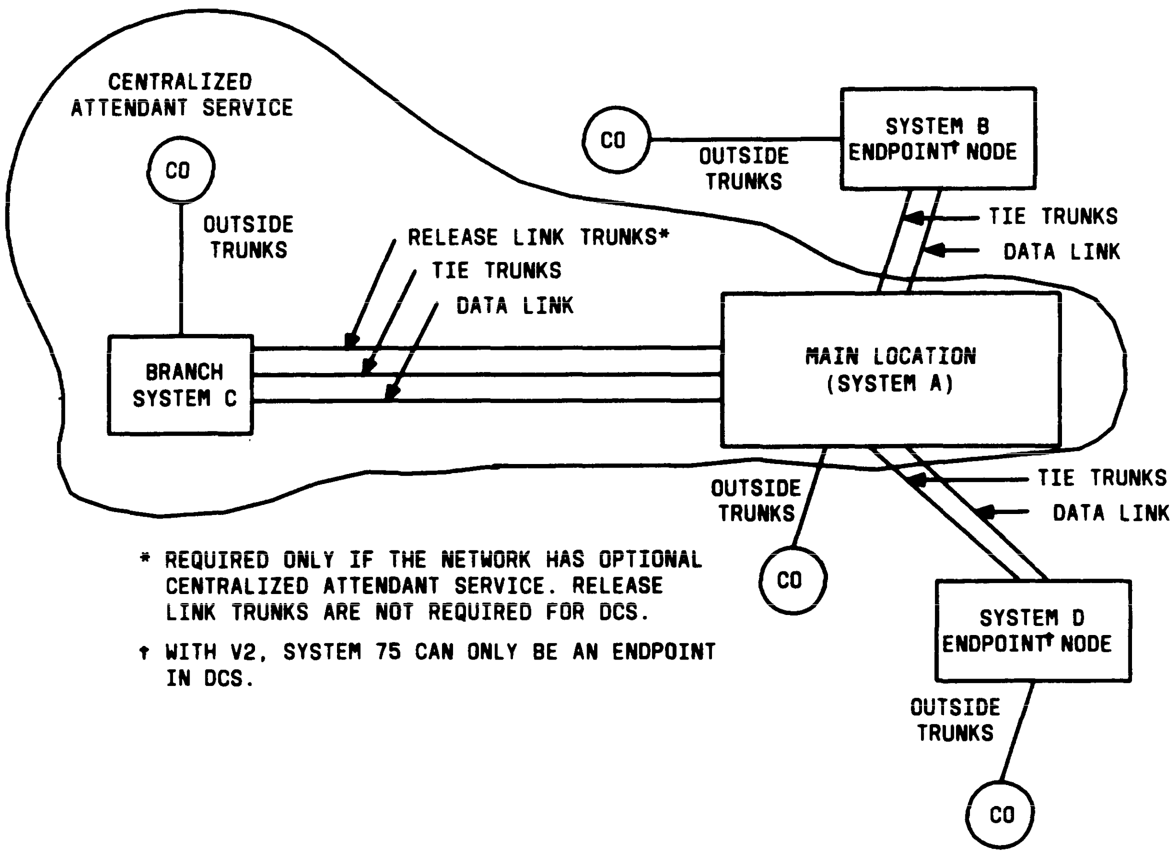


Figure 2-5. Typical Distributed Communications System

Some voice features have transparency in a DCS configuration. The following voice features have unique aspects in a DCS environment and are described in detail in Chapter 3.

- DCS Alphanumeric Display for Terminals
- DCS Attendant Call Waiting (described under DCS Call Waiting)
- DCS Attendant Control of Trunk Group Access
- DCS Attendant Direct Trunk Group Selection
- DCS Attendant Display
- DCS Automatic Callback
- DCS Automatic Circuit Assurance
- DCS Busy Verification of Terminals and Trunks
- DCS Call Forwarding All Calls
- DCS Call Waiting—Termination (described under DCS Call Waiting)
- DCS Distinctive Ringing
- DCS Leave Word Calling
- DCS Multi-Appearance Conference/Transfer
- DCS Priority Calling (described under DCS Call Waiting)
- DCS Trunk Group Busy/Warning Indication.

Abbreviated Dialing and Last Number Dialed also have transparency in a DCS configuration. These features operate the same in a DCS as they do at a single switch.

A DCS cluster can consist of up to 64 nodes. Since the Applications Processor (AP), Audio Information Exchange (AUDIX), and the Call Management System (CMS) each require the same data link facilities as a node, each of these included in the system reduces the number of available data links which, depending on the system configuration, may reduce the maximum number of nodes.

DCS Message Hopping lets a DCS message route through an intermediate node without tandeming an associated trunk call. This is accomplished through the use of hop channels. The system provides Message Hopping through up to two hops. For a detailed description of DCS Message Hopping, see the *AT&T System 75—Application Notes—Distributed Communications System,* 555-209-003.

DCS transparency is more restricted when the tandem node is an Enhanced DIMENSION PBX or a System 85 Release 2 Version 1 than when it is a System 85 Release 2 Version 2, or later, or a DEFINITY Generic 2.1. (See the DCS Alphanumeric Display for Terminals and DCS Leave Word Calling feature descriptions in Chapter 3.)

Certain feature capabilities are unique to a particular type of node (for example, a System 75 or DEFINITY Generic 1 endpoint node). Therefore, a detailed feature description should be consulted for each type of node.

The Centralized Attendant Service (CAS) feature can be used as an advantage in DCS networks where all attendants are at one node. CAS reduces traffic volume on interconnecting tie trunks caused by incoming attendant-seeking calls at the endpoint nodes. A V1 or V2 System 75 cannot serve as the main location for CAS attendants. A V3 System 75 or a DEFINITY Generic 1 can serve as the main location for CAS attendants. Centralized Attendant Service capabilities are given in detail in Chapter 3.

A call from one DCS node to another DCS node can redirect through the Call Coverage feature. The Coverage tone, which indicates that the call has redirected to Coverage, is heard by the calling party at the distant node. However, the call cannot redirect to a distant node. The principal and the covering user must be located at the same node. An exception to this is when CAS is used. However, DCS transparency is not provided for the CAS call. Only the release link trunk name will be displayed at the attendant console, not the name or extension of the user on the remote switch that is covering to the attendant.

Main/Satellite/Tributary

Figure 2-6 shows a Main/Satellite/Tributary configuration. It can function independently or serve as an Electronic Tandem Network (ETN) 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. This is accomplished with the optional Uniform Dial Plan software.

Tributary and Satellite locations are similar except that a Tributary has one or more attendant positions and its own Listed Directory Number.

System 75 Version 1 can serve as a Satellite or Tributary. Version 2, Version 3, or DEFINITY Generic 1 can serve as a Main, Satellite, or Tributary.

A small business can start with a single Main/Satellite or Main/Tributary complex and add trunk and switching facilities as the business grows. In this situation, tie trunks connect the main locations within an urban area and intercity traffic is routed via the public network. This arrangement favors a medium-size organization or one that has small isolated locations where the intercity traffic is too small to justify the cost of tie trunks.

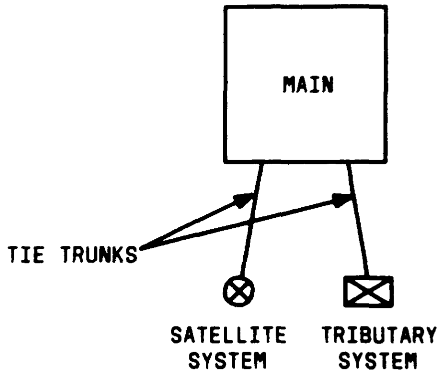


Figure 2-6. Main/Satellite/Tributary Configuration

Trunking

Trunking is the use of communications links to interconnect two switching systems, such as connecting the switch to a local central office or to another switch. These links, called trunks, can be grouped together in Trunk Groups when all the trunks in the group perform the same function. This grouping simplifies administration since the required service characteristics (parameters) are assigned to the group rather than to each trunk. Grouping also simplifies call processing. Calls requiring a trunk are routed to the appropriate trunk group and an idle trunk, if available, is selected from the group.

The following types of trunk groups can be used with the system:

- Auxiliary—Provides internal trunk applications for features such as Loudspeaker Paging and Music-on-Hold.

- Central Office (CO)—Provides a link with the local CO for calls except Direct Inward Dialing (DID) calls.

- Direct Inward Dialing (DID)—Provides a link with the local CO.

- DS1 Tie Trunk—Provides for two types of digital tie trunk interfaces: Voice-Grade DS1 and Alternate Voice/Data (AVD) DS1 tie trunks. The Voice-Grade DS1 tie trunks are an alternative to 4-wire analog E&M tie trunks and may be used to interface with other properly-equipped switching systems. AVD DS1 tie trunks permit alternate voice and data calling between a System 75 or DEFINITY Generic 1 and a System 85 or DEFINITY Generic 2. DS1 tie trunks can also be used with Release Link trunks for Centralized Attendant Service, and can be used with MEGACOM Service.

- Foreign Exchange (FX)—Provides a link with a CO other than the local CO.

- Integrated Services Digital Network—Primary Rate Interface (ISDN-PRI)—Provides end-to-end digital connectivity and supports a wide range of voice and non-voice services. Calls to a variety of switched nodal services such as MEGACOM, WATS, and ACCUNET and calls destined for different interexchange carriers can be processed.

- Tie and Release Link (V2, V3, and G1 only)—Provide a link with another private switching system for calls between the systems. Release link trunks are used only with Centralized Attendant Service. Tie trunks are used on calls to or from the following:
  - A Private Branch Exchange (PBX)
  - An Electronic Tandem Network (ETN) switch
  - An Enhanced Private Switched Communications Service (EPSCS) or Common Control Switching Arrangement (CCSA) office
  - MEGACOM Service.
- Wide Area Telecommunications Service (WATS)—Provides a link with an Outward WATS office or an 800 Service office.

Tie trunks used with the system are administered as either internal or external. The internal or external designation controls the type of ringing received at a voice terminal when an incoming tie trunk call arrives and controls the routing of the call if it is redirected through the Call Coverage feature:

- Incoming internal tie trunk calls cause 1-burst ringing and will redirect according to the redirection criteria administered for internal calls.
- Incoming external tie trunk calls cause 2-burst ringing and will redirect according to the redirection criteria administered for external calls.

The Call Coverage feature is described in detail in Chapter 3.

Selection of the trunk group to be used for a given call is determined by digit translation on the trunk access code. Assuming that an idle trunk in the selected group is found, a seizure signal (service request) is sent to the distant switch. If the distant switch requires the called number, a start dial signal is normally returned to the calling switch, indicating readiness to accept digit transmission.

The start dial signal(s) used is dictated by the serving FX office, WATS office, or local CO. For interconnection with other private switching systems, the System Manager may select the start dial signal(s) to be used.

"Trunk type" refers to the physical design of a trunk circuit. Trunk type and the start dial signal are often used interchangeably, although trunk type is a more accurate term. A brief description of the available trunk types follows:

- Ground Start—A ground signal is sent over the trunk ring lead and is received over the trunk tip lead.

- Loop Start—A closure signal is sent through the loop formed by the trunk leads.
- Immediate Start—No start dial signals are used. On outgoing calls, the system waits at least 80 milliseconds after sending the seizure signal before sending the digits required at the distant switch. This gives the distant switch enough time to attach a digit receiver to the call.
- Wink Start—A momentary signal (wink) is sent to the distant switch.
- Delay Dial—A steady signal is sent to the distant switch and is removed when the trunk is ready to receive digits.
- Automatic—No start dial signals are used. The seizure signal sent or received is sufficient to route the call. The call destination is specified when the trunk group is administered. The destination can be the attendant group or any extension number assigned in the system.

Trunk groups connecting with a WATS office, FX office, or local CO can be ground or loop start. DID trunk groups can be immediate or wink start. Tie trunk groups can be delay dial, wink start, immediate start, or automatic.

Trunk groups can be 1-way incoming, 1-way outgoing, or 2-way. Whether the trunk group is available for incoming, outgoing, or 2-way traffic is called direction. A 2-way loop-start trunk is subject to glare. Glare occurs when the distant switch is trying to use a given trunk for a call to System 75 or DEFINITY Generic 1 at the same time System 75 or DEFINITY Generic 1 is trying to use the same trunk for a call to the distant switch. Incoming calls are not aborted because of glare. The incoming call will complete, and the outgoing call will receive reorder tone. Queuing at both ends of a 2-way trunk group compounds the possibility of glare and, therefore, is not recommended.

Each non-DCS outgoing and 2-way trunk group can have a queue. If all trunks in the group are busy, the call waits in the queue until a trunk becomes idle. The queue length, which is the number of calls waiting, may be from 1 to 100. A queue length of 0 (zero) indicates no queue has been established. This information is entered on the trunk group form when the trunk group is administered.

Dual tone multifrequency (DTMF) signaling or rotary dial (dial pulse) signaling can be used between switches. (DTMF is also referred to as touch-tone signaling.) The system can send or receive either type of signaling required by the distant switch.

An incoming trunk call to the system can be connected to another trunk, a voice terminal, an attendant console, or an announcement. When the call is answered, "an answer supervision" signal is sent to the distant Enhanced Private Switched Communications Services (EPSCS), local CO, FX, WATS, or 800 Service office. This signal initiates the recording of the call details normally used for charging. Any CO call routed outward is deemed "answered" 10 seconds after the last digit is dialed or when answer supervision time-out expires, whichever comes first. Tie trunk calls are deemed "answered" when answer supervision is returned from the far end or when answer supervision time-out expires. Also, if there is a trunk incoming from one of the previously listed offices on a call of this type, then answer supervision is sent to that office. An incoming call to a Direct Department Calling or Uniform Call Distribution recorded delay announcement is deemed "answered" when the calling party is connected to the announcement. Other types of announcements, such as unassigned number announcements, are treated as an unanswered call.

System Management

Overview

System Management provides the capabilities to control and maintain the system and also provides system usage reports to help determine if the system is being used as intended. In short, System Management allows the System Manager to establish the system, monitor its use, and make additions and/or changes as necessary.

System Management features and functions are described in Chapter 3. Functions are more fully described in the following documents.

- *AT&T System 75—Implementation, Release 1 Version 1,* 555-200-650
- *AT&T System 75—Implementation, Release 1 Version 2,* 555-200-651
- *AT&T System 75—Implementation, Release 1 Version 3,* 555-200-652
- *DEFINITY Communications System Generic 1—Implementation,* 555-204-654
- *DEFINITY Communications System Generic 1 and System 75—Administration and Measurement Reports,* 555-200-500
- *AT&T System 75—Maintenance,* 555-200-105
- *DEFINITY Communications System Generic 1—Maintenance,* 555-204-105.

Changes made to system translations are effected only at the single system for which the changes were made. If a system is part of a network, changes may have to be made at more than one system to effect the desired changes to the network. Similarly, changes intended for only a single system could affect the network. Therefore, the System Manager must consider the effect on the network before making any changes.

System Management Features

The following features are associated with System Management:

Customer-Provided Equipment (CPE) Alarm (XEV2, XEV3, or G1)
Facility Test Calls
Move Agent From CMS (V3 or G1)
Recent Change History (G1)
Report Scheduler and System Printer (G1)
Station Message Detail Recording
System Measurements
System Status Report (V2, V3, or G1)

System Administration

Allows the user to implement (initialize) and administer all the terminal and system features and system parameters. System Administration allows the following:

- Initializing the system
- Managing system, voice terminal, and data terminal features on a day-to-day basis
- Performing system back-up procedures
- Monitoring, detecting, and determining system performance
- Maintaining system security.

System administration and maintenance are performed at the System Access Terminal (SAT) (V1, V2, or V3) or the Manager I terminal (G1), a Remote Administration terminal, or Customer Services Support Organization (CSSO). The SAT and Manager I terminal perform the same functions and are referred from here on as the administration terminal.

The administration terminal can be any of the following:

- 513 Business Communications Terminal (BCT)
- 515 BCT (functions as a 513)
- 4410 Terminal (does not provide print capabilities)
- 4425 Terminal
- 610 BCT (must be optioned as a 4410 or a 513 BCT)
- 615 MT BCT.

The administration terminal must be located within 50 feet of the system cabinet and must be connected directly to the Maintenance circuit pack. The administration terminal consists of a video display and keyboard which allow a System Manager to input system commands and translations. The administration terminal is first used to initialize the system. After initialization, the administration terminal is used to reconfigure translations and to monitor system performance.

The CSSO is a service available from an AT&T Service Center and has the same administrative capabilities as the administration terminal.

Remote Administration

Allows the system to be administered from a remote terminal located either on or off the customer's premises. A terminal located more than 50 feet from the system cabinet is considered remote. A remote administration terminal can be on the same premises as the local administration terminal or it can be off-premises. The remote administration terminal performs the same functions as the local administration terminal.

The 513 BCT, 515 BCT, 610 BCT, 615 MT BCT, 4410 terminal, or 4425 terminal may be utilized as either an on-premises or off-premises remote terminal. The 510D terminal can be used as an on-premises remote terminal if it is connected directly to the switch or as an off-premises remote terminal, if modem pooling is used.

If the remote terminal is a 4410 terminal, 513 BCT, or a 610 BCT, it must be connected to the system through a Processor Data Module (PDM), Digital Terminal Data Module (DTDM), or Data Line circuit pack. If one of the other terminals is used as a remote terminal, a PDM or DTDM is not required. The cabling distance from the system to the remote terminal is determined by the type of module associated with the terminal. Distance limitations are as follows:

- Remote terminal to PDM—5000 feet using 24-gauge wire or 4000 feet using 26-gauge wire
- Remote terminal to DTDM—3400 feet using 24-gauge wire or 2200 feet using 26-gauge wire.

For a description of the data modules and BCTs, refer to *DEFINITY Communications System and System 75 and System 85, Terminals and Adjuncts—Reference,* 555-015-201.

Only three users can be logged into the administration functions at one time. This includes a user of the administration terminal.

Customer Services Support Organization (CSSO)

Allows system administration and maintenance from a remote location.

The CSSO allows its user to access the system and perform administrative tasks assigned to the System Manager. The administrative commands used by the System Manager are also available to the CSSO users. The CSSO can also be used to perform maintenance routines.

During system access, the CSSO automatically receives major and minor alarm notifications from the system. When an alarm is received, CSSO users can access the system and perform the following tasks:

- Display alarms
- Display errors
- Clear errors
- Test and busyout circuit packs, voice terminals, and trunks
- Set time and date
- Receive backup translations for the system

- Download a copy of the system tape
- Perform any required administration.

Hospitality Services (V3 or G1)

Overview

The Hospitality Services features of the system will meet the lodging industry's need to provide services for their guests. The basic feature set is included in the basic voice application software and is sometimes referred to as the hotel/motel feature software package.

Hospitality Services Features

The following features are associated with Hospitality Services:

- Automatic Wakeup
- Do Not Disturb
- Names Registration (G1)
- Property Management System Interface
  - Check-In/Check-Out
  - Controlled Restriction
  - Housekeeping Status
  - Message Waiting Notification
  - Room Change/Room Swap
  - Guest Information Input/Change (G1)

Call Management (V3 or G1)

Overview

The Call Management features of the system support industries, such as airlines and travel agencies that have a large number of calls that are similar, and allow balanced call distribution to a group of voice terminals.

Call Management Features

The following features are associated with Call Management:

Abandoned Call Search
Agent Call Handling
Automatic Call Distribution (ACD)
Basic Call Management System (G1)
Intraflow and Interflow
Move Agent From CMS
Queue Status Indications
Service Observing

CHAPTER 3. FEATURE DESCRIPTIONS

Figures

Tables

CHAPTER 3. FEATURE DESCRIPTIONS

Overview

This chapter defines the system features associated with Voice Management, Data Management, Network Services, and System Management. The features are arranged in alphabetical order, regardless of the functional area to which they apply. The information for each feature is presented under five headings: Description, Considerations, Interactions, Administration, and Hardware and Software Requirements.

- Description

  Defines the feature, tells what it does for the user or how it serves the system, and briefly describes how it is used.

- Considerations

  Discusses the applications and benefits of the feature, followed by the feature parameters and any other factors to be considered when the feature is used.

- Interactions

  Lists and briefly discusses other features that may significantly affect the feature being described. Interacting features are those that:

  — Depend on each other—One of the features must be provided if the other one is.

  — Cannot coexist—One of the features cannot be provided if the other one is.

  — Affect each other—The normal operation of one feature modifies, or is modified by, the normal operation of the other feature.

  — Enhance each other—The features, in combination, provide improved service to the user.

- Administration

  States whether or not administration is required, how the feature is administered, who administers the feature, and lists items requiring administration.

- Hardware and Software Requirements

  Lists any additional hardware and/or software requirements needed for the feature.

AAR/ARS Partitioning (V3 or G1)

Description

Provides for the Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) services to be partitioned among as many as four (V3) or eight (G1) different groups of users within a single System 75 or DEFINITY Generic 1. This provides individual routing treatment for the different groups of users.

A partitioned user group consists of those users who are grouped together and share the same Partition Group Number (PGN). The PGN is not a restriction, but a means used to indicate the choice of routing tables to be used on a particular call. Each Class of Restriction (COR) is assigned a specific PGN. Different CORs may be assigned the same PGN. Therefore, it is possible for members of the same partitioned user group to have different CORs.

When the AAR/ARS Partitioning feature is used in a hotel/motel or a hospital environment, different facilities access is provided through ARS for guest/patient voice terminals and administrative staff member voice terminals. For example, within a hotel or motel, the guests and staff voice terminals might be partitioned into two user groups. When a guest places an inter-state call, the guest user group's ARS tables may specify that the call be routed using AT&T QUOTE Service, a telephone billing information system that is used to bill back or allocate long-distance charges. A similar call placed by a staff member might be routed over a Direct Distance Dialing (DDD) trunk.

All partitioned user groups share the same pool of Routing Patterns. (See the Automatic Alternate Routing and Automatic Route Selection features for further explanations on routing.) The 3-digit translation tables (HNPA, FNPA, or RNX) that specify the Routing Pattern number are unique for each partitioned user group. Routing Patterns may be shared among the user groups or may be dedicated to a particular user group. Once a user activates the ARS or AAR feature and dials enough digits for the system to search for the Routing Pattern, the PGN of the originator's COR is used to select the table to look up the Routing Pattern.

Users of AAR/ARS Partitioning include the following:

- Single-Line Voice Terminals
- Multi-Appearance Voice terminals
- Attendants
- Remote Access Users
- Data Endpoints
- Incoming Tie Trunks
- Other Trunks, such as those used when calls are forwarded to an off-premises number.

Considerations

With AAR/ARS Partitioning, different groups of users, within the same system, can receive individual routing treatment. For example, the following types of situations may require AAR/ARS Partitioning:

- Groups of users who have different routing preferences for calls to a given area due to special billing needs
- Groups of users who wish to have dedicated use of a particular network facility
- Groups of users in different businesses in one or more buildings serviced by a single system
- Data users who require special facility types on outgoing calls.

Partition user groups are only used with AAR and ARS. There is no capability to access the partitioned user groups directly. Operation of the groups is completely transparent.

Interactions

The following features interact with the AAR/ARS Partitioning feature.

- Bridged Call Appearance

  If a Bridged Call Appearance is used for an AAR or ARS call, the system will use the PGN of the bridged principal's extension instead of the PGN of the originating user's extension.

- Call Forwarding All Calls (V3 or G1)

  If a call terminates at a voice terminal that has Call Forwarding All Calls activated and the forwarded-to number uses AAR or ARS, the COR of the calling user is used to look up the PGN for the call.

- Distributed Communications System (DCS)

  The AAR/ARS Partitioning feature can cause different Routing Patterns to be used on DCS calls. For example, one user's Routing Pattern may specify a DCS trunk group as a member of the pattern. A user of a second PGN may use a different Routing Pattern which does not specify the DCS trunk group. In this case, one user has DCS feature transparency and the second user does not.

  When a call routes over a DCS trunk, no PGN information is sent to the far-end PBX. Thus, the far-end PBX will only be capable of using the incoming trunk's PGN to route the call.

- Remote Access

  If a Remote Access user activates ARS, the COR assigned to the barrier code dialed (or the Authorization Code, if required) will be used to select the PGN for the call.

- Station Message Detail Recording (SMDR)

  The PGN used to route the call is not recorded in SMDR.

- Straightforward Outward Completion and Through Dialing

  If the attendant assists or extends a call for a user and activates ARS, the attendant's COR is used to select the PGN for the call.

- Uniform Dial Plan

  Since Uniform Dial Plan (UDP) calls expand the dialed digits into 7-digit numbers and then use AAR to route the call, these calls will make use of partitioning. Once the call begins to be handled by AAR, the user's active COR will be used to identify the proper PGN to handle the call.

Administration

AAR/ARS Partitioning is administered by the System Manager. The following items require administration:

- Different FNPA, HNPA, and RNX tables must be administered for each partitioned user group.
- A PGN must be assigned to each COR table. Up to four (V3) or eight (G1) PGNs can be used.

Hardware and Software Requirements

No additional hardware or software is required.

Abandoned Call Search (V3 or G1)

Description

Provides identification of abandoned calls. Before an incoming CO trunk call to a hunt group or Automatic Call Distribution (ACD) split rings the hunt group member or agent, the system checks to make sure the calling party has not abandoned the call (hung up). If the calling party has abandoned the call, the call does not ring the hunt group member or agent.

To see if the calling party has abandoned the call, the system must determine if the calling party is still connected to the ground-start trunk at the central office (CO). To do this, the system flashes (opens the tip-ring loop for 150 to 200 ms) the CO end of the trunk. If the calling party is still connected, the CO will not respond. If the calling party has hung up on the call, the CO will send a disconnect signal within 700 to 800 ms. The system interprets this as an abandoned call, releases the trunk, and the call does not ring the hunt group member or agent.

After it is administered for a trunk group, this feature is performed automatically by the system. No operation is required by system users.

Considerations

Abandoned Call Search allows agents and hunt group members to answer more calls because time is not wasted on abandoned calls. In addition, call handling statistics generated by the Call Management System (CMS) are more accurate, because the CMS knows when a call is abandoned.

Abandoned Call Search only works with older COs that do not provide disconnect supervision (Abandoned Call Search works only with ground start analog trunks). These older COs can take as long as 2 minutes to notify the PBX of a disconnect and, thus, require the PBX to determine whether a call has been abandoned prior to extending the call. Most COs provide disconnect supervision and, therefore, do not require the Abandoned Call Search feature. Even with Abandoned Call Search or disconnect supervision, a small probability exists that a call will be extended to the destination hunt group after the caller has hung up. Abandoned Call Search and disconnect supervision serve to significantly reduce that probability.

Interactions

None.

Administration

Abandoned Call Search is administered on a per trunk group basis by the System Manager. Each ground start CO, FX, and WATS trunk group is administered as either having Abandoned Call Search or not having it.

Hardware and Software Requirements

Abandoned Call Search requires the use of a TN747B CO Trunk circuit pack, if the serving CO is a No. 1 or No. 5 Crossbar switch.

No additional software is required.

Abbreviated Dialing

Description

Provides lists of stored numbers that can be accessed to place local, long-distance, and international calls; to activate features; or to access remote computer equipment. Stored numbers can be accessed by voice terminal users and data terminal users. With V2, V3, or G1, certain stored numbers can also be accessed by attendants.

List Types

Desired called numbers are stored in any of three (V1 or V2) or four (V3 or G1) types of lists, and each stored number is one list entry. To use Abbreviated Dialing, a user merely accesses the appropriate list through a dial access code, and then dials the 1-, 2-, or 3-digit (V3 or G1) list entry number where the desired called number is stored. The number is then dialed automatically by the system. For a frequently called number, the list and list entry number can be stored on an abbreviated dialing button. In this case, simply pressing the button places the call.

The types of lists where desired called numbers are stored are as follows:

- Personal Number Lists

  Allow voice and data terminal users to have a personal set of stored numbers. With Version 1, a user can have one Personal Number List with 5 or 10 list entries. With Version 2, Version 3, or DEFINITY Generic 1 a user can have up to 3 Personal Number Lists with 5 or 10 entries per list. As many as 400 (V1), 800 (V2 or V3), or 1600 (G1) Personal Number Lists are allowed in the system. The user, or the System Manager, programs the Personal Number Lists. The System Manager sets which users will have a personal list and the size of each list (5 or 10 entries).

- Group Number Lists

  Allow access by a group of users, such as purchasing or personnel departments, who frequently dial the same numbers. As many as 100 Group Number Lists are allowed in the system. The Group Number Lists are administered by the System Manager. Each Group Number List can have up to 15 (V1) or 90 (V2, V3, or G1) list entries (in multiples of 5). An individual user can access up to 3 specific Group Number Lists, as set by the System Manager.

- System Number List

  Can have up to 50 (V1) or 90 (V2, V3, or G1) entries (in multiples of 5). The System Number List can contain any number or dial access code. The System Manager programs the System Number List and sets which users can access the list. One System Number List is allowed per system.

- Enhanced Number List (V3 or G1)

  Can have up to 1000 entries. One Enhanced Number List is allowed per system in addition to the System Number List. The Enhanced Number List can contain any number or dial access code. The System Manager programs the Enhanced Number List and sets which users can access the list.

List Entries

List entries for the Personal Number Lists are numbered 1 through 9 and 0. List entries for the Group Number Lists are numbered 11 through 25 (V1) or 11 through 99 and 00 (V2, V3, or G1). List entries for the System Number List are numbered 11 through 60 (V1) or 11 through 99 and 00 (V2, V3, or G1). List entries for the Enhanced Number List (V3 or G1) are numbered 000 through 999. This numbering scheme is used because the system expects either one, two, or three (V3 or G1) digits to identify entries on a given list, not a mixture.

List Assignments and Designations

Each extension number can be assigned up to three Abbreviated Dialing Lists—List 1, List 2, and List 3. Each of these three lists is designated as being either Personal, Group, System, or Enhanced (V3 or G1). With V1, the three lists may be any combination of the above, as long as there is no more than one Personal List and/or System List in the combination. With V2 or V3, the three lists may be any combination of the above as long as there is no more than one System and/or Enhanced (V3 or G1) List. When a list is designated as being a Group List, the particular number of the Group List is specified (for example, group list 42).

To access Abbreviated Dialing, the user accesses List 1, List 2, or List 3 either by dialing the access code or by using a button programmed with the access code. The access codes for List 1, List 2, and List 3 are the same systemwide. Therefore, it is possible for a System List or a particular Group List to have a different access code at different voice terminals. For example, suppose the feature access codes for List 1 and List 2 are 101 and 102, respectively. One voice terminal may have List 2 administered as "group 42." Another voice terminal may have List 1 administered as "group 42." In this case, the access code for "group 42" is 102 for the first voice terminal and 101 for the second voice terminal.

Privileged Lists

All Group Number Lists, the System Number List, and the Enhanced Number List can be designated as Privileged by the System Manager. Calls automatically dialed from a Privileged List are completed without Class of Restriction or Facilities Restriction Level (FRL) checking. [FRLs are associated with the Automatic Route Selection (V1) and Automatic Alternate Routing (V2, V3, or G1) features.] This allows access to selected numbers that certain voice terminal users might otherwise be restricted from manually dialing. For example, a voice terminal user may be restricted from making long-distance calls. However, the number of another office location may be long distance. This number could be entered in a list designated as Privileged. The user could then call the office location using Abbreviated Dialing, while still being restricted from making other long-distance calls.

Special Characters

A number stored in an Abbreviated Dialing List can be a combination of numerical digits and special characters. A special character instructs the system to take a different action when dialing reaches the point where the character is stored. Each special character counts as two digits toward the maximum number of digits in a list entry. The following special characters can be stored:

- Pause

  When a Pause precedes, or is included in, a string of stored digits to be outpulsed over a trunk, outpulsing of the digit(s) following the Pause will be delayed 1.5 seconds. Outpulsing will automatically resume after expiration of the delay timing.

  The Pause is useful when the probability of dial tone being returned within 1.5 seconds is high. Typical applications include tandem switching through private networks and end-to-end signaling over the public or a private network.

- Wait

  When a Wait precedes, or is included in, a string of stored digits to be outpulsed over a trunk, outpulsing of the digit(s) following the Wait will be delayed 4 seconds (V1), 5 to 25 seconds or until dial tone is detected (V2, V3, or G1), or until the user initiates an End-Wait signal, whichever occurs first. Outpulsing will resume after the End-Wait signal is received or when delay timing expires. In Version 2, Version 3, and DEFINITY Generic 1 systems that have 748B tone detectors, outpulsing will resume as soon as precise dial tone is received, if it is received before delay timing expires.

  The Wait is useful in cases where dial tone delays of variable length and/or network blocking outside the system are frequently experienced. Typical applications include tandem switching through private networks and end-to-end signaling over the public or a private network.

- Mark

  When a Mark precedes, or is included in, a string of stored digits, all digits following the Mark are treated as end-to-end signaling digits to be outpulsed over an outgoing trunk in touch-tone signal form even if a dial pulse trunk was used to set up the call. As a typical application, a data call can be made over a dial pulse trunk.

- Suppress

  When a Suppress precedes, or is included in, a string of stored digits, the system treats all digits following the Suppress the same as any other digits for call setup and digit outpulsing. The Suppress character only affects the display of the stored number. Stored numbers are normally shown when an alphanumeric display is provided through the Voice Terminal Display feature; however, the digits following the Suppress character are not displayed. The display shows the lowercase letter s instead of the stored digits.

The Pause and Wait special characters are not needed to delay outpulsing of the initial digits following access of an outgoing trunk; the system always knows when to start outpulsing over a trunk. Use of these characters as the very first character could cause calls to be aborted. These characters are used when outpulsing should be delayed until dial tone is returned from a distant point reached through a switched connection outside the system.

List Access Options

Stored numbers can be accessed by any of the following options:

- Abbreviated Dialing-Code (AD Code)

  This option allows users to access a stored number by dialing the AD feature access code and a list entry number. Each AD code automatically dials the number stored in the list the user accessed.

- Abbreviated Dialing-Button (AD Button)

  This option allows multi-appearance voice terminal users and attendants to access stored numbers by pressing one or more buttons. Each AD button automatically dials the number stored in the list and the list entry number administered to the button. Access to any list and associated list entry number can be programmed in an AD button on a multi-appearance voice terminal. An AD button on an attendant console can be programmed to access a Group List, the System List, or the Enhanced List (V3 or G1) and associated list entry number.

  The System Manager administers the AD button. If the button is administered to access a number in the user's Personal Number List, the user can change the number that is assigned to the button. However, if the number assigned to the button accesses an entry on a Group List, the System List, or the Enhanced List (V3 or G1), only the System Manager can make the change.

A separate list, called the 7103A Group Number List, is used only by 7103A Fixed Feature voice terminal users as a group. This list allows button access to stored numbers and can have eight list entries. Any number can be stored in the 7103A Group Number List; however, it is intended primarily for feature access codes. The System Manager programs the 7103A Group Number List.

All users can program their Personal Number List, and users with an assigned AD button can program the button. Programming is done by dial access or by pressing the Program button, if assigned.

Programming Personal Lists and AD Buttons

To program an entry in a Personal Number List, the user dials the AD Program access code or presses the AD Program button, then dials the list number (V2, V3, or G1), the list entry number, and the number to be stored [up to 16 digits (V1) or 24 digits (V2, V3, or G1)], and then presses the # button. Confirmation tone is heard when the number is stored. While in the program mode, users can program all Personal Number List entries, if desired. To exit the program mode, the user simply hangs up.

To program an AD button administered to access a particular entry in the Personal Number List, the user dials the AD Program access code or presses the AD Program button, if assigned. The user then presses the AD button, dials the desired number [up to 16 digits (V1) or 24 digits (V2, V3, or G1)], and then presses the # button. Confirmation tone is heard when the number is stored. While in the program mode, the user can program as many assigned AD buttons as desired. To exit the program mode, the user simply hangs up.

Only the System Manager and multi-appearance voice terminal users can program special characters. Voice terminal users need Pause, Mark, and Suppress buttons or a Function Entry button to program special characters. Pressing a Pause, Mark, or Suppress button programs the special character administered to the button. Pressing the AD Function Entry button and then dialing 1, 2, 3, or 4 programs Pause, Mark, or Suppress, respectively.

Considerations

Abbreviated Dialing provides easy access to selected numbers by decreasing the number of dialed digits required to place the call. Instead of dialing the entire number, the user merely dials a short code to access the desired number. The system then dials the stored number automatically. For frequently called numbers, an abbreviated dialing button can be assigned, allowing the call to be placed by merely pressing the button. By assigning a Privileged list of numbers, a user is allowed to place calls to selected numbers that might otherwise be restricted.

Users can be assigned access to three AD lists. With Version 1, the three lists can be made up of any combination of one Personal Number List, up to three Group Number Lists, and the System List. With Version 2, the three lists can be made up of any combination of up to three Personal Lists, up to three Group Lists, and the System List. With Version 3 or DEFINITY Generic 1, the three lists can be made up of any combination of up to three Personal Lists, up to three Group Lists, the System List, and the Enhanced List.

An Abbreviated Dialing Personal List cannot be administered to an attendant console.

A maximum of 502 (V1), 802 (V2 or V3), or 1600 (G1) lists and a maximum of 2500 (V1), 4010 (V2 or V3), or 8000 (G1) entries are allowed for the system. The 502 (V1), 802 (V2 or V3), or 1600 (G1) lists include a maximum of 400 (V1), 800 (V2 or V3), or 1600 (G1) Personal Number Lists, 100 Group Number Lists, a 7103A Group Number List, a System Number List, and an Enhanced Number List (V3 or G1). (See the System Capacities table in Section 4 for a summary of Abbreviated Dialing Parameters.)

A number stored in any list in the switch can contain up to 16 (V1) or 24 (V2, V3, or G1) digits. A special character used for Pause, Wait, Mark, or Suppress counts as two digits.

Interactions

The following features interact with the Abbreviated Dialing feature.

- Audio Information Exchange (AUDIX) Interface (V3 or G1)

  When using an Abbreviated Dialing button to access AUDIX, the user's login and password should not be assigned to the button. The system ignores button entries after the AUDIX number.

- Last Number Dialed

  This feature will place a call to the same number as called previously, even if Abbreviated Dialing was used on the previous call. However, if any special characters (Mark, Wait, Pause, and/or Suppress) are included in the previous call, they are not used on the Last Number Dialed call.

  With Version 2, Version 3, and DEFINITY Generic 1, if the previously called number was in an Abbreviated Dialing Privileged List, and if the user is not normally allowed to dial the number because of his or her Class of Restriction, Intercept Treatment is given when using Last Number Dialed. To redial the number, the user must again use the Abbreviated Dialing Privileged List.

- Bridged Call Appearance

  A user, accessing Abbreviated Dialing while on a bridged call appearance, accesses his or her own Abbreviated Dialing lists. The user does not access the Abbreviated Dialing lists of the primary extension associated with the bridged call appearance.

- Remote Access

  Remote Access users cannot access Abbreviated Dialing.

Administration

Abbreviated Dialing is administered by the System Manager. However, an Abbreviated Dialing Personal List can be programmed by either the System Manager or the voice terminal user.

A Personal Number List must be assigned to a voice terminal before the System Manager can establish that list. For example, during implementation, a voice terminal must first be assigned a Personal Number List on the individual voice terminal form. The actual list can then be established on the Abbreviated Dialing Personal List form.

The following items, if required for a given system, are set by the System Manager:

- Feature Access Codes for List 1, List 2, and List 3, and for programming a personal list
- Voice Terminal Assignments
  - AD buttons, if desired
  - AD Program button, if desired
  - Mark, Pause, Suppress, and Function Entry buttons, if desired
  - Access to as many as three lists
- Data Module Assignments (Access to an Abbreviated Dialing list)
- Abbreviated Dialing Lists
  - Personal Number Lists
  - Group Number Lists
  - System Number List
  - Enhanced Number List (V3 or G1)
  - 7103A Group Number List
- Wait Delay Interval (5 to 25 seconds)
- Attendant Console Parameters.

Hardware and Software Requirements

No additional hardware is required for Version 1 systems. With Version 2, Version 3, and DEFINITY Generic 1 systems, additional 748B tone detectors (up to five per system) may be required if the special "wait" character is used frequently.

No additional software is required for Version 1 and Version 2 systems. With Version 3 and DEFINITY Generic 1 systems, optional software is required for the enhanced Abbreviated Dialing list.

Agent Call Handling (V3 or G1)

Description

Provides Automatic Call Distribution (ACD) agents with the various capabilities required to answer and process ACD calls.

The agent capabilities provided by this feature are as follows:

- Agent Log-In and Log-Out
- Agent Answering Options
  - Automatic Answer
  - Manual Answer
- ACD Work Modes
  - Auxiliary Work Mode
  - After Call Work
  - Auto-In
  - Manual-In
- Agent Request for Supervisor Assistance
- ACD Call Disconnecting.

Agent Log-in and Log-out

To receive ACD calls, the agent must log into the system. An agent logging into a split automatically enters the Auxiliary Work mode (described later) for that split. An agent can be logged into as many as three splits at a given time. An agent may or may not be required to enter a log-in identification number when logging in.

To log in, an agent must go off-hook and dial the log-in feature access code, followed by the split group number and the log-in identification number (if required). If the log-in procedure is successful, the agent enters the Auxiliary Work mode and the lamp associated with that split's Auxiliary Work button, if provided, lights steadily on the agent's terminal. At the same time, the system sends two messages to the Call Management System (CMS): a message that the agent has logged in (including identification number if applicable) and a message that the agent has entered the Auxiliary Work mode.

If, during the log-in process, any of the following situations occur, the log-in attempt is canceled and the agent receives Intercept Treatment.

- The agent dials an invalid log-in feature access code.
- The agent dials an invalid split group number.
- The agent is already logged into three splits. In this case, Intercept Treatment is received after dialing the split group number.
- The agent dials a split group number for a split that he or she is already logged into.
- The agent dials the wrong number of digits.

When an agent leaves his or her position for an extended period of time and is therefore unavailable for ACD calls, the agent should log out. If an agent is administered to be measured by CMS and logs out, a message is sent to the CMS so that it no longer measures the agent's status. If an agent is logged into more than one split, he or she must log out of each individual split.

To log out of a split, the user has to go off-hook and dial the log-out feature access code followed by the split group number. If the log-out attempt is successful, the agent hears confirmation tone and all lamps associated with work mode buttons (described later) go dark. If the agent is logged into more that one split, logging out of one split does not affect the state of the other split.

If, during the log-out process, any of the following situations occur, the log-out attempt is canceled, and the agent receives Intercept Treatment.

- The agent dials an invalid log-out feature access code.
- The agent dials an invalid split group number.
- The agent dials a split group number for a split that he or she is not logged into.

If an agent is in the Automatic Answer mode (described later), he or she can log out simply by hanging up. If an agent in the Automatic Answer mode is using a headset instead of a handset, the agent can log out by turning off the headset. If this method is used to log out, the agent is automatically logged out of all splits that he or she has logged into.

If calls are in the split queue, the last available agent can still log out of the split by dialing the log-out feature access code or, if the agent has a multi-appearance voice terminal, by going on-hook.

Agent Answering Options

An agent can answer ACD calls by using either a headset, handset, or speakerphone. An agent can be assigned one of two answering options: Automatic Answer or Manual Answer.

Automatic Answer

If the agent is assigned Automatic Answer, he or she can be connected directly to incoming calls without ringing. Instead of the usual process where an agent receives ringing and then goes off-hook and answers the call, the agent hears zip tone through the headset, handset, or speakerphone and is automatically connected to the incoming ACD call.

It is recommended that Automatic Answer be used with a headset. In this case, the agent hears zip tone through the headset and is then automatically connected to the call. (If the incoming trunk group is data restricted, the zip tone is not heard. If the agent's extension is data restricted, the zip tone is not heard. A headset user should not be assigned data restriction.)

Although possible, it is not recommended that a handset or speakerphone be used with Automatic Answer. In order for an agent with Automatic Answer and a handset or speakerphone to answer an ACD call, the handset or speakerphone must be off-hook (handset lifted or speakerphone turned on). While off-hook, the agent hears zip tone through the handset or speakerphone.

Manual Answer

If the agent is assigned Manual Answer, the agent hears ringing, and then goes off-hook to answer the incoming call. The agent can use either a headset, handset, or speakerphone to answer the call.

ACD Work Modes

At any given time, an agent can be in one of four work modes. An agent can change work modes at any time. If an agent is not active on a call or has a call on hold, the mode change is immediate. However, if an agent is active on a call and tries to change modes, the mode is not changed until the agent is disconnected from all of the calls. An agent can change modes by using either button or dial access. The four work modes are described in the following paragraphs.

- Auxiliary Work
- After Call Work
- Auto-In
- Manual-In.

Auxiliary Work Mode: An agent should enter the Auxiliary Work mode for a particular split whenever he or she is doing non-ACD activities such as taking a break or going to lunch. This makes the agent unavailable for ACD calls to that split.

When an agent logs into a split, he or she automatically enters this mode for that split. To change to the Auxiliary Work mode while in another mode, the agent can dial the feature access code for the Auxiliary Work mode followed by the split group number or can press the Auxiliary Work button for that split. If the attempt to change modes is successful and the agent has no active or held calls, the lamp associated with the Auxiliary Work button lights steadily and the CMS is informed of the mode change for that agent. If the attempt to change modes is successful, but the agent has any active or held calls, the lamp flashes until all calls are dropped, at which point the CMS is informed. The attempt is canceled and the agent receives intercept treatment if the agent:

- Tries to enter the Auxiliary Work mode for an invalid split
- Tries to enter the Auxiliary Work mode for a split of which he or she is not a member
- Dials an invalid feature access code.

If an agent is the last agent logged into the split and calls are in queue for that split, the agent cannot enter the Auxiliary Work mode until the queued calls are handled. An attempt to enter the Auxiliary Work mode under these conditions prevents new calls from entering the queue for that split.

Once an agent has entered the Auxiliary Work mode for a particular split, the agent is no longer available to answer other ACD calls to that split. However, the agent is still available for ACD calls to other splits that the agent is logged into and is still available for non-ACD calls. The CMS is notified whenever an agent in the Auxiliary Work mode receives an incoming non-ACD call or makes an outgoing call.

After Call Work Mode: An agent should enter the After Call Work (ACW) mode when he or she needs to perform ACD-related activities. For example, an agent may need to fill out a form as a result of an ACD call. The agent can enter the ACW mode to fill out the form. The agent is unavailable for ACD calls to all splits while in the ACW mode.

When an agent is in the Manual-In mode (described later) and disconnects from an ACD call, he or she automatically enters this mode. To change to the ACW mode while in another mode, the agent can dial the feature access code for the ACW mode followed by the split group number, or press the ACW button for that split. If the attempt to change modes is successful, the lamp associated with the ACW button lights steadily and the CMS is informed of the mode change for that agent. If the attempt to change modes is successful, but the agent has any active or held calls, the lamp flashes until all calls are dropped, at which point the CMS is informed. If the agent tries to enter the ACW mode for an invalid split or for a split of which he or she is not a member, if the agent is already in the ACW mode for another split, or if the agent dials an invalid feature access code, the attempt is canceled and the agent receives intercept treatment.

Once an agent has entered the ACW mode for a particular split, the agent is no longer available to answer ACD calls to that or any other split. However, the agent is still available for non-ACD calls. The CMS is notified whenever an agent in the ACW mode receives an incoming non-ACD call or makes an outgoing call.

Auto-In Mode: When an agent enters the Auto-In mode, he or she, upon disconnecting from an ACD call, automatically becomes available for answering new ACD calls.

To change to the Auto-In mode while in another mode, the agent can dial the feature access code for the Auto-In mode followed by the split group number or can press the Auto-In button for that split. If the attempt to change modes is successful, the lamp associated with the Auto-In button lights steadily and the CMS is informed of the mode change for that agent. If the attempt to change modes is successful, but the agent has any active or held calls, the lamp flashes until all calls are dropped, at which point the CMS is informed. If the agent tries to enter the Auto-In mode for an invalid split or for a split of which he or she is not a member, or if the agent dials an invalid feature access code, the attempt is canceled and the agent receives intercept treatment.

Manual-In Mode: When an agent enters the Manual-In mode, he or she, upon disconnecting from an ACD call, automatically enters the After Call Work mode for that split, and is not available for any ACD calls. The agent must then manually re-enter either the Auto-In mode or Manual-In mode to become available for ACD calls.

To change to the Manual-In mode while in another mode, the agent can dial the feature access code for the Manual-In mode followed by the split group number or can press the Manual-In button for that split. If the attempt to change modes is successful, the lamp associated with the Manual-In button lights steadily and the CMS is informed of the mode change for that agent. If the attempt to change modes is successful, but the agent has any active or held calls, the lamp flashes until all calls are dropped, at which point the CMS is informed. If the agent tries to enter the Manual-In mode for an invalid split or for a split of which he or she is not a member, or if the agent dials an invalid feature access code, the attempt is canceled and the agent receives intercept treatment.

Agent Request for Supervisor Assistance

Agents can request assistance (whether on an active ACD call or not) from the split supervisor by pressing the Assist button or dialing the Assist feature access code, followed by the split group number. If a split supervisor is not assigned, the agent receives intercept tone.

To request supervisor assistance using the Assist button, the agent does as follows:

- If the agent is active on an ACD call, the agent presses the Assist button for that split. This automatically places the ACD call on hold and places a call to the split supervisor. The CMS is notified of the request and the supervisor's display (if provided) shows that the call is a request for assistance. After the agent has talked to the supervisor, the agent can drop the assist call and return to the ACD call, or the agent can set up a conference call with the agent, the supervisor, and the calling party. The agent can also transfer the call to the split supervisor, if desired.

  If the agent is an attendant, he or she should first press the Start button before pressing the Assist button. This will allow the attendant to later transfer the call.

- If the agent is not active on a call, the agent goes off-hook and presses the Assist button. This automatically places a call to the split supervisor. The CMS is notified of the request and the supervisor's display (if provided) shows that the call is a request for assistance.

To request supervisor assistance using the Assist feature access code, the agent does as follows:

- If the agent is active on an ACD call, the agent places the ACD call on hold, receives dial tone, and then dials the Assist feature access code followed by the split group number. The Assist call is then placed to the split supervisor. The CMS is notified of the request for assistance, and the supervisor's display (if provided) shows that the call is a request for assistance. After the agent has talked to the supervisor, the agent can drop the Assist call and return to the ACD call, or the agent can set up a conference call with the agent, the supervisor, and the calling party. The agent can also transfer the call, if desired.

- If the agent is not active on an ACD call, the agent goes off-hook and then dials the Assist feature access code followed by the split group number. The Assist call is then placed to the split supervisor. The CMS is notified of the request for assistance, and the supervisor's display (if provided) shows that the call is a request for assistance.

ACD Call Disconnecting

An agent can be disconnected from an ACD call in either of three ways. The agent can press a Release or Drop button (if provided), the call can be dropped by the calling party, or the agent can go on-hook (hang up). The agent hears dial tone after pressing the Drop button. Dial tone is not heard after the Release button is pressed.

Agents using Automatic Answer are logged out of all splits when they disconnect from an ACD call by going on hook (hanging up).

Considerations

The Agent Call Handling feature is really a combination of features and functions that allow ACD split agents to handle ACD calls quickly and efficiently.

An agent, although he or she can be assigned to one or more splits, can only be logged into three splits at a time.

An agent can only be connected to one ACD call at a time. However, an agent active on an ACD call can receive non-ACD calls.

The number of digits in the log-in identification number must equal the number assigned through system administration (0 to 9). The agent's individual identification number is used for record keeping purposes only. The system checks only the number of digits in the identification number. It does not check to see if the identification number is a valid number although the CMS utilizes the log-in identification for reports.

For each split to which an agent is assigned, he or she can be assigned a maximum of one of each of the following feature function buttons:

- Manual-In
- Auto-In

- Auxiliary Work
- After Call Work.

A terminal or console can be assigned a maximum of one ACD Release button. This button is in addition to the fixed Release button on the attendant console.

The last available agent in a split cannot enter the Auxiliary Work mode if any calls are remaining in the split queue. An attempt by the last available group member to enter the Auxiliary Work mode results in the following:

- The Auxiliary Work button flashes.
- New calls to the ACD split either receive busy tone or redirect to coverage.
- Calls already in the split queue continue to route to the last available agent until the split queue is empty.
- At the last available voice terminal or console, the status lamp associated with the Auxiliary Work button, if provided, flashes until the split queue is empty. When no more calls remain in the split queue, the Auxiliary Work mode is entered and the associated status lamp, if provided, lights steadily. (The same sequence applies when the Auxiliary Work mode is dial activated instead of button activated, except there is no status lamp.)

If an agent is logged into more than one split, the agent may become unavailable for calls to one split, because of activity at another split. For example, an agent may answer a call or enter the After Call Work mode for one split. This makes the agent unavailable for calls to other splits the agent is logged into.

An ACD agent on conference with more than three parties may cause inaccurate CMS measurements.

An agent should not log into a split while a call is on hold at his or her extension.

Interactions

The following features interact with the Agent Call Handling feature.

- Abbreviated Dialing

  An agent may have Abbreviated Dialing buttons assigned to make the log-in process easier. An Abbreviated Dialing button can be programmed to dial the access code, split number, and/or identification number.

- Automatic Call Distribution

  The CMS may or may not be administered. The Agent Call Handling features function the same whether it is administered or not. If the CMS is not administered, the system does not send information to the CMS as described in the Agent Call Handling feature description.

- Hold

  If an agent places an ACD call on hold, no information is reported to the CMS. Therefore, the CMS considers the agent still active on the call and the agent is unavailable to receive other ACD calls.

Administration

Agent Call Handling is administered by the System Manager. The following items require administration on a per-terminal or per-console basis:

- Whether it has Automatic Answer or Manual Answer
- Whether or not it has Idle Appearance Preference
- Manual-In button
- Auto-In button
- Auxiliary Work button
- After Call Work button
- Assist button
- Release button.

In addition to the above, the following items require administration on a per-system basis:

- Feature access codes:
  - Agent Log-In
  - Agent Log-Out
  - Manual-In
  - Auto-In
  - After Call Work
  - Auxiliary Work
  - Assist
- Number of digits in log-in identification.

Hardware and Software Requirements

No additional hardware is required. ACD software is required.

AP Demand Print (V1, V2, or V3)

Description

Allows the voice terminal user to print his or her own undelivered messages without calling the AP-based Message Center.

The Message lamp at each voice terminal indicates whether or not any undelivered messages are waiting for the voice terminal user. The lamp is lighted when there are undelivered messages and goes dark when there are no undelivered messages. When the Message lamp is lighted at a voice terminal, the voice terminal user can have the message(s) printed.

The voice terminal user whose messages are to be printed is called the requesting extension. (Messages are requested by entering the extension number of the person for whom the messages were left.) The requesting extension can activate the AP Demand Print feature or another extension can activate the AP Demand Print feature for the requesting extension. The extension that activates the AP Demand Print feature is called the originating extension. Thus, if a user activates the feature from his or her own voice terminal to print his or her own messages, then the assigned extension number is both the requesting and originating extension.

Each requesting extension has an assigned printer which is used to print the AP Demand Print messages. However, some extensions may be associated with an overriding printer. If the originating extension is associated with an overriding printer, the messages will be printed on that printer instead of the printer assigned to the requesting extension.

A requesting extension can be an individual voice terminal, a Personal CO Line Group, a Uniform Call Distribution Group, a Direct Department Calling Group, or a Terminating Extension Group. Each requesting extension is assigned an authorization password. The password consists of four digits. Each digit can be 0 through 9. This password allows the originating extension to access the requesting extension's messages.

AP Demand Print is activated either by dialing the feature access code or by pressing the Print Msgs button. After the originating extension does this, the requesting extension's number and the requesting extension's authorization password must be entered. The messages are then printed. If an overriding printer is used by an originating extension, the system attempts to print the messages on that printer. Otherwise, the system attempts to use the printer assigned to the requesting user. In either case, if the printer is inoperable, the messages are routed to the AP-system default printer. After the messages are printed, the message waiting lamp at the requesting extension goes dark.

To illustrate how the AP Demand Print feature functions, assume the following:

- The Message lamp lights at extension 321.
- Extension 432 is to activate the AP Demand Print feature to print the message(s) for extension 321.

- Extension 432 is associated with an overriding printer.

In the given situation, extension 321 is the requesting extension and extension 432 is the originating extension. Since extension 432 is associated with an overriding printer, the message(s) for extension 321 will be printed on that printer. If there is no overriding printer, the messages will be printed on the printer assigned to extension 321. Under the given conditions, the following sequence of events will take place:

1. To print extension 321's message(s), extension 432 activates the AP Demand Print feature either by dial access or button.

2. Extension 432 then enters the extension number and authorization password for extension 321.

3. The messages are then printed at the overriding printer associated with extension 432, and the Message lamp at extension 321 goes dark.

Considerations

AP Demand Print reduces the Message Center work load by allowing voice terminal users to print their own undelivered messages. The Message Center benefits from this feature by gaining more time for other operations. The voice terminal users also benefit from this feature, because they have control of retrieving their own messages. Up to ten overriding printers can be assigned.

Interactions

In addition to Message Center messages, all undelivered Leave Word Calling messages can be printed with AP Demand Print.

When an AP is provided, Leave Word Calling is provided by the AP. Operation of this feature, when AP-based, differs from the operation described elsewhere in this section. AP Demand Print will notify a user if he or she has AUDIX messages.

Administration

AP Demand Print is administered by the System Manager. The following items require administration:

- Requesting Extensions
  - Authorization passwords
  - Printers (Administered on AP)

- Originating Extensions
  - Print Msgs buttons, if desired
- Overriding Printers
  - Up to 10 overriding printers can be assigned on the AP
- Feature Access Code
  - Access code for activating AP Demand Print.

Hardware and Software Requirements

This feature requires an AP Interface. No additional software is required.

Attendant Auto-Manual Splitting

Description

Allows the attendant to announce a call or consult privately with the called party without being heard by the other party on the call.

This feature is activated automatically when the attendant, active on a call, presses the Start button, a Group Select button and a Direct Extension Selection button (if provided), or a Trunk Group Select button. Any of these actions temporarily separates the party on the call from the connection and allows the attendant to call and talk privately with another party.

The connection is reestablished when the attendant presses one of the following buttons:

- Cancel—Cancels the call attempt and reconnects the attendant and the separated party.
- Split—Establishes a 3-way conversation with the attendant, the separated party, and the called party.
- Release—Connects the separated party and the called party and disconnects the attendant.

Considerations

Attendant Auto-Manual Splitting automatically provides a splitting of the called party. Splitting allows the attendant to privately determine if the called party can and will accept the call.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Call Waiting

Description

Allows an attendant originated or extended call to a busy single-line voice terminal to wait at the called terminal. The attendant is free to handle other calls.

Attendant Call Waiting is activated automatically (V2) or when administered for a single-line station (V3 or G1) whenever an attendant originates or extends a call to a busy single-line voice terminal. The attendant hears a Call Waiting ringback tone and the busy voice terminal user hears a 2-burst tone. The 2-burst tone is heard only by the called voice terminal user.

When the Attendant Call Waiting is activated the attendant may choose to cancel the call, release the call, or hold the call on the console. However, releasing an attendant-originated call results in the call being dropped completely. The call waits until the voice terminal is idle or until the administered interval (Return Call Timeout or Timed Reminder on Hold) expires. If the interval expires, the call returns to the console. With V1 systems, the call in progress at the voice terminal cannot be placed on hold. It must be terminated. With V2, V3, or G1 systems, the call in progress at the voice terminal can be placed on hold. In order to answer the waiting call, after receiving recall dial tone, the user then dials the answer call waiting access code. After answering the waiting call, the voice terminal user can use the Hold feature to return to the held call or toggle back and forth between the two calls.

As an example of how Attendant Call Waiting is used, assume extension 123, a single-line voice terminal, is busy. An attendant extends a call to extension 123 and hears the Call Waiting Ringback Tone which indicates that Attendant Call Waiting is activated. The attendant may choose to announce the call waiting condition to the calling party. However, after doing this, the attendant cannot cancel the call. The attendant could cancel the call and ask the calling party to call again later, or the attendant could release the call or place the call on hold at the console. This allows the attendant to handle other calls. The voice terminal user at extension 123 hears a 2-burst tone and knows a call is waiting. The voice terminal user at extension 123 can then terminate the call in progress (V1), or place the call in progress on hold (V2, V3, or G1), and answer the waiting call. If the waiting call is not answered before a preassigned time interval (Return Call Timeout or Timed Reminder on Hold) expires, the call returns to the attendant.

Considerations

Attendant Call Waiting allows an attendant to originate or extend calls to a busy single-line voice terminal, while allowing the attendant to handle other calls. Since the attendant is able to handle other calls while a call is waiting, more calls can be answered.

Attendant Call Waiting applies only for calls to single-line voice terminals within the system. Only one call per voice terminal can wait at a time.

Interactions

The following features interact with the Attendant Call Waiting feature.

- Automatic Callback

  If Automatic Callback is activated at the called voice terminal, Attendant Call Waiting is denied.

- Call Coverage

  Attendant Call Waiting calls may redirect to coverage if the called voice terminal has Data Privacy or Data Restriction activated. If one of these conditions exist, if Call Coverage is assigned to a voice terminal, and if Send All Calls is activated or coverage criteria are met, the call will not wait and can redirect to the coverage path. In some cases, the call can wait and then redirect to coverage. In other cases the call returns to the console, rather than redirecting to coverage. Operation is as follows:

  — The Coverage Don't Answer interval (2 to 9 ringing cycles or the equivalent time) specifies how long a call remains directed to the called voice terminal before redirecting to coverage. This interval applies to both the Busy and Don't Answer criteria. If Attendant Call Waiting is applicable on the call, this feature is active for the duration of the Don't Answer interval only. At the expiration of this interval, the call redirects to coverage.

  — If the Timed Reminder interval (10 to 1020 seconds) expires before the Don't Answer interval expires, the call does not go to coverage but returns to an attendant console. If the Don't Answer interval expires first, the call redirects to coverage, but can still return to the attendant console if a coverage point does not answer the call before the Return Call Timeout or Time Reminder on Hold interval expires.

  — If Send All Calls is active or if the redirection criterion is Cover All Calls, the call immediately redirects to coverage instead of waiting.

  — An attendant can release from an extended call at any point during the call, without affecting the preceding operations.

- Data Privacy

  If Data Privacy is activated at the called voice terminal, Attendant Call Waiting is denied.

- Data Restriction

  If Data Restriction is activated at the called voice terminal, Attendant Call Waiting is denied.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  Calls to a DDC or UCD group do not wait; however, such calls can enter the group queue, if provided.

- Loudspeaker Paging Access

  If Loudspeaker Paging Access is activated at the called voice terminal, Attendant Call Waiting is denied.

- Music-on-Hold Access

  Music-on-Hold can be heard by the calling party, if the call is a trunk transferred call and this type of call is administered to receive Music-on-Hold for call waiting calls. Otherwise, the calling party does not hear Music-on-Hold, but hears ringing.

- Recorded Telephone Dictation Access

  If Recorded Telephone Dictation Access is activated at the called voice terminal, Attendant Call Waiting is denied.

- Timed Reminder

  The Timed Reminder interval (10 to 1020 seconds) determines how long a call will wait before returning to an attendant console. If the call is not answered or does not redirect to coverage before this interval expires, the call returns to the attendant console.

Administration

Attendant Call Waiting is a standard system feature. With V2, no administration is required for the feature itself. With V3 or G1, Attendant Call Waiting is assigned to single-line voice terminals on a per-terminal basis. With all versions, the waiting interval is administered through the Timed Reminder feature by the System Manager. Also, transferred trunk calls that are waiting to be answered can be administered to receive Music-on-Hold.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Control of Trunk Group Access

Description

Allows the attendant to control trunk groups, and prevents voice terminal users from directly accessing a controlled trunk group.

Each attendant console has 12 designated Trunk Group Select buttons to be used with the Attendant Direct Trunk Group Selection feature. With G1, each console may have up to 12 of its feature buttons administered as additional Trunk Group Select buttons, for a total of 24 Trunk Group Select buttons per console. The attendant gains direct access to an outgoing trunk group by merely pressing the button assigned to that trunk group.

All Trunk Group Select buttons (including any administered on the console's feature buttons) have a Busy lamp that lights when all trunks in the associated trunk group are busy. If one of the two-lamp feature buttons on an enhanced console is administered as a Trunk Group Select button, the bottom lamp is used as the Busy lamp (the top lamp is not used). Six of the designated buttons (basic console) or all 12 designated buttons (enhanced console) have two additional lamps that are used for Attendant Control of Trunk Group Access. The two additional lamps are as follows:

- Warn (warning) lamp

  Lights when a preset number of trunks are busy in the associated trunk group (the busy threshold of the trunk group is reached.

- Cont (control) lamp

  Lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group.

The attendant activates Attendant Control of Trunk Group Access by pressing a Cont Act (Control Activate) button followed by the desired Trunk Group Select button. (The Trunk Group Select button used must have a Cont [control] lamp.) If a user attempts to access a controlled trunk group directly, the call automatically redirects to the attendant. If the attendant decides to allow the call to go through, the attendant can connect the user to the desired trunk group by pressing the associated Trunk Group Select button. The attendant can then release the call or hold the call on the console.

Calls that are already in queue for a trunk are not affected by the activation of Attendant Control of Trunk Group Access for that trunk group. For example, if an attendant activates Attendant Control of Trunk Group Access for a specific trunk group while a user is waiting in queue for an outside trunk in that trunk group, the call is not affected. The call will remain in queue until an idle trunk becomes available, at which time the call is connected to that idle trunk.

The attendant deactivates Attendant Control of Trunk Group Access by pressing the Cont Deact (Control Deactivate) button followed by the desired Trunk Group Select button. (The Trunk Group Select button used must have a Cont [control] lamp.) Attendant Control of Trunk Group Access is activated and deactivated separately for each trunk group.

After an attendant presses a Cont Act or Cont Deact button, the attendant can perform other operations before pressing the desired Trunk Group Select button. This has no effect on the activation or deactivation of the feature. For example, if the attendant presses the Cont Act button and then has to answer another call, the desired Trunk Group Select button can be pressed after answering the call. Attendant Control of Trunk Group Access is then activated for the associated trunk group.

Considerations

By activating Attendant Control of Trunk Group Access, the attendant obtains control of access to specific trunk groups. This allows the attendant to monitor the use of these trunk groups. By watching the lamps associated with the trunk groups, the attendant can determine if the number of busy trunks in a specific trunk group has reached a preset warning level and if all trunks in a specific trunk group are busy. The attendant can then handle other calls to these trunk groups accordingly.

This feature can be activated for any trunk group assigned to a Trunk Group Select button with an associated control lamp. Each attendant in the system can control access to 6 (basic console) or 12 (enhanced console) different trunk groups. With V1, V2, and V3 systems, the enhanced console is treated the same as the basic console. Therefore, only the first six designated Trunk Group Select buttons can be used for Attendant Control of Trunk Group Access in these systems.

If Attendant Control of Trunk Group Access is activated, and no attendant is assigned, or the attendant is later removed, calls to a controlled trunk group route to the attendant queue.

Interactions

The following features interact with the Attendant Control of Trunk Group Access feature.

- Attendant Direct Trunk Group Selection

  This feature must be assigned with Attendant Control of Trunk Group Access.

- Attendant Display

  When a call redirects to the console because Attendant Control of Trunk Group Access is activated, the alphanumeric display identifies the calling party and shows that the call has attempted to access a controlled trunk group.

- Automatic Route Selection and Automatic Alternate Routing

  Activating Attendant Control of Trunk Group Access removes the controlled trunk group(s) from the Automatic Route Selection and Automatic Alternate Routing patterns. Deactivating the feature reinserts the group(s) into the patterns. Automatic Route Selection calls are not routed to the attendant.

- Trunk Group Busy/Warning Indicators to Attendant

  This feature keeps the attendant informed of trunk group status. This status can be used to determine when to activate control.

- Uniform Dial Plan

  Activating Attendant Control of Trunk Group Access removes the controlled trunk group(s) from Uniform Dial Plan preferences. Deactivating the feature enables the Uniform Dial Plan to access the trunk groups.

Administration

Attendant Control of Trunk Group Access is assigned on a per-attendant console basis by the System Manager. The following items require administration:

- Attendant Console
  - Trunk groups which are to be controlled
  - Cont Act and Cont Deact buttons
- Controlled Trunk Groups
  - Busy Threshold.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Direct Extension Selection With Busy Lamp Field

Description

Allows the attendant to place or extend calls to as many as 800 (V1, V2, or V3) or 2000 (G1) extension numbers assigned to the system by pressing a Group Select button and a Direct Extension Selection (DXS) button instead of dialing the extension number. These extension numbers may be voice terminal extensions, hunt group extensions, off-switch extensions (such as Uniform Dial Plan extensions), or other non-voice terminal extensions.

Eight Group Select buttons and 100 DXS buttons are located on the basic selector console. The enhanced selector console has 20 Group Select buttons and 100 DXS buttons. G1 provides up to 12 additional Group Select buttons which can be assigned to feature buttons on the attendant console. However, if these feature buttons are used, the total number of Group Select buttons per attendant (including both the attendant console feature buttons and the selector console buttons) cannot exceed 20. Each Group Select buttons is labeled with a different hundreds group number used in the system. For example, if a system uses 4-digit extension numbers, the Group Select buttons could be labeled 2400, 2500, 2800, etc. Likewise, a 3-digit system could have these buttons labeled as 100, 200, 300, etc. A 2-digit system would have a 0 Group Select number. A 5-digit V2, V3, or G1 system, for example, could have group select buttons labeled 28400, 28500, 28600, etc.

The 100 DXS buttons are labeled 00 to 99, and each button represents the last two digits of an extension number. Each DXS button, when combined with a Group Select button, represents a unique extension number. To place a call to an extension number, the attendant merely presses the appropriate Group Select button followed by the appropriate DXS button. For example, to call extension 4321, the attendant would press Group Select button 4300 followed by DXS button 21.

A lamp associated with each Group Select button indicates the selected hundreds group. A selected hundreds group remains selected until another Group Select button is pressed. The associated lamp lights and remains lighted until another Group Select button is pressed. Each DXS button also has an adjacent lamp, which is used to determine the idle/busy active status of the facility associated with the button. When a facility is busy/active, the lamp at the associated DXS button is lighted. When the associated facility is idle the lamp is dark. The 100 lamps adjacent to the DXS buttons are referred to as a busy lamp field. Although the Group Select and DXS buttons may be used to dial any extension, the busy lamp field only reflects the status of on-switch resources.

After the Group Select button is pressed, if the lamp adjacent to the desired DXS button is lighted to indicate busy status, the call can sometimes still be placed or extended. Attendant Call Waiting will be activated for a single-line voice terminal. A multi-appearance voice terminal user receives the call on an idle appearance. If no idle appearances are available, the call can route to coverage, if available, or receive busy tone.

Considerations

With the Attendant Direct Extension Selection With Busy Lamp Field feature, the attendant can place calls to as many as 800 (V1, V2, or V3) or 2000 (G1) system users without having to dial the extension number. The attendant simply presses a Group Select button and a DXS button. If the desired Group Select button is already pressed, the attendant needs only to press the desired DXS button. This feature also provides the attendant with a visual indication of the idle/active status of the extension numbers assigned to the selected hundreds group.

A maximum of 100 extension numbers can be monitored for idle/active status at any one time, using the selector console busy lamp field.

Interactions

The following features interact with the Attendant Direct Extension Selection With Busy Lamp Field feature.

- Attendant Display

  When the attendant uses the Direct Extension Selection With Busy Lamp Field, the call is identified on the alphanumeric display through the Attendant Display feature.

- Call Coverage

  If Send All Calls is activated, or if the Call Coverage redirection criteria are met, then an extended call will redirect to the coverage path.

- Centralized Attendant Service (CAS)

  When a DXS button is used to make a CAS call, it takes a few seconds before the attendant hears ringback tone.

Administration

The only administration required is the hundreds group assignment for each of the up to eight Group Select buttons. Assignments are made by the System Manager.

Hardware and Software Requirements

Requires a selector console. No additional software is required.

Attendant Direct Trunk Group Selection

Description

Allows the attendant direct access to an idle outgoing trunk by pressing the button assigned to the desired trunk group.

Each attendant console has 12 designated Trunk Group Select buttons to be used with the Attendant Direct Trunk Group Selection feature. With G1, each console may have up to 12 of its feature buttons administered as additional Trunk Group Select buttons, for a total of 24 Trunk Group Select buttons per console. Each button allows the attendant direct access to an outgoing trunk group by simply pressing the button assigned to that trunk group.

All Trunk Group Select buttons (including any administered on the enhanced console's feature buttons) have a Busy lamp that lights when all trunks in the associated trunk group are busy. If one of the two-lamp feature buttons on an enhanced console is administered as a Trunk Group Select button, the bottom lamp is used as the Busy lamp (the top lamp is not used). Six of the designated buttons (basic console) or all 12 designated buttons (enhanced console) also have a Cont (control) lamp and a Warn (warning) lamp. The Warn lamp lights when a preset number of trunks in the associated trunk group are busy. The Cont lamp lights when the attendant has activated the Attendant Control of Trunk Group Access feature for the associated trunk group.

Instead of trunk groups, Loudspeaker Paging zones can be assigned to Trunk Group Select buttons. In this case, the Busy lamp indicates the idle/busy status of the associated Loudspeaker Paging zone.

Considerations

Attendant Direct Trunk Group Selection eliminates the need for the attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. A label associated with each Trunk Group Select button identifies its destination or use, for example, Chicago, Foreign Exchange (FX), or Wide Area Telecommunications Service (WATS). Pressing the button selects an idle trunk in the desired group.

Each attendant console has 12 designated Trunk Group Select buttons. With G1, each console may have up to 12 of its feature buttons administered as additional Trunk Group Select buttons, for a total of 24 Trunk Group Select buttons per console.

With V1, V2, and V3 systems, the enhanced console is treated the same as the basic console. Therefore, only the first six designated Trunk Group Select buttons will use the warning and control lamps.

Interactions

If the Attendant Control of Trunk Group Access feature is provided, this feature must also be provided.

Administration

Attendant Direct Trunk Group Selection is assigned on a per-attendant basis by the System Manager. Administration consists of assigning trunk groups or Loudspeaker Paging zones to the Trunk Group Select button.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Display

Description

Shows call-related information that helps the attendant to operate the console more efficiently. Also shows personal-service and message information. Information is shown on the alphanumeric display on the attendant console.

The following display modes can be assigned to the eight buttons in the display area of the console, or any of the programmable feature buttons on the console. The Normal and Test modes are always provided; the others are optional.

- Normal Mode

  Displays call-related information for the active call appearance. The alphanumeric display is in the Normal mode unless the attendant selects one of the other modes. The display must be in the normal mode to answer incoming calls.

- Inspect Mode

  Displays call-related information for a call on hold or an unanswered call.

- Stored Number Mode

  Displays the number assigned to a button administered through the Facility Busy Indication feature or the number assigned to an Abbreviated Dialing button.

- Date/Time Mode

  Displays the current date and time of day.

- Test Mode

  Displays a test pattern representing each of the 40 characters that can be displayed. The Lamp Test switch is provided on the console; an additional button assignment is not needed.

- Elapsed Time

  Displays elapsed time in hours, minutes, and seconds. The timing starts or stops when the button is pressed.

- Integrated Directory

  Turns off the touch-tone signals and allows the touch-tone buttons to be used to key in the name of a system user. After a name is keyed in, the display shows that name and associated extension number. (Refer to the Integrated Directory feature.)

- Coverage Message Retrieval Mode

  Retrieves and displays Leave Word Calling and Call Coverage messages for system users. Messages can be retrieved at any time. The attendant can be active on a call and still retrieve messages.

Three additional buttons should be assigned to the console when the Coverage Message Retrieval mode or the Integrated Directory mode is assigned. These buttons and their functions are as follows:

- Next Message

  Retrieves and displays the next message, displays NO MESSAGES, or displays END OF MESSAGES, (PUSH Next TO REPEAT) when in the Coverage Message Retrieval Mode. Displays the next name in the alphabetical listing when in the Integrated Directory mode. This button should be assigned when the Retrieval mode button is assigned.

- Delete

  Deletes the currently displayed message. This button must be assigned when the Retrieval mode button is assigned.

- Return Call

  Automatically returns the call requested by the currently displayed message or the currently displayed name and extension number. This button is optional.

The system provides the following call-related information:

- Call Appearance Identification

  The attendant call appearance buttons are labeled alphabetically beginning with the letter "a". The display shows, for example, a= for a call incoming on the first call appearance button, b= for a call incoming on the second call appearance button, and so on.

- Calling Party Identification

  — Version 1

    When the call is from a system user, the display shows the caller's extension number, the caller's name, or a unique identification administered for the voice terminal being used. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, assigned to the trunk group used for the call.

  — Version 2, Version 3, and DEFINITY Generic 1

    When the call is from a system user, the display shows the caller's name or a unique identification administered for the voice terminal being used, along with the calling party's extension number. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, and the trunk access code assigned to the trunk group used for the call.

With the Integrated Services Digital Network—Primary Rate Interface (ISDN-PRI) feature (G1), additional calling party information is provided. See the Integrated Services Digital Network—Primary Rate Interface feature description elsewhere in this chapter for details.

- Called Party Identification

  — Version 1

  On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name. If no name is assigned, the called party's extension number is displayed.

  On outgoing calls, the display shows the digits as they are dialed or the name assigned to the trunk group being used. The System Manager can suppress the name of any trunk group.

  — Version 2, Version 3, and DEFINITY Generic 1

  On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name and extension number. If no name is assigned, only the called party's extension number is displayed.

  On outgoing calls, the display shows the digits as they are dialed or the name and trunk access code assigned to the trunk group being used. The System Manager can suppress the name of any trunk group.

  With the ISDN-PRI feature (G1), additional called party information is provided. See the ISDN-PRI feature description elsewhere in this chapter for details.

- Internal Caller's Class of Restriction (COR)

  All system users have a COR to define their calling privileges. The COR is a 2-digit number followed immediately by a hyphen and a 4-character identifier. With V1, the display shows a user's COR whenever the attendant makes or answers an internal call. With V2, V3, or G1, a COR button must be pressed to display a user's COR. The COR information can be obtained from the System Manager. The restriction identifiers are as follows:

  ORIG—Origination restriction

  OTWD—Outward restriction

  TOLL—Toll restriction

CODE—Code restriction

NONE—No restriction

- Call Purpose

  This refers to calls that are directed, redirected, or returning to the console. The call purpose identifiers are as follows:

  co—Controlled Outward Restriction Call (V3 or G1)—Indicates that a call from an internal user has been redirected to the attendant because the user has Controlled Outward Restriction and has attempted to make an outgoing call.

  ct—Controlled Termination Restriction Call (V3 or G1)—Indicates that a call has been redirected to the attendant because a user has Controlled Termination Restriction and the calling party has tried to call that user.

  cs—Controlled Station-to-Station Restriction Call (V3 or G1)—Indicates that a call from an internal user has been redirected to the attendant because the user has Controlled Station-to-Station Restriction and has tried to make a station-to-station call.

  ic—Intercept Call—Indicates that the incoming call has been redirected to the attendant as a result of Intercept Treatment.

  he—Held Call—Indicates that the preset time limit has expired for a call on hold at the console. This identifier appears only on R1V1 systems.

  ld—DID LDN Call—Indicates that the incoming call is a Listed Directory Number (LDN) call on a Direct Inward Dialing (DID) trunk.

  rt—Returned Call—Indicates that an attendant-extended call was not answered within the administered interval and the call has returned to the console.

  rc—Recall Call—Indicates that an internal user, active on a call held on the console, is requesting attendant assistance.

  tc—Trunk Control—Indicates that an internal user attempted to access an attendant-controlled trunk and the call was redirected to the console.

  f—Call Forwarding—Indicates that an internal user has calls forwarded automatically to the attendant.

  When the Call Coverage feature is active and the attendant is a covering user, the following call purpose identifiers are displayed:

  s—Send All Calls—indicates that the called voice terminal user is temporarily sending all calls to coverage.

d—Don’t Answer or Cover—Indicates that the called voice terminal was not answered or that the calling system user has sent the call to coverage, or the called voice terminal user is not available. This identifier also indicates that the called voice terminal user has a temporary bridged appearance of the call.

b—Busy—Indicates that the called voice terminal user is active on a call, and the called voice terminal user has a temporary bridged appearance of the call.

B—Busy—Indicates that the called voice terminal user is active on a call, and the called voice terminal user does not have a temporary bridged appearance of the call. All calls to single-line voice terminals that go to coverage will display "B".

The attendant console has a 1-line 40-character alphanumeric display. Some typical displays are as follows:

Internal call originated by the attendant (V1):

```
a=3602
```

then

```
a=      TOM BROWN        04-NONE
```

or

```
a=      EXT 3602      04-OTWD
```

Internal call originated by the attendant (V2, V3, or G1):

a=3602

then

TOM BROWN	3062

or

a=	EXT 3602	3602

Outgoing trunk call originated by the attendant (V1):

b=87843541

Where 8 is the trunk access code and 784-3541 is the number dialed.

then

b=	OUTSIDE CALL

or

b=	WATS

Outgoing trunk call originated by the attendant (V2, V3, or G1):

```
b=87843541
```

Where 8 is the trunk access code and 784-3541 is the number dialed.

then

```
b=    OUTSIDE  CALL              8
```

or

```
b=          WATS            101
```

Where 101 is the trunk access code of the outgoing trunk group.

Incoming trunk call to the attendant (V1):

```
a =    OUTSIDE  CALL
```

Incoming trunk call to the attendant (V2, V3, or G1):

```
a =    OUTSIDE  CALL              102
```

Where 102 is the trunk access code of the incoming trunk group.

Conference call originated by the attendant:

b= CONFERENCE 4

Where 4 is the number of conferees. The number does not include the attendant.

Internal call redirected to coverage:

b= EXT 3174 to EXT 3077 d

or

b= BOB SMITH to JOYCE THOMAS d

Where d indicates that Go to Cover was activated by the calling voice terminal user.

Incoming trunk call redirected to coverage:

b= OUTSIDE CALL to DON SMITH s

Where s indicates that Send All Calls was activated by the called voice terminal user.

Coverage Message Retrieval

IN PROGRESS

then

MESSAGES FOR BETTY R. SIMS

then

JOE JONES 10/16 11:40a 2 CALL 3124

This message means that Joe Jones called Betty R. Sims the morning of October 16. The second message was stored at 11:40 a.m. Joe wants Betty to call his extension number, 3124.

Integrated Directory mode:

CARTER, ANN	3408	3

This display shows the name and extension number as administered in the system. The 3 indicates that three buttons were pressed to reach this particular display.

Considerations

The Attendant Display feature gives the attendant considerable call handling capabilities by displaying call-related information. With this feature, the attendant receives detailed information on incoming and outgoing calls. The display provides such information as the called number on call originations, identification of trunk groups and internal users on a call, and calling party restrictions of internal callers requesting assistance.

Attendant Display also provides the attendant with information associated with certain features such as Leave Word Calling and Integrated Directory. For these features, the display provides such information as names, extension numbers, and messages.

If the attendant group is administered for systemwide message retrieval, attendants can retrieve messages for voice terminal users. Permission to have coverage message retrieval must also be administered for the voice terminal user. It is not possible for selected attendants to retrieve messages for selected voice terminal users.

Interactions

With the Bridged Call Appearance feature, a call from the primary extension number or a bridged call appearance of the primary extension number is displayed as a call from the primary extension number.

If prefixed extensions are used in the system's dial plan, the prefix is not displayed when the extension is displayed. With V3, the Return Call button cannot be used to dial prefixed extensions, because this button causes the system to dial the displayed number, which does not contain the entire extension. With G1, the Return Call button can be used to dial prefixed extensions, because the G1 system will dial the prefix, even though it is not displayed.

Administration

The Attendant Display feature is administered on a per-attendant basis by the System Manager. Administration consists of assigning feature related buttons to each attendant console. The following buttons can be assigned:

- Coverage Message Retrieval
- Date and Time (one button)
- Delete Message (must be assigned if the Coverage Message Retrieval button is assigned)
- Elapsed Time
- Inspect Mode
- Integrated Directory

- Next Message (must be assigned if the Coverage Message Retrieval button is assigned)
- Normal Mode
- Return Call (optional, used with the Retrieval mode or the Integrated Directory mode)
- Stored Number
- COR.

The display must be in the Normal mode for the attendant to answer incoming calls.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Recall

Description

Allows voice terminal users on a 2-party call, or on an Attendant Conference call held on the console, to recall the attendant for assistance.

Single-line users press the Recall button or flash the switchhook to recall the attendant.

Multi-appearance users press the Conference or Transfer button to recall the attendant and will remain on the connection when either button is used.

Considerations

Attendant Recall provides a convenient means for a voice terminal user, on a call held on the console, to recall the attendant if further assistance is required.

The call must be held on the console.

Interactions

The following features interact with the Attendant Recall feature.

- Individual Attendant Access

  If a hunt group call to an individual attendant is being held on the console, a system user, active on the call, cannot recall the attendant. However, he or she can transfer calls or make conference calls.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Attendant Release Loop Operation

Description

Allows the attendant to hold the connection of any call off the console if completion of the call is delayed (such as a call extended to a busy single-line voice terminal or to a voice terminal that does not answer). This feature frees the attendant to handle other calls.

When an incoming call arrives on a call appearance at an attendant console and is answered, extended, and released by the attendant, the call is released from that call appearance. The console is then available to receive the next call.

Timed Reminder (Return Call Time-out) starts once the call is off the console. If the called terminal or coverage point user does not answer before the administered interval expires, the call returns to the attendant queue. Once the call comes out of queue and terminates at a console, the special recall tone is applied and the alphanumeric display shows the call identification.

Considerations

Attendant Release Loop Operation improves efficiency in handling calls by allowing the attendant to release from a call without having to wait for an answer. The attendant is immediately available to handle other calls.

Interactions

The following features interact with the Attendant Release Loop Operation feature.

- Timed Reminder

  Timed reminder tone is provided by this feature.

  The Return Call Time-out interval is provided by this feature.

- Attendant Display

  Call identification is provided by this feature.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Audio Information Exchange (AUDIX) Interface (V3 or G1)

Description

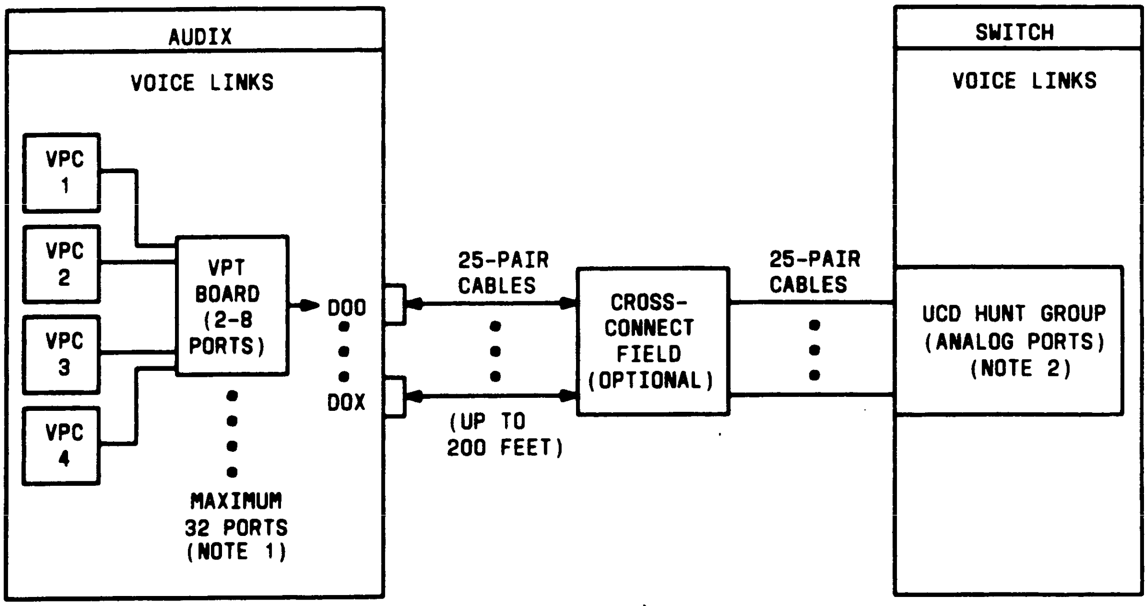
AUDIX is a message-handling system for recording and distributing spoken messages or "voice mail." It contains stored voice prompts that guide users to create, send, retrieve, answer, save, and forward spoken messages.

The following activities are available for use by AUDIX subscribers:

- Create Message—Record or modify a new message, address it, schedule it for delivery, and save a copy (optional).
- Scan Incoming Mailbox—Review new messages and reply or redirect them with an added comment, and review or delete old saved messages.
- Personal Greeting Administration—Record or modify a personal greeting to be played for callers who reach AUDIX through the Call Answer feature; select either the personal greeting or standard AUDIX greeting.
- Scan Outgoing Mailbox—Review, modify, or redirect messages scheduled for delivery; check the status of delivered messages; and review, modify, redirect, or delete messages saved in the file cabinet.
- Password and List Administration—Change user's personal AUDIX password and create, modify, review, or delete mailing lists.

The interface between the system and AUDIX consists of up to 32 analog (voice) connections, for exchange of voice messages, and a data link for status and control information exchange. AUDIX is available in both one-cabinet and two-cabinet configurations. The one-cabinet configuration provides up to 16 ports. The two-cabinet configuration provides up to 32 ports.

The analog ports on the system can be provided by TN742 circuit packs. Figure 3-1 shows typical voice connections between the system and AUDIX.



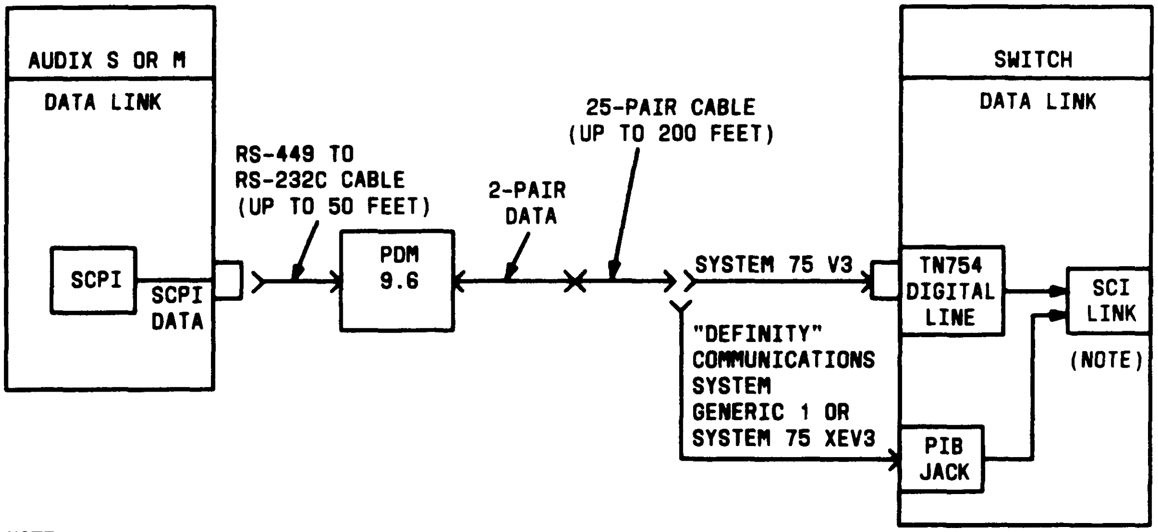
NOTES:

1. AUDIX-S AND AUDIX-M PROVIDE UP TO 16 VOICE PORTS AND IN MOST CASES WILL BE ADEQUATE. HOWEVER, IF THE NEED ARISES (FOR EXAMPLE, FOR DCS, ETC.), AUDIX-L MAY BE USED TO PROVIDE UP TO 32 PORTS.
2. ANALOG PORT CIRCUIT PACKS MAY BE: TN742 (8 PORTS), TN746 (16 PORTS), OR TN769 (8 PORTS).

Figure 3-1. Voice Connections—DEFINITY Generic 1 to AUDIX

The control link (data link) connection between AUDIX and the switch is via a TN754 or TN784 (G1) Digital Line circuit pack or Processor Interface (PI) Board (XEV3 and G1) to the Switch Communication Interface (SCI). The SCI is provided by the TN765 Processor Interface circuit pack or the TN716 Interface 1, TN738 Interface 2, and TN719 Interface 3 circuit packs.

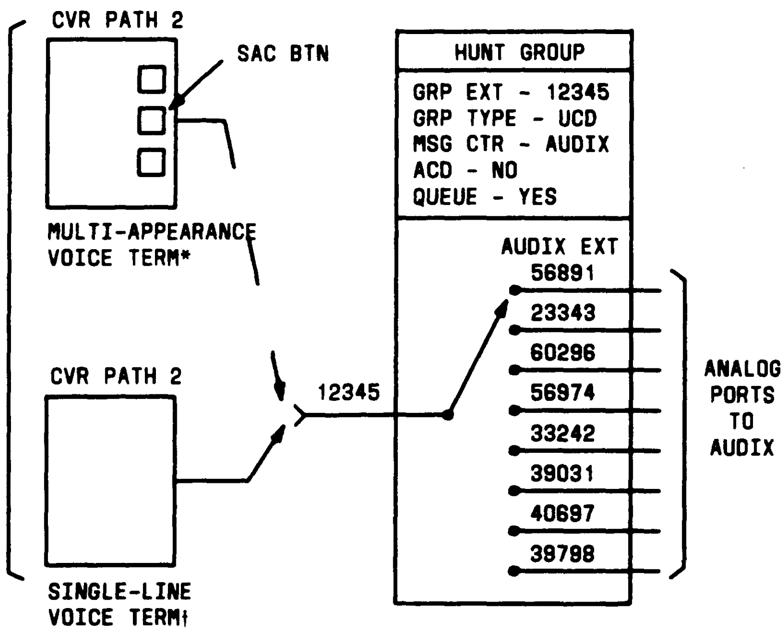
The control link connection to System 75 XEV3 or DEFINITY Generic 1 is directly to the PI jack (connected to the TN765 Processor Interface circuit pack). If the PI jack is already in use for another adjunct, a TN754 or TN784 (G1) Digital Line circuit pack is required. The SCI for System 75 XEV3 or DEFINITY Generic 1 is provided by the TN765 Processor Interface circuit pack. Figure 3-2 shows a typical control link connection between the switch and AUDIX.



NOTE:
THE SWITCH COMMUNICATION INTERFACE (SCI) LINK FOR SYSTEM 75 V3 IS PROVIDED BY TN716 INTERFACE 1, TN738 INTERFACE 2, AND TN719 INTERFACE 3 CIRCUIT PACKS OR BY THE TN765 PROCESSOR INTERFACE CIRCUIT PACK. THE SCI LINK FOR "DEFINITY" COMMUNICATIONS SYSTEM GENERIC 1 AND SYSTEM 75 XEV3 IS PROVIDED BY THE TN765 PROCESSOR INTERFACE CIRCUIT PACK. DIGITAL LINE CIRCUIT PACK NOT REQUIRED WHEN CONNECTING TO P1B JACK.

Figure 3-2. Data Link Connection—AUDIX

System analog ports connected to AUDIX must be assigned to a Uniform Call Distribution (UCD) hunt group uniquely identified as an AUDIX hunt group, so that AUDIX may be accessed directly via the hunt group extension number; and so that the hunt group may also be assigned as a point in a coverage path. (There is no restriction in the system as to where in a coverage path that AUDIX can be placed; it can be first, last, second, etc.). Figure 3-3 shows a simplified AUDIX arrangement.



* TO HAVE INCOMING CALLS ANSWERED BY AUDIX, COVERAGE REDIRECTION CRITERIA IS MET OR MULTI-APPEARANCE USER PUSHES SEND ALL CALLS BUTTON; TO ACCESS AUDIX, USER DIALS 12345.

† TO HAVE INCOMING CALLS ANSWERED BY AUDIX, COVERAGE REDIRECTION CRITERIA IS MET OR SINGLE-LINE USER DIALS SEND ALL CALLS ACCESS CODE; TO ACCESS AUDIX, USER DIALS 12345.

Figure 3-3. Simplified AUDIX Arrangement

System 75 V3 or DEFINITY Generic 1 allows AUDIX as an Automatic Call Distribution (ACD) split. An AUDIX hunt group may be administered as an ACD split by setting the ACD and Measured by MIS fields on the hunt group to "y". This allows AUDIX traffic to be measured via the ACD's Call Management System (CMS). AUDIX messages enabling voice ports are recorded as logins and AUDIX requests to the switch to disable voice ports are recorded as logouts by the CMS.

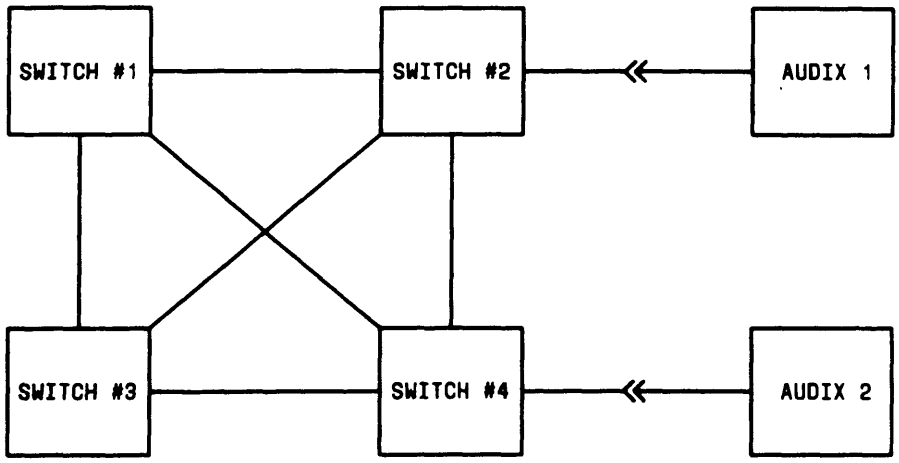
This is a feature for use by systems currently equipped with an ACD and CMS. For those systems not equipped with an ACD and CMS, AUDIX traffic measurements can be obtained using the System Measurements feature functionality and AUDIX traffic measurements.

System 75 V3 or DEFINITY Generic 1 provides Call Transfer Into AUDIX. This allows a user who is an AUDIX subscriber or a covering user for a principal who is an AUDIX subscriber to transfer a call to AUDIX so that the caller can leave a voice message in the principal's mail box. This may be accomplished by pressing the Transfer button, dialing the Transfer to AUDIX Feature Access Code (FAC), and then pressing the Transfer button again, or by depressing an abbreviated dialing button programmed with the access code and then pressing the Transfer button.

System 75 V3 or DEFINITY Generic 1 provides call progress feedback to the calling party for the Call Transfer Out of AUDIX feature. This feedback is in the form of call ringing and voice messages if the called party is busy. Also, a called party with a display-equipped terminal is informed of the call type (direct or redirected) and receives associated information about the call. Call Transfer Out of AUDIX is a joint system—AUDIX feature. It must be administered on the AUDIX machine, and the AUDIX machine must be V2 or later to support call progress feedback.

Only one AUDIX may be directly connected to a switch; however, System 75 V3 or DEFINITY Generic 1 allows the use of AUDIX in a Distributed Communications System (DCS). Each switch can have its own AUDIX which serves only the users connected to that switch; or the single AUDIX connected to the system may serve other switches in a DCS network. These other switches may be System 75 V3, DEFINITY Generic 1, or other switches that support DCS AUDIX (for example, System 85 R2V3).

A DCS network is not restricted to only one AUDIX. That is, one AUDIX connected to a system can serve all switches in a DCS network or the DCS network can be split for AUDIX coverage. One AUDIX connected to a switch can serve one part of the network while another AUDIX connected to another switch serves another part of the network; and each part of the network served by a particular AUDIX will have no knowledge of the existence of the other AUDIX(s). Figure 3-4 shows a simplified DCS AUDIX arrangement.



NOTE:
AUDIX 1 CAN SERVE ALL SWITCHES; AUDIX 2 CAN SERVE ALL SWITCHES; AUDIX 1 CAN SERVE SWITCHES 1 AND 2 WHILE AUDIX 2 SERVES SWITCHES 3 AND 4; AUDIX 1 CAN SERVE SWITCHES 2 AND 3 WHILE AUDIX 2 SERVES SWITCHES 1 AND 4; OR ANY LOGICAL COMBINATION CAN BE USED.

Figure 3-4. Simplified DCS AUDIX Arrangement

In a DCS configuration, the switch with the direct physical connection to AUDIX is referred to as the host switch; the other switches in the DCS are referred to as remote switches. In a remote switch, calls directed to AUDIX are routed to a remote AUDIX hunt group on that switch (the hunt group on the remote switch must be administered as a remote AUDIX hunt group), then subsequently routed to the AUDIX hunt group on the host switch.

The remote AUDIX hunt group is a dummy hunt group which has no analog port connections. It provides status and control information to the AUDIX hunt group on the host switch; AUDIX voice connections between the remote switch and the host switch are made via DCS tie trunks just as with any other DCS call.

Status and control information exchange between the remote switches and AUDIX is via hop channels. A hop channel allows a remote switch to exchange control and status information with AUDIX without a direct physical connection. The hop channel splices together the logical portions of the different data links to form one extended data link. The data is passed over the extended data link as if the two endpoints were directly connected.

With a DCS arrangement, AUDIX may be a coverage point in a call coverage path at a remote switch not directly connected to AUDIX. On the remote switch, a remote AUDIX hunt group in a call coverage path allows calls to be covered to an AUDIX hunt group on the host switch. The covered call then completes to AUDIX unless all the ports in the AUDIX hunt group are in use and the hunt group queue is full; in which case busy tone will be returned to the caller. If AUDIX is not accessible (data link down, all ports out of service, etc.), reorder tone will be returned to **any** user attempting to access AUDIX (for example, caller whose call was forwarded, AUDIX subscriber attempting to retrieve a message, etc.).

AUDIX Feature Use Description

General

The feature use description is divided into two parts: *Stand-Alone Switch* (AUDIX connected to a single switch) and *DCS AUDIX* (AUDIX in a DCS arrangement).

Direct Access to AUDIX

Stand-Alone Switch

An AUDIX subscriber may access his or her voice mailbox by dialing the extension number of the AUDIX hunt group. If the AUDIX hunt group queue is full, the caller hears busy tone. If AUDIX is inaccessible for some other reason, the caller hears reorder tone. Otherwise, the subscriber hears ringback (or optionally an announcement) until AUDIX answers the call. When AUDIX answers, the subscriber can log into voice mailbox.

DCS AUDIX

An AUDIX subscriber on any switch in the network may access his or her voice mailbox by dialing the extension number of the AUDIX hunt group on the host switch. If the AUDIX hunt group queue is full, the caller hears busy tone. If AUDIX is inaccessible for some other reason, the caller hears reorder tone. Otherwise, the subscriber hears ringback until AUDIX answers the call. When AUDIX answers, the subscriber may log into voice mailbox.

Subscribers on remote switches may dial the remote AUDIX hunt group extension local to their switch; this is useful for avoiding long-distance charges when accessing AUDIX from home. If the AUDIX hunt group queue is full, the caller hears busy tone. If AUDIX is inaccessible for some reason, the caller hears reorder tone. Otherwise, the subscriber hears ringback until AUDIX answers the call. When AUDIX answers, the subscriber may log into voice mailbox.

Two conditions prevent the call from routing to the host AUDIX switch. If all trunks to the host switch are busy, then the caller hears busy tone. If either the AUDIX data link to the remote switch is down or the DCS data link to the host switch is down, then the caller hears reorder tone.

Redirected Calls To AUDIX

Stand-Alone Switch

If an AUDIX hunt group is in a subscriber’s coverage path, calls redirected to coverage may terminate at AUDIX. If AUDIX is inaccessible, then normal call coverage treatment for an unavailable coverage point is given. Otherwise, the caller hears ringback; eventually AUDIX will answer and place the caller in the subscriber’s Call Answering Service.

Calls may also be forwarded to an AUDIX hunt group.

DCS AUDIX

If the called subscriber is on the host switch, the feature usage is the same as for the stand-alone switch.

On a remote switch, if the remote AUDIX hunt group is in the subscriber's Coverage Path, call coverage treatment is modified as follows: as long as the call can be forwarded with DCS transparency from the remote AUDIX hunt group to the host switch, then the call is terminated to the remote AUDIX hunt group coverage point. The caller hears ringback which may be followed by one of the following: if the AUDIX hunt group queue is full, the caller hears busy tone; if AUDIX is inaccessible for some reason, the caller hears reorder tone. The ringback source changes from the local switch to the remote/host switch in the middle of the call. This may create a noticeable change in ringback.

Two conditions prevent forwarding the call from the remote AUDIX hunt group. If all trunks to the host switch are busy or the DCS data link to the host switch is down, then the remote AUDIX hunt group is treated as a busy coverage point. If there is a coverage point in the coverage path beyond the AUDIX hunt group, then the call will terminate there.

Leave Word Calling (LWC) Store on AUDIX

Stand-Alone Switch

A calling party can leave a message for the called party by either dialing the LWC feature access code or pushing the LWC feature button. A covering user (via coverage, call pickup, call forwarding, etc.) may use the Coverage-Callback feature button to leave a message for the principal to call the calling party. In an established two-party call, either party can leave a message for the principal by simply pushing the LWC feature button.

LWC messages are stored on AUDIX if the principal's station is administered for AUDIX LWC. If the message storage is successful, the activating party hears a confirmation tone, and the receiving party's Message Waiting lamp is lighted. If the data link between the AUDIX and the switch is down, the attempt to leave a message is unsuccessful, and the activating party hears reorder tone.

This feature is similar to LWC on the switch or Messaging Service Adjunct, except the message is stored on AUDIX.

DCS AUDIX

If the calling party and called party are on the same switch, the feature works the same as for the stand-alone switch; when the parties are on different switches, DCS LWC is used.

If the LWC message is successfully delivered, the activating party hears confirmation tone and the receiving party's Message Waiting lamp is lighted.

Call Transfer Into AUDIX

Stand-Alone Switch

To invoke this feature, the user must have answered a call that was originally directed to a principal who has AUDIX as a coverage point; also, the user may be the principal. The operation follows the same procedure as a normal transfer except for the following: When the user hears secondary dial tone as a prompt, the user enters the Transfer to AUDIX Feature Access Code (FAC). There is no corresponding Transfer to AUDIX button in the system, but the FAC may be programmed into an Abbreviated Dialing button.

If the principal's Coverage Path does not contain AUDIX, then intercept tone is returned. If the principal's coverage path does contain AUDIX, the call is redirected to AUDIX Call Answering, just as if it had been redirected according to original redirection criteria. Forwarded calls that are transferred to AUDIX appear to AUDIX as forwarded calls. Direct calls transferred to AUDIX appear to AUDIX as calls redirected via Send All Calls, even if the principal's coverage path does not have the Send All Calls coverage criteria assigned. The switch treats the calls transferred to AUDIX as redirected calls and allows no further redirection if AUDIX is unavailable.

When the user that invoked the feature presses transfer again, the user is dropped from the call as with normal transfer.

This feature may be invoked by attendants and all station types.

DCS AUDIX

If the transferring party and principal are on the host switch, then the operation is the same as for the stand-alone switch. If the transferring party is on a remote switch, the operation is the same, but failure conditions are different.

There are two cases to consider: principal and transferring party on the same remote switch, or principal and transferring party on different switches.

If principal and transferring party are on the same remote switch, the principal's coverage path must contain the remote AUDIX hunt group. The transfer initially directs the call to the remote AUDIX hunt group on the remote switch. This in turn forwards the call to the AUDIX hunt group on the host switch. The same DCS AUDIX failure conditions that affect direct and redirected calls affect calls transferred via this feature.

If principal and transferring party are on different switches, the call is transferred without verifying that the principal has AUDIX as a coverage point. If the principal is not an AUDIX subscriber, then, when the call terminates to AUDIX, the calling party hears ringback indefinitely, because AUDIX will only answer calls for subscribers (this is true regardless of the location of the principal and the calling party).

Call Conference Into AUDIX

Stand-Alone Switch

This feature is similar to Call Transfer into AUDIX, but is invoked via the Conference feature. The user that invoked the feature remains with the call. Failure conditions are the same as for Call Transfer into AUDIX.

This feature may be invoked by attendants and other stations.

DCS AUDIX

Use of this feature for DCS AUDIX is the same as for Call Transfer into AUDIX for DCS AUDIX.

Call Transfer Out of AUDIX

Stand-Alone Switch

Before invoking this feature, the user must have established a call with AUDIX either by direct access or call redirection. To invoke the feature, the user enters touch-tone digits (*t, *0) that are transmitted directly to AUDIX without switch intervention or interpretation; the user does not use the transfer button.

The caller can either transfer out to a default host destination administered on AUDIX by entering (*0), or to a caller specified destination by entering (*t).

For redirected calls, if the caller transfers out to a default host destination (*0), then the call will be treated as a redirected call, and all of the call related information (called party, redirection reason, etc.) will be displayed at the destination station. If the call is transferred out to a caller specified destination (*t), then the call will be treated as a direct call.

If the transfer is unsuccessful, AUDIX may inform the user of reasons for failure, such as invalid extension, too many digits, or station busy. If the transfer is successful, the call will be terminated at the extension supplied by AUDIX. From this point on, AUDIX is no longer involved in the call; when the call is dropped remotely, the user is NOT returned to AUDIX.

DCS AUDIX

If the destination of the transferred call is on the host switch, the operation is the same as the stand-alone switch. If the destination is on a remote switch, then the call is placed as a DCS call and the call will be treated as a direct call. If the caller transfers out to a default host destination on a remote switch, the call will appear as a direct call, and call related information (called party, redirection reason, etc.) will not be displayed at the destination station. Leave Word Calling cannot be used with DCS AUDIX after a Call Transfer Out of AUDIX.

Return Call

Stand-Alone Switch

A subscriber who has dialed into AUDIX, and accessed a message from another user, may invoke the AUDIX Return Call feature to return the call. The switch sees this as a Call Transfer out of AUDIX with reason for redirection of Direct Call.

DCS AUDIX

Use of this feature for DCS AUDIX is the same as for stand-alone switch. Also see the DCS Leave Word Calling interaction in this feature description.

Considerations

Only one AUDIX can be directly connected to a system. However, System 75 V3 and DEFINITY Generic 1 allow a system connected to and AUDIX to be a member of a DCS network, with that AUDIX serving the entire network, or any part of the network (see Figure 3-4).

The maximum number of analog ports provided by AUDIX is 32. This maximum is provided by a two-cabinet configuration. A one-cabinet configuration provides up to 16 ports.

AUDIX ports on the system must be assigned to a UCD hunt group. These ports are administered as 2500-type voice terminals. The hunt group can also be an ACD split.

AUDIX traffic is not restricted to the number of ports provided. Hunt group queuing allows more calls to be directed to AUDIX at one time than the unique number of ports provided. The System Administrator can set the size of the UCD hunt group queue (1 through 100).

In a DCS arrangement, the remote AUDIX hunt group(s) on the remote switch(es) is a dummy hunt group which passes only status and control information; they do not contain analog ports.

The system does not queue at the remote AUDIX hunt group **if the DCS data link is down** and treats the call as AUDIX unavailable.

There is no restriction in the system as to where in a coverage path that AUDIX can be placed; it can be first, last, second, etc., depending on the customer's requirements.

Coverage calls from a remote switch that reach AUDIX as a coverage point cannot be returned to the original coverage path on the remote switch.

If coverage to a remote AUDIX hunt group fails because all DCS trunks are busy or the DCS link is down, the system will attempt to terminate the call at a coverage point beyond the remote AUDIX, if one exists.

Transfer Into AUDIX cannot be used unless the principal's coverage path contains AUDIX as one of the coverage points.

A direct call to a hunt group member extension number (not the hunt group extension number) will not be answered by AUDIX.

Inaccurate CMS measurements may result if an ACD agent performs a conference with more than three parties on the call.

Interactions

The following features interact with the Audio Information Exchange (AUDIX) Interface feature.

- Abbreviated Dialing

  The Feature Access Code (FAC) for Transfer Into AUDIX may be programmed into an abbreviated dialing button.

- Attendant Conference

  An attendant that has split a call can conference the call with AUDIX by dialing the Transfer to AUDIX access code. The attendant presses the Release button to drop out of the conference call.

- Automatic Call Distribution (ACD)

  A hunt group can be administered as an AUDIX ACD split. AUDIX traffic measurements are then available utilizing the ACD Call Management System. Login occurs when AUDIX signals the switch to make a voice port available for AUDIX service and logout occurs when AUDIX signals the switch to disable the port.

  The AUDIX and ACD CMS must be connected to the same switch. If the AUDIX in the DCS feature is active, a CMS located on a switch other than the host switch (AUDIX location) will not provide measurements for the AUDIX ports.

  Because AUDIX frequently takes voice ports in and out of service for maintenance testing, high login activity may be seen for the AUDIX split in measurement reports.

  On CMS reports that display an agent's login identifier, AUDIX voice ports will always show a login identifier that is the same as the extension, even if login identifiers are not administered on the switch.

- Call Coverage

  When a coverage call successfully completes to AUDIX or is routed from a remote switch to the host switch because of coverage, the principal is dropped from the call (no temporary bridge appearance is maintained).

  Coverage calls from a remote switch that fail to reach AUDIX as a coverage point cannot be returned to the original coverage path on the remote switch.

- Call Forwarding

  An AUDIX user can forward calls to a remote AUDIX hunt group or to the host AUDIX hunt group.

  The system administrator must correctly administer the AUDIX destination for the remote AUDIX hunt group.

- Call Transfer

  A call transfer out of AUDIX can be to a Uniform Dial Plan (UDP) extension. If the destination extension is a UDP extension on a remote switch, the call is treated as a direct call.

  Calls may be transferred into AUDIX by users handling redirected calls for principals who are AUDIX subscribers.

- DCS Leave Word Calling

  In a DCS network, the called party may be on a different switch than the calling party. If the DCS link is down, attempts to store LWC messages are denied and intercept tone is returned. Leave Word Cancel requests are always denied for principals with AUDIX LWC; in some instances, the request may appear to be activated when it actually is not (see Leave Word Calling).

- Leave Word Calling

  The system administrator has the option of indicating that a principal's LWC messages are kept by AUDIX. This means that an LWC message left for a principal causes the extension of the calling and called parties to be reported to AUDIX. The principal can retrieve the message by calling AUDIX. The principal cannot retrieve the message using other retrieval methods (station display, demand print, Message Center agent, or synthesized voice), but will be notified of the existence of AUDIX messages via these methods.

  If the administrator assigns a principal's LWC to another messaging service, AUDIX can still report the existence of waiting LWC messages for the principal, but not the message content. This means that an LWC message left for a principal causes an indication of a waiting LWC message to be sent to AUDIX. The principal can retrieve the message using other retrieval methods (station display, demand print, Message Center agent, or synthesized voice). However, the principal will still be notified of the existence of AUDIX messages.

  If the data link between the system and AUDIX is down, attempts to activate Leave Word Calling for an AUDIX-covered principal are denied and reorder tone is returned.

  If a caller attempts to cancel an LWC message sent to AUDIX, the caller receives intercept tone if the called party is on the same switch. If the called party is on another switch in the DCS network, then the caller receives confirmation tone as long as the DCS data link to the called party's switch is operational, **even though the message will not actually be canceled.**

- Message Waiting Lamp (MWL) Activation/Deactivatlon

  The MWL interactions are the same whether the switch is a host switch or a remote switch. If a message is left for a principal on AUDIX, the switch lights the principal's MWL when AUDIX tells it there is an AUDIX message.

  If the principal retrieves the message, the switch extinguishes the AUDIX MWL only if the combined status of LWC, Message Center Service (MCS), and AUDIX indicate that there are no more messages.

- Personal Central Office Line (PCOL)

  A PCOL may not be covered by AUDIX.

- Ringback Queuing

  On direct calls to the remote AUDIX, where all trunks to the host AUDIX are busy, busy tone is returned. On coverage calls, if all trunks to the DCS host AUDIX are busy, AUDIX is treated as a busy coverage point. If there are coverage points after AUDIX, then the call will terminate there; otherwise, the call will remain at the principal. In summary, Ringback Queuing does not apply to AUDIX calls.

- Single-Digit Dialing and Mixed Station Numbering

  AUDIX is designed for use with a Uniform Dial Plan. It supports only one extension number length (3-, 4-, or 5-digit) that is used by AUDIX subscribers. Single-Digit and Mixed Station Numbering cannot be used. However, nothing prohibits connecting a switch to AUDIX that provides these features, as long as all AUDIX subscribers have the same extension number length.

- Temporary Bridged Appearance

  Stations that normally would have a temporary bridged appearance with their coverage point will not, if the coverage point is AUDIX.

- Voice (Synthesized) Message Retrieval

  Retrieval of LWC messages via Voice Message Retrieval is separate and distinct from AUDIX voice message retrieval. LWC messages left for a Principal on AUDIX may not be accessed via Voice Message Retrieval; however, the invoker of Voice Message Retrieval will be told if there are any new messages for the principal on AUDIX; it will **voice** that there are message center messages (dialing 8-callout will call AUDIX), and the display retrieval will display "Message Center AUDIX Call". The LWC Messages accessible to Voice Message Retrieval are inaccessible to AUDIX; but AUDIX will inform the invoker that the messages exist.

Administration

AUDIX Interface is administered by the System Manager or the CSSO. The following forms require administration; specific inputs shown here are unique to AUDIX; where AUDIX is not specified, the inputs are determined by the particular system arrangement:

- Interface Data Module (for SCI—V3 and XEV3)
  - Assign Data Module (Interface) when SCI is provided by TN716, TN738, and TN719 circuit packs **or by** the TN765 circuit pack.
- Processor Data Module (for SCI—G1)
  - Assign Data Module (Processor Interface) when SCI is provided by TN765 circuit pack.
- Interface Link
  - Assign link extension number for Interface (V3 or XEV3) or Processor Interface (G1).
  - Assign Destination Number = extension number of MPDM assigned to AUDIX (when MPDM is used).
  - Assign Destination Number = eia (XEV3, or G1), when PI jack is used for AUDIX.
  - Assign DTE/DCE = DTE (for AUDIX).
  - Assign Identification = AUDIX if desired (this field may be left blank).
- Modular Processor Data Module
  - Assign MPDM (when provided).
- Modular Trunk Data Module
  - Assign MTDM (when provided).
- Processor Channel
  - Use Proc Chan 59 for AUDIX
  - Assign Interface Link (1-4—V3, XEV3, and G1 with single-carrier cabinets; 1-8—G1 with multi-carrier cabinet).
  - Assign Interface Chan (1-64).
  - Assign Priority = h (for AUDIX).
  - Assign Remote Proc Channel (1-64).

- Assign Appl. = audix
- Assign PBX-ID (1-64). This is the PBX-ID number associated with the PBX.

• Voice Terminal 2500 (Station Form)—for AUDIX analog port assignment

- Assign extension number = Any unused number that agrees with the dial plan (for AUDIX analog ports).
- Assign Type = 2500
- Assign Port = Port number of AUDIX analog port.
- Assign Name = audix
- Assign COR = Same as COR of AUDIX hunt group.

• Hunt Group—Host (When this switch **is not part** of a **DCS** AUDIX configuration or when this switch is the **Host Switch** in a DCS AUDIX configuration).

- Assign Group Number (Identifies hunt group to system software).
- Assign Group Extension Number.
- Assign Group Type = ucd
- Assign Group Name = for example, audix.
- Assign COR (0-63 — identifies class of restriction of hunt group and hunt group members).
- Assign Message Center = audix
- Assign ACD = y, if AUDIX is an ACD split; = n, if it is not.
- Assign Queue = y
- Assign Queue Length (typically queue length equals the number of audix ports assigned). This number should be large enough so that all callers during peak time will be queued and not given busy treatment.
- Assign Measured By MIS = y, if hunt group traffic data is to be measured by CMS and ACD is assigned; otherwise, = n.
- Assign Hunt Group Members (extension numbers assigned to AUDIX analog ports)

• Hunt Group—Remote (When this switch is a **Remote Switch** in a DCS AUDIX configuration).

- Assign Group Number (Identifies hunt group to system software).

- Assign Group Extension Number.
- Assign Group Type = ucd
- Assign Group Name = for example, AUDIX hu gp.
- Assign COR = Same as COR of Host Switch AUDIX hunt group.
- Assign Message Center = rem-audix.
- Assign ACD = n.
- Assign Queue = n.
- Assign Audix Extension = Host Switch AUDIX Hunt Group extension number.

• Coverage Path

- Assign AUDIX Hunt Group extension number to Point 1, Point 2, or Point 3 as required (it is recommended that AUDIX be placed at the end of the coverage path; however, this is not a requirement).

• Hop Channel on Host Switch (for each Remote Switch, when remote AUDIX is provided).

- Assign Link (two fields) 1-4.
- Assign Chan (two fields) 1-64.
- Assign Priority = h (for AUDIX).

• Class of Service

- Select or Administer a Class-of-Service code (to be assigned to AUDIX user's voice terminals) that allows "Call Forwarding All Calls".

• Feature Access Code (FAC)

- Assign Transfer into AUDIX = Any unused 1-, 2-, or 3-digit feature access code that agrees with the dial plan. However, the Transfer into AUDIX feature access code should not be administered to have the same first digit as another feature access code with a longer length.

• Voice Terminal (AUDIX User)

- Assign COS = COS code previously assigned on Class-of-Service form that allows Call Forwarding All Calls, if desired.
- Assign Coverage Path = Coverage Path number associated with AUDIX hunt group.

- — Assign LWC Reception = "audix" if LWC messages are to be stored on AUDIX. Enter "ap-spe" if voice message retrieval or display message retrieval is to be used.
- — Assign LWC Activation = y
- — Assign Redirect Notification = y

- Voice Terminal (AUDIX User Button Assignment)
  - — Assign Call Forwarding = call-fwd (optional).
  - — Assign Call Coverage—Go To Cover = goto-cover (optional).
  - — Assign Call Coverage—Send All Calls = send-calls (optional).
  - — Assign Leave Word Calling—LWC = lwc-store (optional).
  - — Assign Abbreviated Dialing—AD = abrv-dial ("List:" and "DC:" as assigned on the Feature Access Code form).
- Dial Plan
  - — The host switch PBX ID must be the same as the host switch PBX ID administered on the AUDIX.

Hardware and Software Requirements

AUDIX analog ports (may be up to 32 ports) on System 75 V3, DEFINITY Generic 1, and System 75 XEV3 are provided by TN742 circuit packs.

AUDIX data link hardware is required as follows. Some of this hardware may already be provided; for instance, the Switch Communication Interface (SCI) circuit pack(s) may already be in place for other adjuncts, a vacant port may be available on a digital line circuit pack, etc. If the hardware shown is not already in place, it must be provided.

- System 75 V3 connected to a two-cabinet AUDIX

  AUDIX data link connection between System 75 V3 and a two-cabinet AUDIX requires one port on a TN754 or TN784 (G1) Digital Line circuit pack and an MTDM and an MPDM (see Note). SCI is provided by either the TN716 Interface 1, TN738 Interface 2, and TN719 Interface 3 circuit packs **or** the TN765 Processor Interface circuit pack.

  Note: These MPDMs are required on the AUDIX side of the connection.

- System 75 V3 connected to a one-cabinet AUDIX

  AUDIX data link connection between System 75 V3 and a one-cabinet AUDIX requires one port on a TN754 or TN784 (G1) Digital Line circuit pack and an MPDM. SCI is provided by either the TN716 Interface 1, TN738 Interface 2, and the TN719 Interface 3 circuit packs **or** the TN765 Processor Interface circuit pack.

- System 75 XEV3 connected to a two-cabinet AUDIX

  AUDIX data link connection between System 75 XEV3 and a two-cabinet AUDIX is provided by direct connection to the PI jack (connected to the TN765 Processor Interface circuit pack) through an MPDM (see Note). SCI is provided by the TN765 Processor Interface circuit pack. If the PI jack is already in use for another adjunct, one port on a TN754 or TN784 (G1) Digital Line circuit pack and an MTDM and an MPDM (see Note) are required.

- System 75 XEV3 connected to a one-cabinet AUDIX

  AUDIX data link connection between System 75 XEV3 and a one-cabinet AUDIX is provided by connection to the PI jack (connected to the TN765 Processor Interface circuit pack); in this case, an MPDM is not required. SCI is provided by the TN765 Processor Interface circuit pack. If the PI jack is in use for another adjunct, one port on a TN754 or TN784 (G1) Digital Line circuit pack and an MPDM are required.

- DEFINITY Generic 1 connected to a two-cabinet AUDIX

  AUDIX data link connection between DEFINITY Generic 1 and a two-cabinet AUDIX is provided by direct connection to the PI jack (connected to the TN765 Processor Interface circuit pack) through an MPDM (see Note). SCI is provided by the TN765 Processor interface circuit pack. If the PI jack is already in use for another adjunct, one port on a TN754 or TN784 (G1) Digital Line circuit pack, an MTDM, and an MPDM (see Note) are required.

- DEFINITY Generic 1 connected to a one-cabinet AUDIX

  AUDIX data link connection between DEFINITY Generic 1 and a one-cabinet AUDIX is provided by connection to the PI jack (connected to the TN765 Processor Interface circuit pack);. in this case, an MPDM is not required. SCI is provided by the TN765 Processor Interface circuit pack. If the PI jack is already in use for another adjunct, one port on a TN754 or TN784 (G1) Digital Line circuit pack and an MPDM are required.

  Note: These MPDMs are required on the AUDIX side of the connection.

For DCS AUDIX, DCS software is required.

Authorization Codes (V3 or G1)

Description

Provides the means for extending control of system users' calling privileges.

The Authorization Codes feature is optional, is closely linked to the Facility Restriction Level (FRL) feature, and can be used with the Automatic Route Selection (ARS), Automatic Alternate Routing (AAR), and Remote Access features, as well as with incoming trunk calls.

Authorization codes may be used for any or all of the following reasons:

- To allow a calling user to override the FRL assigned to the originating Class of Restriction (COR)
- To allow a calling user to override the assigned originating FRL on AAR or ARS calls
- To restrict individual incoming tie trunks and remote access trunks from accessing an outgoing trunk
- To identify certain calls on Station Message Detail Recording (SMDR) records for cost-allocation purposes
- To provide additional security control for the system.

When an authorization code is dialed, the FRL assigned to the extension number, attendant console, incoming trunk group, or remote access trunk group being used for the call, is replaced by the FRL assigned to the authorization code. The new FRL functions the same as the one it replaces; however, the new FRL may represent greater or lesser calling privileges than the FRL that it replaces. Access to any given facility depends on the restrictions associated with the authorization code FRL.

For example, a supervisor may be at a desk of another user and want to make a call that is not normally allowed by the FRL assigned to that extension. The supervisor, however, can still make the call by dialing an authorization code that has been assigned an FRL that is not restricted from making that type call.

For security reasons, authorization codes range from four to seven digits. The number of digits in the codes must be a fixed length for a particular switch. As many as 5,000 codes can be administered.

Incoming trunk groups within a system may be administered to always require an authorization code. The system applies recall dial tone to a call when the user must dial an authorization code. If the user dials the authorization code within 10 seconds (inter-digit time-out), the call will either complete as dialed, route to the attendant, or route to intercept tone, depending on system administration.

Normally, Direct Inward Dialing (DID) trunks should not require authorization codes. However, it can be done and care should be taken when administering DID trunks to require an authorization code, because different type calls could terminate at different endpoints, and requiring an authorization code could be confusing to the caller.

A Cancellation of Authorization Code Request (CACR) may be administered. The CACR cancels the 10-second interval between dialing. When the CACR is dialed, the call immediately routes according to system administration. Incoming trunk calls receive intercept treatment or go to the attendant. Other calls receive intercept treatment unless the user's FRL is high enough to route the call. A CACR from an off-premises extension over DID/Tie trunks uses DID/Tie trunk intercept treatment. Internal calls receive intercept tone.

With G1, the System Manager can obtain a printed list of the system's authorization codes by entering **list auth p** at the Manager I terminal.

AAR and ARS Calls

Each authorization code is assigned a COR that contains an associated FRL. Within a system, access privileges are determined by the FRL assigned to the facility where the call is originated. When an AAR/ARS call is dialed, the system allows or denies the call based on that originating FRL. Class of Restriction (COR) is used to restrict internal or non-AAR/ARS calls.

Authorization codes are given to individual users and provide a method of specifying the level of calling privileges for that user regardless of the originating facility. Once an authorization code is required and dialed on an AAR/ARS call, the FRL assigned to the authorization code becomes the originating FRL and controls and defines the user's privileges.

An AAR or ARS call originated by a system user or routed over an incoming tie trunk may require a dialed authorization code to continue routing. If authorization codes are always required, then an authorization code must be dialed even if the originating FRL was adequate to complete the call.

Extreme care should be taken when administering authorization codes, so that a user does not have to dial the authorization code more than once. For example, if a user makes an AAR or ARS call and the user's FRL is not high enough to access any of the trunks in the routing pattern, the system will prompt the user for an authorization code. If the FRL assigned to the authorization code is high enough to access the next trunk group in the routing pattern, the user should not be required to dial the code again. If AAR or ARS continues to route the call, the user may be required to dial an authorization code again. This type of situation can be avoided through careful administration.

When an authorization code is required on some, but not all, trunk groups, the system will prompt for an authorization code when the originating FRL is not adequate to access the next available trunk group in the routing pattern.

Remote Access Calls

When a remote access caller dials the assigned remote access number and establishes a connection to the system, the system may request the caller to dial an authorization code and/or a barrier code. The authorization code defines the caller's calling privileges within the system.

If the uses of authorization codes are specified, they apply to all remote access trunk groups in the system. If a remote access user must dial an authorization code to gain access to the system facilities, an authorization code will not be requested again even if the user places a call that routes through the ARS or AAR feature.

The system may be administered for a Timeout to Attendant option. This option routes a remote access call to the attendant if the user fails to dial within 10 seconds after receiving the system request for an authorization code. Also, the remote access user can dial the CACR code, if administered, which cancels the 10-second time-out interval. In this case, the call routes immediately to the attendant. If an off-premises user on a DID/tie trunk cancels an authorization code, DID/tie intercept treatment is received.

Considerations

From a remote location, all authorization codes as well as all barrier codes (if required) are normally entered using touch-tone dialing. However, rotary dialing may be used in some cases, depending on where the authorization code is forced and how the trunks are administered. A user with a rotary dial telephone can also dial the Listed Directory Number (LDN) for access to the attendant or, after dialing the remote access number, wait 10 seconds for Timeout to Attendant. In either case, the attendant must extend the incoming call.

The Authorization Codes feature is entirely in addition to, and in no way limits, other methods of call control such as Toll Restriction, Miscellaneous Trunk Restriction, and Outward Restriction.

For security reasons, authorization codes must be assigned randomly. This also makes it difficult for one user to guess the authorization code assigned to another user.

A CACR code, if administered, can be either the # symbol or the digit 1. The # symbol is used when the tandem and main switches are System 75s or DEFINITY Generic 1s. If a System 85, DIMENSION PBX, or DEFINITY Generic 1 switch is part of the network, then the digit 1 is used as the CACR code. If the digit 1 is used as the CACR code, then it cannot also be used as the first digit of an authorization code.

If the Timeout to Attendant option is not administered and if a user dials the CACR code instead of an authorization code, the system assumes that an invalid authorization code was dialed and routes the call to intercept tone.

Calling privileges are affected by the Authorization Codes feature as follows:

- For incoming trunk calls, where an authorization code is required due to administration on the trunk group form, the authorization code does not change the privileges of the user in any way.
- For outgoing calls, where the FRL of the user is insufficient for accessing the routing pattern preference assigned by AAR/ARS, the authorization code will change the FRL of the user only. The FRL used is the one assigned to the COR which is associated with the authorization code entered. No other data assigned to that COR is assigned to the user.

- For remote access calls, where the user is required to enter an authorization code, the user will be assigned the COR of the dialed authorization code, with all connected data, such as the FRL. This COR will override the COR assigned to a barrier code, if a barrier code is also required.

Interactions

The following features interact with the Authorization Codes feature.

- AAR/ARS Partitioning

  Since Partitioning Group Numbers (PGNs) are assigned according to Class of Restriction (COR) and Authorization Codes can change a COR, PGNs can be changed on incoming remote access calls by the use of authorization codes. On originating calls, the user's COR determines the PGN.

- Class of Restriction (COR) and Facility Restriction Level (FRL)

  When an internal system user dials an authorization code on an AAR/ARS call, the FRL associated with the authorization code overrides the FRL assigned to the system user.

  When a remote access user dials an authorization code, the associated COR determines the caller's access privileges to the system's features and services.

- Forced Entry of Account Codes and Station Message Detail Recording (SMDR)

  With V3, the authorization code is output only if the administered account code length is less than six digits. With G1, on the 94A LSU and 3B2 CDRU 18-word records, the authorization code is output only if the administered account code length is less than six digits in length. With G1, on the 59-character record, the authorization code is never recorded.

  When an authorization code is required after the destination address is dialed, that code will be recorded. Thus, all unauthorized attempts to dial an invalid authorization code will be recorded, and a pattern of such calls can be traced using the SMDR printouts.

Administration

The use of authorization codes is optional. However, if authorization codes will be used, the following items must be administered by the System Manager:

- Authorization Code Parameters
  - Enable the Authorization Codes feature

- — Authorization code length—Can be from four to seven digits, and all authorization codes must be the same length
- — CACR—Choice is the digit 1 or the # symbol
- — Whether or not the Timeout to Attendant option will be used
- — The authorization codes themselves—This is a list of all authorization codes and their associated CORs. As many as 5,000 codes may be used. Authorization codes should be selected randomly and cannot begin with the digit 1 if the digit 1 is used as the CACR code.

- Remote Access
  - — Whether or not an authorization code will be required on a remote access call
  - — Whether or not the system will apply recall dial tone to request that an authorization code be dialed.
- AAR and ARS
  - — Assign COR FRLs and Routing Pattern FRLs so that no more than one authorization code is required when making an AAR/ARS call.
- Trunk Groups
  - — Whether or not each incoming or two-way trunk group requires an authorization code for incoming calls on that trunk group to complete to their destination.

Hardware and Software Requirements

No additional hardware is required.

Optional Authorization Codes software is required. Also, optional ARS software is required if Authorization Codes are to be used to access the public network.

Automatic Alternate Routing (V2, V3, or G1)

Description

Provides alternate routing choices for private on-network calls. Also provides digit modification to allow on-network calls to route through the public network when on-network routes are not available.

Automatic Alternate Routing (AAR) provides up to 6 routes for each of the 640 possible private network office codes (RNXs). To use AAR, the user dials the AAR access code and the called number. Feature operation is completely transparent to the user. The AAR access code is normally the digit 8. The called number may be a 7-digit on-network number, a 10-digit public network number, a service code, an International Direct Distance Dialing (IDDD) number, an operator code (0), or a customer-dialed and operator-serviced (CDOS) number (0+ or 01+ the number).

On-network numbers are handled by the AAR feature. All other numbers are directed to the Automatic Route Selection (ARS) feature for processing. An on-network number can be changed into a 7- or 10-digit public network direct distance dialing number, a CDOS number, or an IDDD number, depending on the route selected.

The 640 private network RNXs may match public network central office codes (NXXs). Therefore, the only way to determine the intended network for 7-digit calls is by the dialed AAR or ARS access code. The system can recognize 10-digit public network calls because an RNX never matches an Area Code. When the system detects an Area Code, the call is routed using ARS tables.

The principal use of AAR is to provide routing of private network calls, that is, calls that originate and terminate at a customer location without accessing the public network. The normal scenario is as follows: The calling party dials the AAR access code followed by a 7-digit on-network number. AAR then routes the call to the on-network switch serving the calling party.

AAR and Subnet Trunking provide a convenient means to place IDDD calls to a frequently called foreign city. Such calls route as far as possible over the private network before exiting the network. The RNX is, of course, reserved to represent a particular country and city. At the final on-network switch, the RNX is deleted. The international prefix code (011), the country code, and the city code are inserted. The inserted digits plus the last four digits of the originally dialed number constitute the IDDD number. Subnet Trunking, which also has ARS applications, is described separately in this chapter.

Similar to the IDDD case, certain domestic calls may reach a point on the network where they can route no further, because tie trunks to the next switch are busy or none are provided. In this case, the RNX can be deleted and the appropriate public network code inserted. Calls of this type route off-network via a central office. The central office may be connected to either an Electronic Tandem Network (ETN) tandem or main switch. Toll charges, if any, are from the final ETN switch to the destination.

Assuming an AAR access code of 8, when the system user dials a number of the form 8-RNX-0111 and the RNX is a local RNX (on the same switch as the user), the system will route the call to the attendant group on the local switch. If the RNX is for a distant switch and the call tries to access the public network, one of the following will occur, thus allowing attendant-seeking calls that are overflowing to the public network to be treated differently than station-seeking calls:

- If a CO, FX, or WATS trunk group is selected for the call and the number of digits deleted in the routing pattern is not 7, the trunk group is considered busy and will be skipped over in the routing pattern.

- If the number of digits deleted in the routing pattern is 7, the routing pattern instructions will be followed.

Each RNX can point to any one of 254 Routing Patterns, numbered 1 through 254. More than one RNX can point to the same pattern. A blank pattern provides intercept treatment and pattern 254 is the default for all RNXs. Routing Patterns are shared with ARS. Access to a route within the pattern is controlled by Facility Restriction Level (FRL) assignments. FRLs are described fully elsewhere in this chapter.

The system may serve as an ETN tandem switch. In this case, the system can access or be accessed by Intertandem Tie Trunks to/from other tandem switches and/or Access Tie Trunks to/from ETN main switches. The system can also access Bypass Tie Trunks to an ETN main switch. This distinction as a tandem switch is important with respect to the routing of certain calls.

Considerations

AAR provides efficient use of private network facilities.

AAR provides up to 254 Routing Patterns, each containing up to six routing preferences. Patterns are shared with ARS.

Up to 640 RNXs can be provided. An RNX can represent an actual location on the network, or can be a dummy code to be converted into a public network or IDDD number.

If a customer changes ARS routing assignments, it is the customer's responsibility to notify the Regional Support Center (RSC) network designer and the System Control Office (SCO) technician of the changes in order to receive their continued support.

If a system is the last ETN tandem switch for a main ETN switch that has no tie trunks, but has DID trunks, then digit deletion/insertion can be used to route calls to the ETN main switch.

Interactions

The following features interact with the Automatic Alternate Routing feature.

- Uniform Dial Plan (UDP)

  The leading 1 to 4 digits of the 4- or 5-digit called Distributed Communications System (DCS) extension (PBX Code on Dial Plan form) are converted into an RNX. RNX tables are used to route the Uniform Dial Plan (UDP) call.

- Automatic Route Selection (ARS)

  ARS and AAR can access the same trunk groups and share the same Routing Patterns.

- Abbreviated Dialing

  FRL checking is bypassed on an AAR call made via a privileged Abbreviated Dialing Group List.

- Attendant Control of Trunk Group Access

  Attendant control of a trunk group, in effect, removes a trunk group from the Routing Pattern. A controlled trunk group is never accessed by AAR.

- Authorization Codes

  An AAR or ARS call originated by a system user or routed over an incoming tie trunk may require a dialed authorization code to continue routing. If authorization codes are always required, then an authorization code must be dialed even if the originating FRL was adequate to complete the call.

- Code/Toll Restriction

  Code/Toll Restriction is not checked on AAR calls.

- Controlled Restriction, Origination Restriction, and Outward Restriction

  These features prohibit access to AAR.

- Miscellaneous Trunk Restrictions

  Miscellaneous Restrictions are not checked on AAR calls.

- Ringback Queuing

  Ringback Queuing can be used on AAR calls originated at the switch that provides the queuing. Incoming tie trunk calls will not queue on an outgoing trunk group.

- Station Message Detail Recording (SMDR)

  An AAR call using a trunk group marked for SMDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number.

  Subnet Trunking does not affect SMDR.

  The originating FRL associated with the call is recorded. However, if 15-digit SMDR account codes are used, the FRL value is overwritten.

  If SMDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group. If an SMDR account code is to be dialed with an ARS call, it must be dialed before the ARS access code is dialed.

Administration

AAR is initially assigned on a per-system basis by an AT&T service technician. After the feature is activated, the following items are administered by either the System Manager or the service technician:

- AAR Access Code (one to three digits)
- RNX Translation Table—Points to the appropriate Routing Patterns. Pattern Number 254 is initially assigned to all RNXs.
- Routing Patterns—In addition to normal trunking data, provides subnetwork trunking information which extends a call through a chain of subtending switches. (See Subnet Trunking for details.)
- FRLs—Must be assigned via a Class of Restriction to each originating facility. The minimum FRLs required to access a route are assigned as part of the Routing Pattern. Assignment of these values determines the calling privileges of each individual user of the ETN.
- Whether or not the system returns dial tone after the AAR FAC is dialed on trunk calls.

Hardware and Software Requirements

AAR may require additional tie trunks. These additions are, however, cost effective when compared to the other alternatives for call routing.

AAR is provided as a part of the optional Private Networking software.

Automatic Callback

Description

Allows internal users who placed a call to a busy or unanswered internal voice terminal to be called back automatically when the called voice terminal becomes available.

A single-line voice terminal user activates Automatic Callback by pressing the Recall button or flashing the switchhook and then dialing the Automatic Callback access code. Only one Automatic Callback call can be activated at any given time by a single-line user.

A multi-appearance voice terminal user can activate Automatic Callback for the number of Automatic Callback buttons assigned to the terminal. After placing a call to a voice terminal that is busy or that is not answered, the caller simply presses an idle Automatic Callback button and hangs up.

When Automatic Callback is activated, the system monitors the called voice terminal. When the called voice terminal becomes available to receive a call, the system then originates the Automatic Callback call. A busy voice terminal becomes available when the user hangs up after completing the current call. An unanswered voice terminal becomes available after it is used for another call and is then hung up.

When the called voice terminal becomes available, the system originates the Automatic Callback call and the calling party receives 3-burst ringing. The calling party then lifts the handset and the called party receives the same ringing provided on the original call. The ringing at the called voice terminal occurs immediately after the calling voice terminal user lifts the handset.

If the calling voice terminal user answers an Automatic Callback call, and for some reason the called extension cannot accept a new call, the calling user will hear confirmation tone and then silence. The call will still be queued.

Considerations

The system can process a maximum of 40 (V1), 80 (V2 or V3), or 160 (G1) callback calls at one time.

An Automatic Callback request will be canceled for any of the following reasons:

- The called party is not available within 30 minutes.
- The calling party does not answer the callback call within the administered interval (two to nine ringing cycles).
- The calling party decides not to wait and presses the same Automatic Callback button a second time (multi-appearance voice terminal) or dials the Automatic Callback cancellation code (single-line voice terminal).

Automatic Callback eliminates the need for voice terminal users to continually re-dial busy or unanswered calls to internal voice terminals. Instead, the user simply activates Automatic Callback. The system then calls the user back when the called voice terminal becomes available.

Automatic Callback is administered to individual voice terminals by their Class of Service and cannot be assigned to the attendant(s).

Multi-appearance voice terminals must have an Automatic Callback button to activate the feature.

Interactions

The following features interact with the Automatic Callback feature.

- Bridged Call Appearance

  Automatic Callback calls cannot originate from a bridged call appearance. When a call is originated from a primary extension number, the return call notification rings at all bridged call appearances.

- Call Coverage

  Automatic Callback calls do not redirect to coverage.

- Call Pickup

  A group member cannot answer a callback call for another group member.

- Call Forwarding All Calls

  Automatic Callback cannot be activated toward a voice terminal that has Call Forwarding activated. However, if Automatic Callback was activated before the called voice terminal user activated Call Forwarding, the callback call attempt is redirected toward the forwarded-to party.

- Attendant Call Waiting and Call Waiting Termination

  If Automatic Callback is activated to or from a single-line voice terminal, the Call Waiting features are denied.

- Ringback Queuing

  An Automatic Callback button is used to activate the Ringback Queuing feature.

Voice terminals with the following features cannot activate Automatic Callback:

- Hot Line Service

- Manual Originating Line Service
- Restriction—Origination.

Automatic Callback cannot be activated to the following:

- The attendant console group
- A voice terminal assigned Termination Restriction
- An extension with Automatic Callback already activated toward it
- A data terminal (or data module)
- A Direct Department Calling group
- A Uniform Call Distribution group
- A Terminating Extension Group.

Administration

The System Manager assigns Automatic Callback to individual voice terminals by their Class of Service. The following items also require administration:

- No Answer Time-Out Interval (number of times the callback call rings before it is canceled). This interval is assigned on a per-system basis.
- Feature Access Codes—For activating and deactivating Automatic Callback
- Automatic Callback Buttons—For multi-appearance voice terminals

Hardware and Software Requirements

No additional hardware or software is required.

Automatic Call Distribution (V3 or G1)

Description

Provides automatic connection of incoming calls to specific splits (hunt groups). Calls to a specific split are automatically distributed among the agents (hunt group members) assigned to that split. Automatic Call Distribution (ACD) data, transmitted from the switch to the Call Management System (CMS), is used to generate various reports on the status of ACD agents, splits, and trunks.

An ACD split is simply a hunt group that is designed for use wherever a high volume of similarly natured calls are received. An ACD split can use either of two hunting algoriths (depending on administration) to select an idle available terminal or console. The two types of hunting that can be used are "direct" hunting and "most-idle agent" hunting.

If a split is administered for direct hunting, an incoming call rings the first available extension number in the administered sequence. If the first split agent in the sequence is active on a call (busy), or is not available due to one of the ACD call work modes (described later), the call routes to the next split agent, and so on. In other words, incoming calls always try to complete at the first split agent in the administered sequence. Therefore, the calls are not evenly distributed among the split agents.

If a split is administered for most-idle agent hunting, an incoming call will ring the available split agent that has not completed an ACD split call for the longest period of time (the most-idle agent). In other words, incoming calls to an ACD split extension number will be distributed evenly among the split agents. For this reason, most-idle agent hunting is usually preferred over direct hunting.

Members of a split are called agents. An agent can be a voice terminal extension or individual attendant extension. A voice terminal or individual attendant can be an agent in one or more splits.

In addition to the agents, a split supervisor can be assigned to each split. The split supervisor can monitor the split queue (described later) via queue warning buttons (see Queue Status Indications feature) and can assist agents on ACD calls. Although split supervisors can assist agents on ACD calls, the supervisors themselves do not normally receive ACD calls unless they are also members of the split. The request for assistance comes from the agents. An agent can request supervisory assistance by pressing an Assist button or dialing the Assist feature access code.

Split Queuing and Announcements

A queue can be established for an ACD split. When all agents in the split are active or not available to receive an ACD call (for example, AUX work), the queue allows incoming calls to await an idle terminal. If an agent becomes available while an incoming call is in the split queue, the call is automatically connected to the available agent.

Two announcements can be assigned to each split. The second announcement can be administered so that it will repeat itself.

When an incoming call is directed to an ACD split, the call, depending on the administration of the split, will either try to access a split agent or will automatically be connected to the first announcement (Forced First Announcement), if available.

Forced First Announcement: The first announcement delay interval (0 to 99 seconds) indicates how long a call will remain in queue before the call is connected to the first announcement. If this interval is set to "0" seconds, the incoming call will automatically be connected to the first announcement, if available. The result is a "forced first announcement," and the call will not attempt to access an agent until after the first announcement is heard.

When a forced first announcement is assigned, the system tries to connect the incoming call to the first announcement, with the results being one of the following:

- If the first announcement is available, the caller receives audible ringing followed by the first announcement. The system then tries to connect the call to an agent.
- If the announcement is busy and has no queue, the system will wait 10 seconds and then try to access the announcement again.
- If the announcement is busy and has a queue, one of the following happens:
  - If the queue is full, the system will wait 10 seconds and then try to access the announcement again.
  - If the queue is not full, the call enters the announcement queue and the caller receives audible ringing until the first announcement is heard. The system then tries to connect the call to an agent.
- If the announcement is not busy, but is still unavailable (it might have been deleted), then the system tries to connect the call to an agent.

Entering the Queue: When a forced first announcement is not assigned, the system will try to connect an incoming call to an available agent. If an agent is available, the call is connected to the agent. If all agents in the split are active, and the incoming facility is a Direct Inward Dialing (DID), tie, or DS1 tie trunk, the call enters the split queue. If all agents in the split are active, and the incoming facility is a Central Office (CO) trunk, the caller will hear ringing. If a split queue is not assigned or if the queue is full, the caller receives busy tone (unless the call is a Central Office call) or the call is redirected by Call Coverage or by the Intraflow feature (described later) associated with Call Coverage.

First Announcement: After a call enters a split queue, the caller receives audible ringing and the first announcement delay interval begins. (If there is no first announcement, the second announcement delay interval begins. If there is no second announcement, the call remains in queue until answered or removed from the queue.) If an agent becomes available during the first announcement delay interval, the call is connected to the available agent. Otherwise, the first announcement delay interval expires and the system tries to connect the incoming call to the first announcement, with the result being one of the following:

- If the first announcement is available, the caller receives audible ringing followed by the first announcement.
- If the announcement is busy and has no queue, the caller receives audible ringing and the first announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
- If the announcement is busy and has a queue, one of the following happens:
  - If the queue is full, the caller receives audible ringing and the first announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
  - If the queue is not full, the call enters the announcement queue and the caller receives audible ringing until the first announcement is heard. The system then tries to connect the call to an agent.
- If the announcement is not busy, but is still unavailable (it might have been deleted), the second announcement delay interval begins and the system attempts to connect the call to the second announcement. If there is no second announcement, the call will remain in queue until answered or removed from the queue.

Second Announcement: After the first announcement has completed, the second announcement delay interval begins and the caller hears music (only if the first announcement is not a forced first announcement, in which case the caller hears ringing), if provided. (If there is no second announcement, the call remains in queue until answered or removed from the queue.) If an agent becomes available during the second announcement delay interval, the call is connected to the available agent. Otherwise, the second announcement delay interval expires and the system tries to connect the incoming call to the second announcement, with the result being one of the following:

- If the second announcement is available, the caller receives audible ringing (only if the first announcement has not been heard) followed by the second announcement.
- If the announcement is busy and has no queue, the caller receives audible ringing and the second announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
- If the announcement is busy and has a queue, one of the following happens:
  - If the queue is full, the caller receives audible ringing (only if the first announcement has not been heard) and the second announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
  - If the queue is not full, the call enters the announcement queue and the caller receives audible ringing (only if the first announcement has not been heard) until the second announcement is heard. The system then tries to connect the call to an agent.

- If the announcement is not busy, but is still unavailable (it might have been deleted), the call will remain in queue until answered or removed from the queue.

After the second announcement is heard, the caller hears music (if provided) or silence (if music is not provided), and one of the following occurs:

- If the split has been administered so that the second announcement is repeated, the system will attempt to connect the call to the second announcement after the delay expires.

- If the split has been administered so that the second announcement is not repeated, the call will remain in queue until answered or removed from the queue.

Forced Disconnect: At times, it may be desired to connect an incoming call directly to an announcement and then disconnect the call after the announcement has completed. This can be accomplished two ways:

- The incoming destination can be administered as an announcement extension. This way the calling party will hear the announcement and be disconnected. Also, the call is never queued for a split because it goes directly to the announcement.

- An announcement extension can be administered as a point in a split's coverage path. This way, calls that have been in the queue for a long period of time are forced to go directly to the announcement and are then disconnected.

Intraflow and Interflow: The Intraflow feature allows splits to be assigned coverage paths. Also, a split can be a part of a coverage path. Thus, the Call Coverage feature can be used to redirect ACD calls from one split to another split according to the coverage path's redirection criteria. For instance, a split's coverage path can be administered so that incoming ACD calls are automatically redirected to another split during busy or unanswered conditions.

If Intraflow is provided, the Coverage Don't Answer Interval (1 to 99 ringing cycles) associated with Call Coverage may begin when the call enters the split queue. If the Coverage Don't Answer Interval expires before either of the two announcement delay intervals expires, the call is redirected to coverage. If no coverage point is available to handle the call, the call remains in queue and may then be connected to a delay announcement. If either of the announcement delay intervals expires before the Coverage Don't Answer Interval, the call is connected to a recorded announcement, if available, but the call will still go to coverage after the announcement.

The Interflow feature allows ACD calls to be redirected from one split to a split on another switch. This is accomplished by forwarding calls to an off-premises location via the Call Forwarding All Calls feature.

For a detailed description of the Call Forwarding feature and the Intraflow and Interflow feature, see the individual descriptions in this chapter.

Queue Status Indications

The system provides queue status indications for ACD calls based on the number of calls in queue and time in queue. These indications are provided via lamps assigned to the terminals or consoles of split agents or supervisors. In addition, an auxiliary warning lamp can be provided to track queue status based on time in queue and another for number of calls in queue. Also, display-equipped voice terminals and consoles can display the time in queue of a split's oldest call and the number of calls in that split's queue. For more detailed information, see the Queue Status Indications feature description elsewhere in this section.

Priority Queuing

Priority Queuing allows calls with increased priority to be queued ahead of calls with normal priority. Priority Queuing can be provided two ways:

- A calling party's Class of Restriction (COR) can be assigned Priority Queuing.
- An ACD split can be assigned Priority on Intraflow. This allows calls from the split, when intraflowed into another split, to be queued ahead of nonpriority calls already queued in the other split.

Agent Call Handling

Agent Call Handling is a separate feature that includes the various agent functions and operations. For details, see the Agent Call Handling feature description elsewhere in this chapter. The following is a brief summary of the Agent Call Handling functions and operations.

- Agent Log-in and Log-out—An agent is required to log in before he or she is able to receive ACD calls. The agent may or may not be required to enter a personal identification number, depending on administration. In addition, an agent can log out to let the system know that he or she is unavailable for ACD calls.
- Agent Answering Options
  - Automatic Answer With Zip Tone—An agent with the automatic answering option can be connected directly to incoming calls without audible ringing. It is recommended that this feature be used with a headset. In this case, the agent hears zip tone through the headset and is then automatically connected to the call. (If the incoming trunk group is data restricted, the zip tone is not heard. If the agent's extension is data restricted, parties on the call hear the zip tone. A headset user should not be assigned data restriction.)

    Although not recommended, the automatic answering option can also be used with a handset or speakerphone. The feature works the same as with a headset, except the agent must be off-hook in order to receive the call. Zip tone, in this case, is heard through the handset or speakerphone.

   - Manual Answer—With Manual Answer, the agent hears ringing, and then goes off-hook to answer the incoming call.

- ACD Call Work Modes
   - Auxiliary Work—An agent can enter the Auxiliary Work mode when he or she is doing non-ACD activities such as taking a break or going to lunch. This makes the agent unavailable for ACD calls for that split.
   - After Call Work—An agent can enter the After Call Work (ACW) mode to perform ACD-related activities when needed. For example, an agent may need to fill out a form as a result of an ACD call. The agent can enter the ACW mode to fill out the form. The agent is unavailable for ACD calls from any split while in the ACW mode.
   - Auto-In or Manual-In—An agent can enter either the Auto-In mode or the Manual-In mode.

     When an agent enters the Auto-In mode, he or she, upon disconnecting from an ACD call, automatically becomes available for answering new ACD calls.

     When an agent enters the Manual-In mode, he or she, upon disconnecting from an ACD call, enters the After Call Work mode for that split, and is not available for ACD calls. The agent must then manually re-enter either the Auto-In mode or Manual-In mode to become available for ACD calls.

- Agent Request for Supervisor Assistance—Agents can request assistance (whether on an active ACD call or not) from the split supervisor by using the Assist button or the Assist feature access code.

- ACD Call Disconnecting—An agent can be disconnected from an ACD call in either of three ways. The agent can press a Release or Drop button, the call can be dropped by the calling party, or the agent without the automatic answering option can go on-hook.

Call Management System (CMS)

The CMS is an optional adjunct to the system that collects and processes ACD data. The CMS uses this data to generate various reports on the status of agents, measured splits, and measured trunks. These reports can be stored for later use or can be displayed on a terminal for real-time information.

ACD data is sent to the CMS under the following conditions:

- When an agent is moved from one split to another.
- When the system receives a request from the CMS to determine if translation information has changed.

- When the system receives a request from the CMS to update the full system configuration (trunk and agent configuration and agent log-in status).
- When the system receives a request from the CMS to update agent log-in status.
- When the system receives a request from the CMS to update split parameters.

The following agent status indications are provided to the CMS by the system:

- Agent Available—An agent, not active on any calls, is available to receive an ACD call.
- Agent Active—An agent has answered an ACD call and is still active on the call.
- Agent In After Call Work—An agent has entered the After Call Work (ACW) mode.
- Agent In Auxiliary Work—An agent has entered the Auxiliary Work (AUX) mode. This is also the indication given when an agent is active on an ACD call to another split.
- Agent On ACW Outgoing Call—An agent, while in the ACW mode, has placed an outgoing call.
- Agent On ACW Incoming Call—An agent, while in the ACW mode, has answered an incoming call.
- Agent On AUX Outgoing Call—An agent, while in the AUX or Available mode, has placed an outgoing call.
- Agent On AUX Incoming Call—An agent, while in the AUX or Available mode, has answered an incoming non-ACD call.

The following trunk status indications are provided to the CMS by the system:

- Trunk Idle.
- Trunk Seized.
- Trunk Queued On A Split—An ACD call is in queue for a split, is connected to a forced first announcement, or is ringing an agent's voice terminal.
- Trunk Dequeued And Connected—A trunk call has been answered by an ACD agent.
- Trunk Dequeued And Abandoned—The calling party, on a queued call, has dropped the call.
- Trunk Dequeued And Redirected—An incoming trunk call has been redirected via Interflow or Intraflow.
- Trunk Maintenance Busy—A trunk is out of service due to maintenance.
- Trunk Failure.

- Trunk Transferred To Another Agent—A trunk call is transferred to another agent.

In addition to the preceding information, the following trunk information is also sent to the CMS from the system:

- When a trunk call is redirected, the system lets the CMS know whether the call was redirected via intraflow to an extension or split (identified as split 0 for an unmeasured split or extension), intraflow to an attendant, or forwarded to another split.
- The trunk call destination is sent to the CMS. If the call is dequeued and connected, the agent identification is sent to the CMS as the destination. If the call is queued or redirected, the split number is sent to the CMS as the destination.
- The system also tells the CMS whether the trunk call is priority or nonpriority, and whether the call is incoming or outgoing.

In addition to being used for the display of ACD reports, the CMS terminal can be used to move agents from one split to another. For details of this function, see the Move Agents From CMS feature description elsewhere in this chapter.

Basic Call Management System

The Basic Call Management System (BCMS) feature provides real-time and historical reports which assist a customer in managing individual agents, ACD splits (hunt groups), and trunk groups. These reports, provided by the system, are a subset of those available on the CMS adjunct. BCMS reports can be accessed and displayed on the Manager I terminal or printed on demand on the printer associated with the Manager I terminal. In addition, the historical reports can be scheduled to print on the system printer. A detailed description of the BCMS feature is provided in the Basic Call Management System feature description elsewhere in this chapter.

Abandoned Call Search

The Abandoned Call Search feature is used to identify abandoned calls on ground start, CO, FX, and WATS trunks. When the calling party on an ACD call abandons (drops) the call while waiting to be connected to an agent, the call is not connected to the agent, and the call is reported to the CMS as being abandoned. For a detailed description, see the Abandoned Call Search feature elsewhere in this chapter.

Service Observing

Split supervisors (or other specified users with a Service Observe button) can use the Service Observing feature to train new agents and to observe in-progress ACD calls. While observing a call, the supervisor can toggle between a listen-only and a listen/talk connection to the call. An optional warning tone can be administered to let the agents know that someone is observing the call. For more details on this feature, see the Service Observing feature description elsewhere in this chapter (see Note).

Note: The use of service observing features may be subject to federal, state, or local laws, rules, or regulations and may be prohibited pursuant to the laws, rules, or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable law, rules, and regulations before using these features.

Considerations

ACD is particularly useful whenever a department or answering group receives a high volume of calls of the same type (for example, a catalog ordering department). Members of the department or answering group can be assigned to an ACD split. Call completion time is minimized and, since calls go directly to the split, attendant assistance is not required.

The maximum number of ACD splits is 32 (V3) or 99 (G1). As many as 100 (V3) or 200 (G1) agents can be assigned to a single split. The system maximum is 448 (V3) or 500 (G1) agents. (If an agent is assigned to more than one split, each split assignment applies toward the 448 [V3] or 500 [G1] system maximum.)

A maximum of 200 (V3) or 400 (G1) agents can be measured by the CMS.

The maximum number of ACD split supervisors is 32 (V3) or 99 (G1) (1 per split).

A voice terminal or individual attendant can be an agent in one or more splits.

A maximum number of 100 (V3) or 200 (G1) calls can be in queue for an ACD split. The system maximum is 1000 calls in queue.

Each system can contain up to 64 different recorded announcements (mixture of both analog and digital). Each split queue can be assigned two of these announcements as delay announcements. A delay announcement can be shared among the ACD splits. Callers are always connected at the beginning of the announcement.

Announcements may be either digital or analog. Digital announcements use the 16-port announcement board and queuing is based on whether or not one of the 16 ports is available. When a port becomes available, any of the announcements on the board can be accessed. Therefore, a caller may be in queue for an announcement (because a port is not available), even though that announcement is not being used. The maximum queue length for all digital announcements is 50. Queues for analog announcements are on a per-announcement basis and have a maximum queue length of 150. As many as five users may hear the same digital announcement at the same time. One caller at a time may listen to an announcement on an analog port.

If a delay announcement is used, answer supervision is sent to the distant office when the caller is connected to the announcement. Charging for the call, if applicable, begins when answer supervision is returned.

Calls incoming on a non-DID trunk group can route to an ACD split instead of to an attendant. Calls incoming on any non-DID trunk group can have only one primary destination; therefore, the trunk group must be dedicated to the ACD split.

Multi-appearance voice terminals can receive only one ACD call at a time. A voice terminal is available for an ACD call only if all call appearances are idle.

Leave Word Calling messages can be stored for an ACD split and can be retrieved by a member of the ACD split, a covering user of the split, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization must be assigned to the message retriever. Also, a remote Automatic Message Waiting lamp can be assigned to a split agent to provide a visual indication that a message has been stored for the split. One remote Automatic Message Waiting lamp is allowed per ACD split. The status lamp associated with this button informs the user that at least one message has been left for the split.

Each ACD split and each individual agent is assigned a Class of Restriction (COR). Miscellaneous Restrictions can be used to prohibit selected users from accessing certain splits. Either Miscellaneous Restrictions or restrictions assigned through the COR can be used to prohibit the agents from being accessed individually. Unless such restrictions are administered, each agent can be accessed individually as well as through the split.

CMS measurements may be inaccurate on calls to splits that intraflow to the attendant group.

Incoming calls directed to a split and then abandoned may cause erroneous ringing at agents' voice terminals.

If an agent becomes available while a caller is listening to an announcement (other than a forced first announcement), the call is removed from the announcement and is connected to the available agent.

If all agents are logged out or in the AUX work mode, incoming MEGACOM® telecommunications service calls receive a busy signal if no coverage path is provided.

When a CO call enters a full ACD split queue, there may be a difference in the switch measurement and the CMS measurement. This is because it is a CO call. The switch measurement will indicate the maximum number of calls allowed in the queue. The CMS measurement will indicate all the calls in the ACD split queue plus any call on the CO trunk waiting to terminate on the ACD split.

If an ACD split extension is assigned as the incoming destination of a trunk group, and that split's extension is later changed, the trunk group's incoming destination must also be changed to a valid extension.

Interactions

The following features interact with the Automatic Call Distribution feature.

- Attendant Call Waiting

  An attendant can originate or extend a call to an ACD split. Attendant Call Waiting cannot be used on such calls. However, such calls can enter the split queue, if provided.

- Automatic Callback

  Automatic Callback calls cannot be activated toward an ACD split.

- Call Coverage

  Calls can redirect to or from an ACD split.

  For a call to an ACD split to be redirected to Call Coverage, each voice terminal in the split must be active on at least one call appearance and the queue, if there is one, must be full. If the queue is not full, a call will enter the queue when no voice terminal is available. Queued calls remain in queue for a time interval equal to the Coverage Don't Answer Interval before redirecting to coverage. If any voice terminal in the split is idle, the call directs to that voice terminal.

  Calls can be redirected to another ACD split via Call Coverage to activate the Intraflow feature.

  If a call is queued for an ACD split and redirects via call coverage directly to an announcement, the call will be dropped upon completion of the announcement.

  When a call is redirected via Call Coverage to an ACD split, the calling party will not hear a forced first announcement, if administered. In order for the redirected call to receive an announcement, the announcements must be administered as first or second delay announcements.

- Call Forwarding All Calls

  When activated for an individual extension, the ACD functions of the individual extension are not affected.

  When activated for the split extension, calls directed to the split are forwarded away from the split. No announcements (other than a forced first announcement, if administered) associated with that split are connected to the call. The system reports to the CMS that the call is queued on the split and then reports to the CMS that the call has been dequeued and forwarded.

  Calls can be forwarded to an off-premises extension to activate the Interflow feature.

  On calls forwarded to an ACD split, the caller will hear the split's first delay announcement, if assigned.

- Data Call Setup (to or from a member of an ACD split)

  Voice Terminal Dialing or Data Terminal (Keyboard) Dialing can be used on calls to an ACD split.

- Data Restriction

  If the trunk group used for an ACD call has Data Restriction activated, agents with automatic answer actived will not hear the zip tone that is normally heard.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  Before the System Manager changes a hunt group from ACD to non-ACD (DDC or UCD), all agents in that hunt group must be logged out.

  When the System Manager changes a hunt group from ACD to non-ACD (DDC or UCD), all agents in that hunt group are placed in a "hunt group busy" state by the system software. If any voice terminals in the hunt group have an Aux-Work button, the lamp associated with that button will light. In order to become available for calls, the agent can press the Auxiliary Work button. Voice terminals without Auxiliary Work buttons can dial the Hunt Group Busy Deactivation feature access code followed by the hunt group number to be able to receive calls.

- Distributed Communications System (DCS)

  If a call to an ACD split is forwarded to a split at another DCS node, the caller will not hear the forced first announcement of the forwarded-to split.

  If an ACD split is in night service, with a split at another DCS node as the night service destination, a call to the first split will be connected to the first forced announcement of the split serving as the night service destination.

- Hold

  If an agent puts an ACD call on hold, no information is reported to the CMS. Therefore, the CMS considers the agent still active on the call.

- Individual Attendant Access

  Individual attendant extensions can be assigned to ACD splits. Unlike voice terminal users, individual attendants can answer ACD calls as long as there is an idle call appearance and no other ACD call is on the console.

- Multi-Appearance Preselection and Preference

  All assigned call appearances must be idle before an ACD call is directed to a voice terminal.

- Night Service—Hunt Group

  When Hunt Group Night Service is activated for a split and the night service destination is a hunt group, the caller will hear the first forced announcement, if administered. The call is then redirected to the night service destination hunt group. When an agent in the night service hunt group becomes available, the call goes to that agent. If all agents in the destination hunt group are busy, the caller will hear the following, if assigned: delayed first announcement, music-on-hold or silence, and a second announcement.

- Priority Calling

  A priority call directed to an ACD split is treated the same as a nonpriority call, except that the distinctive 3-burst ringing is heard.

  A call made to an ACD split from a user or trunk group with a COR that has priority queuing is inserted ahead of normal priority calls in the split queue. However, if the call intraflows to another split without priority queuing, it is queued as a normal priority call in the covering split's queue.

- Station Message Detail Recording (SMDR)

  When a Central Office (CO) call enters a full ACD split queue, SMDR and the CMS may show different measurements. SMDR measurements indicate the maximum number of calls allowed in the queue, whereas the CMS measurements indicate all calls in the queue plus any call on the CO trunk waiting to enter the split queue.

- Terminating Extension Group

  A Terminating Extension Group cannot be a member of an ACD split.

- Voice Terminal Display.

  On calls dialed directly to an ACD split extension number, the calling party's identity (trunk name or user name) and the ACD split's identity (split name) is displayed at the called extension.

Administration

ACD is administered by the System Manager. The following items can be administered for each ACD split (hunt group):

- Split extension number, name, and type of hunting. The type of hunting is administered as either DDC (direct) or UCD (most-idle agent).
- Whether or not it is an ACD split. If not, the hunt group is a DDC or UCD hunt group.
- Whether or not changes in split status and parameters are sent to CMS. ACD splits to be measured by CMS must be numbered sequentially, beginning with Group 1.

- Whether or not the split is adjunct-controlled (G1).
- First announcement extension.
- First announcement delay interval.
- Second delay announcement extension.
- Second delay announcement interval.
- Whether or not the second delay announcement is recurring.
- Night service destination.
- Whether or not calls redirected by Intraflow have priority over other calls.
- Inflow threshold (0 to 999 seconds). If the oldest call in queue has been in queue for this amount of time, the split will not accept any redirected calls.
- Split supervisor extension.
- Split coverage path.
- Class of Restriction.
- 4-Digit security code (for use with AP Demand Print feature).
- Type of Message Center the split serves as (AP, AUDIX, or blank).
- Whether or not the split is served by a queue.
- Queue length (1 to 100 calls).
- Queue Warning Threshold for number of calls (1 to 100 calls).
- Queue Warning Threshold for time in queue of oldest call (0 to 999 seconds).
- Port Number assigned to auxiliary queue warning lamp (based on number of calls).
- Port Number assigned to auxiliary queue warning lamp (based on time in queue of oldest call).
- Group Members (extension numbers).

Hardware and Software Requirements

Each auxiliary queue warning level lamp requires one port on a TN742 Analog Line circuit pack. A 21C-49 indicator lamp may be used as a queue warning level lamp. This lamp is approximately 2 inches in diameter and has a clear beehive lens. The lamp operates on ringing voltage and can be mounted at a location convenient to the split.

Each delay announcement requires one port on a TN750 Integrated Announcement circuit pack or announcement equipment and one port on a TN742 Analog Line circuit pack. The four analog announcements should be assigned on the TN742 ports since the TN742 can only ring four ports at a time. If music is to be heard after an announcement, a music source and a port on a TN763 Auxiliary Trunk circuit pack is required. Music sources are not provided by the system.

If a CMS is to be used, CMS hardware is required.

ACD software is required. If a CMS is to be used, CMS software is required. If the BCMS feature is to be used, BCMS software is required.

Automatic Circuit Assurance (V2, V3, or G1)

Description

Assists users in identifying possible trunk malfunctions. The system maintains a record of the performance of individual trunks relative to short and long holding time calls. The system automatically initiates a referral call to an attendant or display-equipped voice terminal user when a possible failure is detected.

Holding time is the elapsed time from the time a trunk is accessed to the time a trunk is released. When the Automatic Circuit Assurance (ACA) feature is enabled by the System Manager, the system measures the holding time of each call.

A short holding time limit and a long holding time limit are preset by the System Manager for each trunk group. The short holding time limit can be from 0 to 160 seconds. The long holding time limit can be from 0 to 10 hours. The measured holding time for each call is compared to the preset limits for the trunk group being used.

A short holding time counter and a long holding time counter associated with each trunk group member are kept by the system. When the measured holding time of a call is compared to the preset limits, these counters are incremented or decremented as follows:

- Measured holding time less than short holding time limit—Short holding time counter is incremented.
- Measured holding time greater than short holding time limit and less than long holding time limit—Short holding time counter is decremented.
- Measured holding time greater than long holding time limit—Long holding time counter is incremented.

The short holding time counter is constantly compared to a preset threshold. This threshold can be from 0 to 30 and is set by the System Manager. The threshold for the long holding time counter is always 1. Each time a counter reaches a preset threshold, two things occur as soon as the system clock reaches the next hour or the call is dropped:

- If ACA referral has been activated by an attendant or voice terminal user, a referral call is sent by the system to a designated attendant console or display-equipped voice terminal.
- An entry is made in an audit trail which stores information on the occurrence.

When ACA is enabled by the System Manager, the ACA measurements are made and the audit trail is updated each time a preset counter threshold is reached. However, in order for a referral call to be sent, ACA referral must be activated. ACA referral is activated whenever an attendant or user presses an ACA button. When this is done, the system can send referral calls to the destination specified by the System Manager.

The referral call destination can be the attendant console group, a specific attendant console, a display-equipped voice terminal, or, if Voice Message Retrieval is provided, a non-display voice terminal. The information appearing on the display identifies the call as an ACA call, identifies the trunk group access code and the trunk group member number, and shows the reason for referral (short or long holding time). When the call is answered, this information is displayed and remains displayed until the call is released.

Each time a counter threshold is reached, a record of the information is stored in the audit trail. The audit trail records are available to the System Manager. Each record contains the following information:

- Time and Date of occurrence
- Trunk group number, trunk access code, and trunk group member
- Type of referral (short or long holding time).

If the referral call destination does not answer the call within 3 minutes, the call times out and this information is entered in the audit trail. The audit trail is examined once each hour. If any entries indicate a referral call was not completed, the call is tried again.

ACA can be enabled or disabled for the entire system by the System Manager. The System Manager can also enable or disable ACA for each individual trunk group. When ACA is disabled, ACA measurements are not made.

Two extensions must be assigned for the purpose of letting the referral call destination identify the type of ACA call (short or long holding time). The two extensions are assigned as a short holding time origination extension and a long holding time origination extension. These extension numbers do not require hardware circuit packs.

As an illustration of how ACA functions, assume the following:

- The ACA is enabled for the entire system.
- The ACA referral destination is extension 389.
- The ACA long holding time origination extension is 423.
- The long holding time limit for trunk group 3 (trunk access code is 9) is 1 hour.
- The ACA referral is activated.

With the above information, assume that a call is made on a trunk in trunk group 3 and the call lasts more than 1 hour. Then, the threshold for the long holding time counter is reached, a referral call is made to extension 389, the display reflects a long holding time call, and the information is entered in the audit trail. The referral destination can then have the operation of the trunk checked and taken out of service if defective.

Considerations

The ACA feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time. Some types of trunk failures cause people to shorten their calls. For example, an excessive number of short calls may indicate a noisy trunk. Similarly, a trunk that remains busy for an abnormally long time may be permanently busy due to a trunk fault. The ACA feature takes advantage of these characteristics to identify possibly defective trunks. Once the trunk has been identified as possibly being defective, the Busy Verification of Terminals and Trunks (V2, V3, or G1) feature can be used to check the trunk.

The audit trail contains a maximum of 64 records at any one time. The oldest information is overwritten by the newest information.

Measurements are not made on personal central office lines, out-of-service trunks, or trunks undergoing maintenance testing.

Interactions

The following features interact with the Automatic Circuit Assurance feature.

- Centralized Attendant Service (CAS)

  When CAS is activated, the referral call destination must be on the local switch. A referral destination of "0" is interpreted as the local attendant, if one exists.

  The CAS attendant cannot activate or deactivate ACA referral calls at a branch location.

- Night Service—Night Station Service

  Referral calls will not be placed if the system is in the Night Service mode.

Administration

ACA is administered by the System Manager. The following items require administration:

- Whether ACA is enabled or disabled (per system).
- Short holding time origination extension (per system). Assigned name must reflect short holding time nomenclature.
- Long holding time origination extension (per system). Assigned name must reflect long holding time nomenclature.
- Referral destination (per system).
- Whether ACA is assigned (per trunk group).
- Short holding time limit (per trunk group).
- Long holding time limit (per trunk group).
- Threshold for short holding time counter (per trunk group).

- ACA activate/deactivate button on attendant console or voice terminal (one per system).

Hardware and Software Requirements

A TN725 Speech Synthesizer circuit pack is required if the referral destination is not a display-equipped voice terminal.

No additional software is required.

Automatic Incoming Call Display (V2, V3, or G1)

Description

Provides display-equipped voice terminal users, who are already active on a call, with the identity of a second or subsequent caller. The identity is displayed on the terminal's alphanumeric display.

The alphanumeric display can be either of the following:

- The digital display module associated with a 7405D voice terminal
- A 515 Business Communications Terminal (BCT)
- Line 1 of the display on a 7407D voice terminal
- Line 1 of the display on a 7406D voice terminal
- A data terminal connected to a 7404D voice terminal with an optional messaging cartridge.
- The display on a CALLMASTER™ digital voice terminal.

This feature applies when an incoming call terminates at a user's voice terminal while the user is active on another call appearance. The information displayed on the current call is replaced by the identity of the incoming call. The identity of the incoming call normally remains displayed for 30 seconds unless there is another incoming call, the user hangs up, or the calling party hangs up. After 30 seconds, the display returns to the current call information. With the "CALLMASTER" digital voice terminal, the display goes blank after 30 seconds.

A third or subsequent incoming call overwrites the information displayed on the previous call and restarts the 30-second interval. In any case, the most recent call to terminate at the user's voice terminal is the call identified by the display.

If the party whose identity is currently being displayed hangs up, the display returns to the current call information. If the user hangs up on the current call before the 30-second interval expires on the incoming call, the display is cleared.

The information displayed on the current call is not replaced by the identity of the incoming call if the called user is in the process of dialing the current call or if the Outgoing Display Option is not administered to the trunk group being used.

Considerations

The Automatic Incoming Call Display feature lets certain users, while active on one call, know the identity of another incoming caller. This is done without the use of an Inspect button. By knowing who is calling, the user can handle the calls accordingly.

The incoming call must terminate at the user's voice terminal in order to be displayed. Calls forwarded to another extension are not displayed.

The 7406D, the digital display module of the 7405D, and the 515 BCT must be in the normal mode to display the identity of the incoming call. This is not required of the 7407D.

If a user with the outgoing display option off is dialing or active on an outgoing trunk call, the Automatic Incoming Call Display does not override the display.

Interactions

This feature enhances the Voice Terminal Display feature by providing automatic identification of incoming calls. The same incoming call information can be provided by putting the display in the inspect mode; however, this is not automatic and must be done manually for each call.

Administration

None required.

Hardware and Software Requirements

Requires a 515 BCT, a display-equipped voice terminal, or a voice terminal capable of displaying information through an attached data terminal. No additional software is required.

Automatic Route Selection (V1)

Description

Routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

Routing is controlled by as many as 16 routing patterns. A routing pattern is an ordered list of the routes (trunk groups plus a Facility Restriction Level [FRL], described later) the system can use to complete a particular call. Each pattern can contain up to six trunk groups arranged in the order of preference. Usually the first-choice trunk group is the least expensive.

An Automatic Route Selection (ARS) pattern may contain any or all of the following types of trunk groups for routing a call:

- Wide Area Telecommunications Service (WATS)
- Foreign exchange (FX)
- Central office (CO)
- Tie trunks.

Each trunk group assigned to an ARS pattern has an associated FRL number from 0 to 7. The same trunk group can be used in more than one pattern and can have a different FRL in each pattern. Each voice terminal user has an FRL assigned via the Class of Restriction (COR) form. A voice terminal assigned an FRL of 0 has the least privileges; an FRL of 7 has the most privileges. A trunk group with an FRL of 0 is least restricted; an FRL of 7 is most restricted.

ARS selects patterns for the following types of calls:

- Calls made to COs within the Home Numbering Plan Area (HNPA) Code. The HNPA is commonly called the Home or local Area Code.
- Calls made to Foreign Numbering Plan Area (FNPA) Codes. FNPA Codes are simply Area Codes other than the local Area Code.
- Calls made to COs located within a particular FNPA. Calls to these COs are referred to as Remote Home Numbering Plan Area (RHNPA) calls. Pattern selection is based on a CO code rather than on the Area Code alone.
- Special service calls such as 911.
- International calls.

The system selects a Routing Pattern from one of the six tables stored in memory. The six tables are: one HNPA, one FNPA, and four RHNPAs. The type of call determines which table a call enters.

A call within the Home Area Code enters the HNPA table. The HNPA table contains an ARS pattern number for each individual office code within the Home Area Code. When a particular office code within the Home Area Code is called, the corresponding Routing Pattern is selected.

Calls to area codes outside the Home Area Codes and special service codes such as 911 enter the FNPA table. The FNPA table contains either an ARS pattern number or an RHNPA table number for each of the 160 possible 3-digit area codes and service codes that can be accessed by the system.

When an area code outside the Home Area Code is called and the FNPA table contains an ARS pattern number for that area code, the call follows the assigned Routing Pattern.

When an area code outside the Home Area Code is called and the FNPA table contains an RHNPA table number for that area code, the call enters the assigned RHNPA table. Each RHNPA table contains an ARS pattern number for each individual office code within the associated area code. When a call enters an RHNPA table, a Routing Pattern is selected according to the called office code within that RHNPA table.

If a call begins with 0, it does not enter any of the six tables stored in memory but, instead, enters the prefix table. All calls beginning with the following 0-type codes use the same assigned Routing Pattern:

- 0 indicates operator access
- 0+ indicates operator assisted calls
- 01+ indicates international operator
- 011+ indicates international direct.

All calls beginning with 10 use the same Routing Pattern as assigned in the prefix table. The FRL denies or allows access to a trunk group in a pattern. When a user places a call, the system selects a Routing Pattern based on the first digit dialed (0-type call), first three digits dialed (office or area code), or the first six digits dialed (area and office code). The route pattern is selected from one of the six translation tables stored in memory.

Once a pattern is selected for the type of call being placed, the system attempts to select a trunk group to handle the call. The trunk groups in the ARS pattern are listed in the order of ascending FRLs. The trunk group with the lowest FRL number is checked first. The system compares the FRL associated with the caller to the trunk group FRL. The system will allow the caller to access the trunk group if the caller FRL is equal to or greater than the FRL of the trunk group.

Once the call satisfies the FRL compatibility requirements, the system automatically checks for an available trunk within the selected trunk group. If there is an idle trunk, the call is placed. If all trunk groups are busy, the call can queue on the first choice trunk group if queuing is provided for this trunk group. If the FRL requirements are met on a call, but no trunk group or queuing is available, reorder tone is returned to the caller. If the originating FRL does not authorize access to any trunk groups in the pattern, intercept tone is sent back to the caller. In this case, the caller should not try the call again, because the call will always be denied regardless of how many trunks in the group are idle.

The following tones are associated with ARS:

- Confirmation—Indicates that the call has queued.
- Busy—Indicates that the called number is busy.
- Reorder (fast busy)—Indicates that all trunks are busy.
- Intercept—Indicates that the originating FRL is not sufficient to allow the call.

Considerations

With ARS, voice terminal users do not have to worry about accessing a particular trunk group to make a long-distance call. The user simply activates ARS and dials the desired number. The system then routes the call to the outgoing trunk group best suited for that call.

A system can have up to 16 ARS Routing Patterns. Each pattern can contain up to six trunk groups.

There is one HNPA table, one FNPA table, and four RHNPA tables per system. The HNPA table and the four RHNPA tables can each contain up to 800 office codes. The FNPA table can contain up to 160 Area Codes.

If a customer chooses a single primary long-distance carrier for all long-distance (1+) calls, then any International Direct Distance Dialing (IDDD) (011), operator (0), Customer-Dialed and Operator-Serviced (CDOS) (0+ or 01+), 700, and 900 calls also go to that carrier. In order to place a call to an area not served by the primary long-distance carrier, the appropriate 10xxx code must be dialed to access a different carrier that has access to the desired area.

If a customer changes ARS routing assignments, it is the customer's responsibility to notify the Regional Support Center (RSC) network designer and the System Control Office (SCO) technician of the changes in order to receive their continued support.

In certain areas, directory assistance is 1411 instead of just 411. In such cases, the 1 must be dialed before 411. The system cannot be administered to insert the 1.

Interactions

The following features interact with ARS:

- Abbreviated Dialing

  The ARS access code may be stored in an Abbreviated Dialing Group or System List. If the group or system list is privileged, the caller's Class of Restriction (which contains the FRL) is never checked and any number in the list will be processed.

- Attendant Console

  If the attendant dials an ARS code for an outgoing call for voice terminal user, the system checks the attendant FRL to determine if the call can be made.

- Attendant Control of Trunk Group Access

  Activation of this feature removes the trunk group from ARS patterns, and deactivation reinserts the trunk group into the Routing Pattern. ARS calls do not redirect to the attendant.

- Controlled Restriction

  All ARS calls are denied when either Controlled Outward Restriction or Controlled Total Restriction has been activated for the calling extension. Controlled Termination Restriction is not checked on the outgoing trunk for ARS calls.

- Miscellaneous Trunk Restriction

  This feature, if provided, does not apply to ARS calls. The route FRL is the controlling factor.

- Origination and Outward Restrictions

  These restrictions prohibit access to ARS.

- Personal Central Office Line Group (PCOLG)

  A PCOLG cannot be assigned to an ARS pattern.

- Remote Access

  The FRL of the incoming trunk group or the Remote Access barrier code, if used on the call, serves as the originating FRL. This FRL is contained in the Class of Restriction assigned to the trunk group or barrier code.

- Ringback Queuing

  When all accessible trunk groups in a Routing Pattern are busy, the call will queue on the most-preferred trunk group (if queuing is provided and the queue is not full). Queuing is automatic for single-line voice terminals; however, multi-appearance voice terminal users must press an Automatic Callback button to activate the Ringback Queuing feature.

- Station Message Detail Recording (SMDR)

  If SMDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group. If an SMDR account code is to be dialed with an ARS call, it must be dialed before the ARS access code is dialed.

  An ARS call is identified by the access code dialed and by a Condition Code. The dialed number is recorded as the called number.

Administration

ARS is administered on a per-system basis by the System Manager. The following items must be established, defined, and administered before ARS can be activated.

- Facility Restriction Levels (FRLs)

  Eight FRLs, numbered 0 through 7, allow and deny access to facilities. An FRL is assigned to each trunk group route within each Routing Pattern. A route assigned an FRL of 0 is the least restricted, a route assigned an FRL of 7 is the most restricted.

  Originating FRLs are assigned by Class of Restriction to a voice terminal, an incoming tie trunk group, a remote access trunk group, and the attendant group. An originating FRL of 0 has the least calling privileges, an originating FRL of 7 has the most calling privileges.

- Routing Patterns

  As many as 16 Routing Patterns, containing up to 6 routes each, can be used to control routing. Routing Patterns are numbered from 1 to 16.

  Patterns are selected from the HNPA, FNPA, RHNPA, or prefix tables. Patterns are administered on the FNPA, HNPA, or RHNPA, or Prefix forms.

  Patterns 2, 3, 4, and 5 are default patterns for the RHNPA Tables 1, 2, 3, and 4, respectively. Pattern 1 is the default for the HNPA Table entry.

  The same trunk group can be assigned to several Routing Patterns. The FRL assigned to the individual route controls whether a call is allowed or denied.

  The FRLs associated with each trunk group must be assigned in order of preference on the ARS pattern form. This keeps the system from searching the entire pattern to find the most-preferred route.

- Home Numbering Plan Area (HNPA) and Foreign Numbering Plan Area (FNPA) tables

  The HNPA table defines up to 800 local CO codes, numbered from 200 through 999. A corresponding Routing Pattern number is also assigned.

The FNPA table defines up to 160 foreign Area Codes and service codes. A corresponding Routing Pattern number or a reference to a Remote Home Numbering Plan Area (RHNPA) table also is assigned. To administer an RHNPA for a particular FNPA, the System Manager must enter r1, r2, r3, or r4 instead of the ARS Pattern Number.

- RHNPA Tables

  Up to four RHNPA tables can be established. These tables are normally reserved for FX trunks or for 6-digit translations. Each table can include the 800 possible CO codes, numbered from 200 through 999, and a corresponding Routing Pattern number. These tables will be associated with an area code referenced from the FNPA table. A pattern number must be entered in the RHNPA if an RHNPA table is used. The system defaults Tables 1, 2, 3, and 4 to ARS patterns 2, 3, 4 and 5, respectively.

- ARS Access Code

  The ARS access code is used to gain access to the ARS features. The access code is assigned on the Feature Access Code form. Only one code can be assigned. This code must be dialed before the 7- or 10-digit number can be dialed.

- Tie Trunk

  Tie trunks assigned as Advanced Private Line Termination (APLT) can be assigned as a trunk group in an ARS pattern. For calls using an APLT trunk group, ARS inserts the HNPA, if not dialed, since private networks require ten digits on all calls destined for the public network.

- Toll Tables

  Up to four Toll Tables can be established. Toll tables are needed when an FX trunk group terminates at a step-by-step CO and requires the digit 1 on all toll calls.

  Toll Tables are numbered from 1 through 4. The number of the Toll Table must be administered to the associated trunk group. They are assigned to CO, FX, and WATS trunk groups.

- Prefix Table

  This table specifies the Routing Pattern number used for calls beginning with 0 or 10. This includes operator-assisted calls, directly dialed international calls, and long-distance carrier calls. The ARS pattern used for these calls is assigned on the ARS Prefix Codes form.

The ARS default patterns are as follows:

- Pattern 0 is the fixed intercept pattern. It is included in the ARS data existing in the system at all times. Intercepted ARS calls are routed to intercept tone. The intercept pattern is the automatic default pattern for all 160 entries of the FNPA table until changed for a specific system through the System Access Terminal (SAT). Intercept, Pattern 0, can also be the destination for any entry of the other NPA tables.

- Pattern 1 is the default pattern for the 800 office codes that make up the HNPA table. Therefore, when defining patterns, Pattern 1 should contain only local CO trunks to provide for completion of local calling area office codes. If less expensive facilities (that is, FX or WATS) have been provided for HNPA long-distance calls, another pattern(s) should be created and that (those) pattern number(s) should replace the Pattern 1 route associated with those office codes in the HNPA table. This is done using the series of ARS change commands at the SAT.

  Any nonworking home area office codes should remain routed to Pattern 1 (local CO trunks) so that they may be intercepted in the CO. This application eliminates the need to continually monitor and update working and nonworking office codes. When the nonworking code is activated in the CO, calls will automatically complete. Again, this arrangement can be changed using the SAT.

- Patterns 2, 3, 4, and 5 exist in ARS software as default patterns for RHNPA tables r1, r2, r3, and r4, respectively. If "r" (that is, RHNPA) tables are not used, these default patterns are also unused, and the pattern numbers (2, 3, 4, and 5) can be used to define any route. It is recommended, however, that these pattern numbers be held in reserve for reasons explained in the following paragraphs.

Primarily, the RHNPA tables exist to support the four possible FX trunk groups that can be included in an ARS configuration. Like the HNPA table, and RHNPA table contains 800 possible office codes including service codes (such as 411 and 911) associated with the CO where the FX trunks terminate. The RHNPA tables are administered and translated in the same manner as the HNPA table.

It should be noted, however, that FX trunk groups may have the same NPA as the local CO trunk group that serves the system. In this case, the HNPA table will serve the FX trunk group and an additional pattern(s) should be created to designate the FX trunk group as the first choice. This additional pattern(s) should then be assigned to the affected HNPA office codes to provide the least-expensive route for long-distance calling.

When the FX trunk group terminates in an area code (NPA) other than the HNPA, calls to office codes that are long distance to the terminating office should be routed through the least-expensive route. Service codes should be routed locally through the FX trunk group. Like the HNPA table, nonworking CO codes should be arranged so that they are intercepted in the CO.

When FX trunks are not provided, all "r" tables are available for 6-digit translation applications. Only one area code may be associated with an "r" table and, once initialized, routes for each of the possible 800 office codes must be considered. That is, the default pattern entry must be changed to reflect the appropriate pattern.

Hardware and Software Requirements

No additional hardware is required. ARS software is required.

Automatic Route Selection (V2, V3, or G1)

Description

Routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

Automatic Route Selection (ARS) provides a choice of up to six routes for any given public network call. The following types of trunk groups can be accessed by ARS:

- Local central office (CO)—Used for local calls and to provide access to a long-distance carrier. Access to the long-distance carrier can be provided either automatically by the CO or by a carrier access code.
- Foreign exchange (FX)—Used to emulate local calling in an area not served by the local CO. Like the local CO, the FX office provides a choice of long-distance carriers.
- Wide Area Telecommunications Service (WATS)—Used to provide calling to predefined geographic areas at a rate based on expected usage.
- Tie trunks—Used to provide access to an Electronic Tandem Network (ETN), or to an Enhanced Private Switched Communications Service (EPSCS) or Common Control Switching Arrangement (CCSA) office. (In some cases, it is preferable to allow a private network to handle the routing of calls destined for the public network.)

ARS is particularly useful when one or more long-distance carriers and WATS are provided. The system selects the most-preferred (normally, least-expensive) route for the call. Long-distance carrier code dialing is not required on routes selected by the system. Long-distance carrier codes are set in translations to best benefit the customer on any given call. These codes are inserted as needed to guarantee automatic carrier selection.

Dial access to a long-distance carrier's operator is also provided. Carrier access codes are of the form 10xxx, where xxx are digits to identify a particular carrier. If 10xxx plus a 0-type (operator or operator-assisted) number is dialed, the call routes to the long-distance carrier's operator.

The system may serve as an ETN tandem switch. In this case the system can access or be accessed by Intertandem Tie Trunks to/from other tandem switches and/or Access Tie Trunks to/from ETN main switches. The system can also access Bypass Tie Trunks to an ETN main switch. This distinction as a tandem switch is important with respect to the routing of certain calls.

Dialing

The ARS access code is normally the digit 9. With V3 or G1, two different ARS access codes can be assigned. The called number may be a service code, a 7- or 10-digit public network number, an International Direct Distance Dialing (IDDD) number, an operator code (0), a customer-dialed and operator-serviced (CDOS) number ("0+" or "01+"), or a long-distance carrier's operator (10xxx + 0+). Dialing 10xxx plus a 7- or 10-digit number is also possible, although route selection is based on the 7- or 10-digit number, not on the 10xxx.

To use ARS, the user simply dials the ARS access code (two different codes can be assigned with V3, or G1) and the called number. Users at subtending switches access the system, then follow the same dialing procedures as a user at the system.

Domestic Call Routing

The domestic calling area is divided into areas called numbering plan areas, or NPAs. Each NPA is identified by an NPA code, normally just called an Area Code. There are 160 such codes: 1 for the local (or home) numbering plan area (HNPA), and 159 for the other (foreign) numbering plan areas (FNPAs). Within a given NPA, all office codes (NXXs) are unique.

With ARS, call routing is determined by the first three or six digits of the called public network number, in other words, by the NPA or the office code or by both the NPA and the office code. Two 3-digit translators are provided: one for the office codes within the home NPA and one for the foreign NPAs. Thirty-two 6-digit translators are provided, allowing call routing based on the office codes within the foreign NPA rather than on the NPA alone. (The 6-digit translators are actually 3-digit translators that are accessed from the foreign NPA translator. The foreign NPA translator can yield one of the 6-digit translators based on the NPA. The 6-digit translator only translates the office code, which is the second three digits of the called number. At this time, six digits have been translated. Thus, for clarity, it is common to refer to these translators as 6-digit translators. These translators are also known as Remote Home Numbering Plan Area [RHNPA] tables since the call routes on an office code the same as a call within the home NPA does.)

Digit translation yields one of 254 Routing Patterns, numbered 1 through 254. More than one translator can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with ARS, digit translation should always point to a Routing Pattern. This way, calls to unassigned office codes will be intercepted by the CO, not by the system. By allowing the unassigned codes to be intercepted by the CO, the System Manager does not have to keep track of which office codes are in service. If calls to some codes are to be denied, this should be handled by Facility Restriction Level (FRL) assignment, not by intercept on the codes. FRLs are described elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of the trunk groups that can be used for the call. Trunk group access is controlled by FRLs. If access to the public network is through a main switch (an Access trunk group is selected for the call), then the call will route through the main to one of the public network offices serving the main. The digit manipulation necessary to route the call is controlled by the Subnet Trunking function. (Subnet Trunking is described elsewhere in this chapter.) Otherwise, the digit string to be outpulsed is controlled by ARS. ARS digit manipulation is called code conversion. Code conversion is used to determine whether or not to outpulse the digit 1 on toll calls and whether to insert, keep, or delete the NPA on toll calls. Whether or not the digit 1 should be dialed on an ARS call is a completely separate subject and has nothing to do with outpulsing a 1. Each of these items is described separately in the following paragraphs.

Digit 1 Dialing

Normally, the prefix digit 1 is not dialed on a 7- or 10-digit call routed by ARS.

However, there are two cases where the digit 1 must be dialed. Some metropolitan areas are so densely populated that there simply are not enough traditional office codes, that is, those that do not conflict with NPAs. In areas where NPA codes also serve as office codes, the digit 1 must be dialed if a toll (NPA) call is intended. The digit tells the system whether to route the call as a 7-digit call via the home NPA translator (1 not dialed) or as a 10-digit call via the foreign NPA translator (1 dialed). Digit 1 dialing may also be required in areas near an NPA boundary. In these areas, certain calls to the adjacent NPA may be local calls rather than toll calls. However, office codes are duplicated in the home and adjacent NPAs. Thus, if the digit 1 is not required on certain adjacent NPA calls, then it must be dialed on the home NPA calls so the system can differentiate between the intended destinations.

Digit 1 Outpulsing

The digit 1 may or may not be required at the public network office to which the call will be routing. (If "1" is dialed on 7-digit calls at a stand-alone system [non-ETN], the "1" is outpulsed by the system.) In the other cases, the "1" requirements are indicated in the system. Since any given call may have a choice of up to six routes, some of which may require a 1 and some of which may not, this indication is associated with each individual route. Four choices are available and are identified in translations by a Prefix Mark. Values and their meaning are as follows:

- Prefix Mark 0—Never send a 1.
- Prefix Mark 1—Send a 1 on 10-digit calls, but not on 7-digit calls.
- Prefix Mark 2—Send a 1 on all toll calls.
- Prefix Mark 3—Send a 1 on all toll calls and keep or insert the NPA to insure that all toll calls are 10-digit calls.

Which of the four possible treatments of the 1 prefix digit is applicable on a given route is based on the characteristics of the distant office. Prefix Mark 0 is straight forward. The system never sends a 1 prefix digit. Prefix Mark 1 causes the system to send a 1 prefix on all 10-digit calls.

With Prefix Marks 2 and 3, the decision is based on whether the call is a toll call. Toll Lists are provided in the system to furnish this information. A Toll List simply indicates if the office code associated with the call constitutes a toll call from the interconnecting office (not from the local system). Up to 32 Toll Lists are provided. The applicable list number, if any, for the call is given in the Routing Pattern.

Prefix Marks are only applicable on public network routes. No Prefix Mark is given on tie trunks. Also, the digit 1 is never transmitted over an intertandem tie trunk, even if dialed by the calling party. Requirements for a 1 are specified via Prefix Marks when the call accesses the public network. Thus, a 1 is never needed on an intertandem tie trunk.

NPA Deletion and Insertion

Each public network route in the ARS Routing Pattern contains an indication of the NPA of the distant end of the trunk group. If this NPA is the same as the NPA associated with the call, the NPA is deleted prior to outpulsing unless the Prefix Mark is 3 and the call is a toll call in the associated Toll List, or the trunk group is an integrated Services Digital Network—Primary Rate Interface (ISDN-PRI) trunk group. ISDN-PRI preferences attempt to form a 10-digit number, so the NPA will not be deleted.

The NPA is inserted on 7-digit calls if the distant NPA is different from the home NPA or if the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

The preceding paragraphs describe NPA deletion or insertion when the call is accessing the public network (and exiting an ETN, if an ETN provided partial routing of the call). When an ARS call accesses an ETN is one case of NPA insertion. If the call is to a destination within the home NPA and if the calling party did not dial the NPA, the system inserts the home NPA before sending the call to the ETN switch. Therefore, all ARS calls (to a domestic destination) accessing an ETN are 10-digit calls. This enables the system to distinguish between ARS calls and the 7-digit on-network calls.

IDDD and Service Code Dialing

IDDD calls other than those generated by Subnet Trunking need not be modified before outpulsing. Since international numbers can be of variable length, the system awaits a dialing time-out before processing the call. (Dialing time-out is 3 seconds for the 0 and 1 prefix digits, but is 10 seconds for the called number.) The calling party can speed up call processing by dialing the end-of-dialing digit (#) after the called number. Receipt of this digit cancels the remaining time-out interval. The system always outpulses the # digit for use by the distant switch, whether dialed by the calling party or not.

Subnet Trunking is not required for service codes. If the prefix digit 1 is dialed before the code, it is outpulsed.

ARS can provide individual Routing Patterns for each 0-type call. A 0-type call can be processed via the foreign NPA translator, meaning that 6-digit translation can be used. This is particularly useful on international calls, since the 6-digit translation can be used on the country code. Thus, call routing can be determined according to the called country, rather than handling all international calls alike.

Operator and Operator-Assisted Calls

Calls to an operator (0 by itself) require a 3-second time-out or dialing of the # digit before the call is processed. Operator-assisted calls (0 plus a 7- or 10-digit number) require 10-digit dialing if the call is within a home NPA and there are office codes within the HNPA which look like NPAs. (On directly-dialed calls, this distinction was made by prefix digit 1 dialing.) All other dialing is the same as direct dialing.

Operator-assisted calls, like IDDD calls, can be routed on the first three digits of the called number. Through the use of Subnet Trunking, this means that different long-distance carriers can be selected for different calls.

Directory Assistance Calls

Local Directory Assistance Calls always route to a telephone company operator. Long-distance Directory Assistance Calls may be routed to the long-distance carrier operator via an associated 6-digit RHNPA table.

Tones

The following tones are associated with ARS:

- Busy—Indicates that the called number is busy.
- Confirmation—Indicates that the call has queued.
- Intercept—Indicates that the originating FRL is not sufficient to allow the call.
- Reorder (fast busy)—Indicates that the call cannot be completed at this time because at least one required facility is not available. (Multi-appearance voice terminal users may be able to queue the call.)

Special Call Routing

The system recognizes certain types of dialing patterns on outgoing calls and routes these calls via special entries in the HNPA or FNPA table. The rules for ARS routing are given in Table 3-A which lists the special dialing patterns along with the associated HNPA or FNPA table entry through which that type of call is routed.

Table 3-A. ARS Routing Table

Call Type	Digits Dialed	Routes On Pattern Assigned FOR	Translator Table
OPERATOR	0	000	FNPA
TOLL OPERATOR	00	002	FNPA
INTERNATIONAL (SPECIAL ACCESS)	00I	004	FNPA
INTERNATIONAL OPERATOR	010	010	FNPA
INTERNATIONAL-DIRECT DIAL	011X...X	011	FNPA
TOLL OPERATOR-DIRECT DIAL	00NX...X	003	FNPA
INTERNATIONAL-OPERATOR ASSIST	01NX...X	012	FNPA
OPERATOR ASSIST	0NX...X	001	FNPA
LONG DISTANCE SERVICE	(1)N11	N11	FNPA
LONG DISTANCE IN NPA	(1)NXX-XXXX	NXX	HNPA
LONG DISTANCE-TOLL FREE	(1)800-NXX-XXXX	800	FNPA
LONG DISTANCE-DIRECTORY ASSIST	(1)NIX-555-XXXX	005	FNPA
LONG DISTANCE IN HOME NPA	(1)HNPA-NXX-XXXX	NXX	HNPA
LONG DISTANCE OUT SIDE OF NPA	(1)NIX-NXX-XXXX	NIX	FNPA
LDC-ACCESS CODE	10XXX	119	FNPA
LDC-OPERATOR	10XXX-0	100	FNPA
LDC-TOLL OPERATOR	10XXX-00	102	FNPA
LDC-INTERNATIONAL OPERATOR	10XXX-010	110	FNPA
LDC-INTERNATIONAL DIRECT DIAL	10XXX-011X...X	111	FNPA
LDC-TOLL OPERATOR-DIRECT DIAL	10XXX-00NX...X	103	FNPA
LDC-INTERNATiONAL OPERATOR DIRECT DIAL	10XXX-01NX...X	112	FNPA
LDC-OPERATOR ASSIST	10XXX-0NX...X	101	FNPA
LDC-DIRECTORY ASSISTANCE	10XXX (1)555-XXXX	555	HNPA
LDC-LOCAL TOLL CALL	10XXX (1)NXX-XXXX	NXX	HNPA
LDC-TOLL FREE LONG DISTANCE	10XXX (1)800-NXX-XXXX	800	FNPA
LDC-LONG DISTANCE DIRECTORY ASSIST	10XXX (1)NIX-555-XXXX	005	FNPA
LDC-TOLL CALL WITHIN HOME NPA	10XXX (1)HNPA-NXX-XXXX	NXX	HNPA
LDC-LONG DISTANCE OUTSIDE OF NPA	10XXX (1)NIX-NXX-XXXX	NIX	FNPA

Legend: N — any digit 2-9
I — digit 0-1
X — any digit 0-9
( ) — an optional digit
LDC — Long-Distance Carrier

Considerations

ARS provides the most-preferred usage of public network facilities available at a system.

Up to 254 Routing Patterns, shared with AAR, can be provided.

Two 3-digit translators (per partition [V3]) are provided.

Up to 32 Toll Lists can be provided.

Up to 32 RHNPA tables (per partition [V3]) can be provided.

If a customer chooses a single primary long-distance carrier for all long distance (1+) calls, then any IDDD (011), operator (0), CDOS (0+ or 01+), 700, and 900 calls also go to that carrier. In order to place a call to an area not served by the primary long-distance carrier, the appropriate 10xxx code must be dialed to access a different carrier who has access to the desired area.

If a customer changes ARS routing assignments, it is the customer's responsibility to notify the Regional Support Center (RSC) network designer and the System Control Office (SCO) technician of the changes in order to receive their continued support.

In certain areas, directory assistance is 1411 instead of just 411. In such cases, the 1 must be dialed before 411. The system cannot be administered to insert the 1.

With V2, if the first digit to be inserted in a routing pattern is a special character such as # or * (anything other than 0 through 9) the dialed digits as well as the inserted digits in the routing pattern will not be outpulsed.

Interactions

The following features interact with the Automatic Route Selection feature.

- Automatic Alternate Routing (AAR)

  ARS and AAR can access the same trunk groups and share the same Routing Patterns.

- Abbreviated Dialing

  FRL checking is bypassed on an ARS call made via a privileged Abbreviated Dialing Group List.

- Attendant Control of Trunk Group Access

  Attendant control of a trunk group, in effect, removes the trunk group from the Routing Pattern. The trunk group is never accessed by the ARS feature. ARS calls do not route to the attendant.

- Code/Toll Restriction

  Code/Toll Restriction is not checked on ARS calls.

- Controlled Restriction, Origination Restriction, and Outward Restriction

  These features prohibit access to ARS.

- Forced Entry of Account Codes

  Prefix marks and other digits inserted from routing patterns will not be used in determining whether a call is a toll call. If Forced Entry of Account Codes is desired for ARS calls, ARS should be administered to require a "1" prefix.

- Generalized Route Selection (G1)

  Generalized Route Selection (GRS) works with ARS to provide call routing over the appropriate trunking facilities. Routing is determined by the type of call being made. With GRS, calls may be routed differently than they would with just ARS. For details on GRS, see the Generalized Route Selection feature description elsewhere in this chapter.

- Miscellaneous Trunk Restrictions

  Miscellaneous Restrictions are not checked on ARS calls.

- Ringback Queuing

  Ringback Queuing can be used on ARS calls originated at the switch that provides the queuing. Incoming tie trunk calls will not queue on an outgoing trunk group.

  If a multi-appearance voice terminal user has an Automatic Callback button, makes an ARS call, and all trunks are busy, Ringback Queuing is activated automatically.

- Station Message Detail Recording (SMDR)

  An ARS call using a trunk group marked for SMDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number. Subnet Trunking does not affect SMDR.

  If SMDR generation is administered for a trunk group assigned to a Routing Pattern, data will be collected for all calls routed through the trunk group. If an SMDR account code is to be dialed with an ARS call, it must be dialed before the ARS access code is dialed.

- Ten-Digit to Seven-Digit Conversion (G1)

  If Ten-Digit to Seven-Digit Conversion is used, normal ARS call routing may be affected. This feature converts certain public network numbers to private network numbers prior to call routing. See the Ten-Digit to Seven-Digit Conversion feature description elsewhere in this chapter for more details.

- Termination Restriction

  No form of Termination Restriction is checked on a trunk used for an ARS call

Administration

ARS is initially assigned on a per-system basis by an AT&T service technician. After the feature is activated, the following items are administered by either the System Manager or the service technician:

- ARS Access Code 1 (1 to 3 digits)

- ARS Access Code 2 (1 to 3 digits) (V3 or G1)
- Three-digit Home NPA Table—Points to the appropriate Routing Pattern for each office code within the home NPA.
- Three-digit Foreign NPA Table—Points to the appropriate Routing Pattern for each nonlocal NPA or points to a 6-digit translator so the call will be routed on both the NPA and the office code.
- Up to 32 Remote Home NPA Tables—Provides 6-digit translation on selected foreign NPAs. Since calls accessing one of these tables route on an office code, similar to the way home NPA calls route, the term Remote Home NPA is used.
- Toll Lists—Provides an indication of whether each office code (with respect to the distant end of the trunk group) is a local or toll call.
- FRLs—Must be assigned via a Class of Restriction to each originating facility. Minimum Facility Restriction Levels (FRLs) required to access a route are assigned as part of the Routing Pattern. Assignment of these values determines the calling privileges of each individual user of the ETN.
- Routing Patterns—Provide an indication of the NPA at the distant end of the trunk group selected for the call and the applicable Toll List number, if any. The Routing Pattern also provides FRL and Subnet Trunking data. (Refer to the Facility Restriction Level and Subnet Trunking feature descriptions for details.)
- Whether or not the system returns dial tone after the ARS feature access code (FAC) is dialed on trunk calls.

Hardware and Software Requirements

ARS may be used on a stand-alone system or may be an integral part of a private network. No additional hardware is required for a stand-alone system. A private network may require additional tie trunks and TN748B Tone Detector circuit packs. These additions are, however, cost effective when compared to the alternatives for call routing.

Optional ARS software is required.

Automatic Wakeup (V3 or G1)

Description

Allows attendants, front desk users, and guests to request that a wakeup call be placed automatically to a certain extension number at a later time. Wakeup requests may be placed from 5 minutes to 23 hours and 55 minutes in advance of the wakeup call.

When a wakeup call is placed and answered, the system can provide a recorded announcement, speech synthesis announcement, music, or simply silence.

All wakeup times entered into the system are rounded to the nearest 5 minutes. For example, a requested time of 6:58 a.m. would be stored in the system as 7:00 a.m. Time validity checks are based on the rounded figure.

Wakeup calls are placed within 2-1/2 minutes of the requested time, and are never rerouted, forwarded, or sent to coverage. Prior to placing the wakeup call, the system deactivates Do Not Disturb for the extension, if applicable.

If a wakeup call attempt is not answered or if the extension is busy, the system will try two more times at 5-minute intervals. If the call is not completed after the three attempts, the system can leave a Leave Word Calling (LWC) message for a designated extension, if administered. In addition, the system maintains a complete record of all wakeup call activity for the past 24 hours.

Touch-tone dialing is required for a wakeup request to be entered. Users with rotary dial terminals must call the attendant to request a wakeup call.

The Automatic Wakeup feature can be activated either by dialing the feature access code (FAC) or by pressing the Automatic Wakeup Entry button. If the FAC is used, the system provides voice prompting. If the Automatic Wakeup Entry button is used, the system provides display prompting.

- Voice Prompting

  A guest can enter his or her own wakeup call request; however, the request can be entered **only** for the extension number where the call is originated.

  After the user dials the Automatic Wakeup FAC, the system generates voice prompts (through the use of a Speech Synthesizer circuit pack). These prompts tell the user when to enter information and what information is needed. The touch-tone buttons are used to enter the required information, and military and standard time are accepted. The user must dial the Automatic Wakeup FAC again to change or delete a wakeup request.

  If invalid entries are made, a standard message is generated to notify the user of the error. The system then repeats the original prompt for input. If an invalid input occurs on the second try, the system informs the user to dial the attendant for assistance.

- Display Prompting

  Display prompting is provided to attendants, front desk users, and to other users with display-equipped voice terminals. Front desk users are administered with console permission Class of Service (COS) and can perform the same actions as the attendant. Other users can enter a wakeup request only for the extension number where they are originating the call.

  The attendant presses the Automatic Wakeup Entry button to activate the feature. If the attendant is on an active call with a system user, the user's extension number will be displayed as the default extension by pressing the pound sign. If the displayed entension number is not the extension number of the user desiring the wakeup call, the attendant can change it. Display prompting continues until the attendant has entered all necessary information and the request for the wakeup call is confirmed.

  If a condition exists that the system does not accept the wakeup request, the system displays the reason for denial. Wakeup requests may be denied for one of the following reasons:

  — Too Soon—Indicates that the requested wakeup time is within the current 5-minute wakeup interval.

  — System Full—Indicates that the maximum number of system wakeup calls has been reached.

  — Interval Full—Indicates that the maximum number of wakeup calls in any 15-minute interval has been reached.

  The attendant can change or cancel a wakeup call request at any time.

When the system places a wakeup call, one of the following will occur:

- Extension Is Busy—The wakeup call will be placed again later.

- No Answer—The system will apply ringing for 30 seconds. If the call is not answered, the system will try again later.

- Ringing Blockage—If four or more ports on the same Analog circuit pack are already ringing, the system will wait 16 seconds and try again. If the second attempt is blocked, the call is considered to have failed and the system will wait 5 minutes before trying again.

- Call Is Answered—When a wakeup call is answered, the guest hears music, a recorded announcement (from the Speech Synthesis circuit pack or from the Audichron* Recorder/Announcer), or silence, according to system administration.

* Registered Trademark of Audichron Corporation

If a wakeup call was not completed because of a busy, no answer, or ringing blockage, the system attempts to place the call two more times at 5-minute intervals. If the call is not completed after the three attempts, the system leaves an LWC message for the designated extension.

A special extension, called the Wakeup Messages Extension, must be administered exclusively for receiving failed wakeup call LWC messages. When such a message is retrieved, the display shows the date, time, and extension number for the failed wakeup call attempt.

An Automatic Message Waiting (AMW) button and associated lamp can be assigned to attendant consoles or front desk terminals. The number associated with the button can be the Wakeup Messages extension. The AMW lamp lights when a failed wakeup message is waiting. The user may retrieve the message by invoking Coverage Message Retrieval on the wakeup message extension. When the button associated with the AMW lamp is pressed, the console or terminal is placed in the Coverage Retrieval mode. The user then retrieves the failed wakeup call attempt messages. Only attendants and specified voice terminal user can retrieve and delete the failed wakeup messages.

The system maintains an audit trail record of all wakeup call activity for the past 24 hours. The System Manager can request that wakeup events be displayed at the System Access Terminal (SAT) (V3) or Manager I terminal (G1), or printed at a designated printer. If the system has a journal printer, wakeup events are printed as they occur.

The audit trail record contains the following information:

- Type of event, such as:
  - Entry of a new request
  - Changed request
  - Canceled request (including a wakeup request that was automatically canceled when the Check-Out feature was activated)
  - Wakeup call placed successfully
  - Wakeup call attempt failed (unsuccessful or skipped)
- Time of the event
- Extension number receiving the call
- Time of the wakeup request
- Extension number (or 0 for the attendant) where the event took place
- Number of call attempts that were placed
- An indication of why a wakeup call attempt failed.

In addition, all wakeup time changes are recorded. This record shows the original time requested and the changed time. The audit trail record is not backed up and all wakeup data is lost if a system failure occurs.

The following reports can be scheduled for printing on a daily basis:

- Wakeup Activity Report—This report summarizes the wakeup activity for each extension that had any wakeup activity over the past 24 hours.
- Wakeup Summary Report—This report gives an hour-by-hour summary of the number of scheduled wakeup calls, the number of wakeup calls that were completed, and a list of extensions to which wakeup calls were attempted but not completed during that hour. The report covers all Automatic Wakeup events for each hour over the past 24-hour period.

Considerations

The Automatic Wakeup feature lessens the attendant's workload since each user can activate the feature and request his or her own wakeup call. In addition, the system places the wakeup calls automatically.

The voice and display prompting assures the user that his or her request is confirmed. Also, the audit trail record information assures the staff that users will not miss their wakeup calls.

The following items should also be considered:

- Verification of Wakeup Announcements—A special access code can be administered for the attendant or front desk users to verify that wakeup announcements are operating properly.
- If an announcement resource is not available or is not operating properly when a wakeup call is placed, the user still receives the call but hears silence instead of an announcement.
- A time change entered at the SAT or Manager I terminal may cause some calls to be skipped. Moving the system clock ahead will skip the calls scheduled during the skipped interval; moving the clock back a maximum of 1-1/2 hours has no effect on wakeup calls. If an initial call attempt was made before the time change, the retry call attempts will still be placed.
- Once a wakeup call request has been completed, skipped, or failed after three attempts, the request is deleted from the system. A record of the call request is, however, maintained in the audit trail record.
- One wakeup request at any one time is allowed per extension number.
- As many as 200 extension numbers may have active wakeup requests for any 15-minute interval.

- The maximum number of wakeup requests per system is 800 (V3) or 1600 (G1). This maximum number is shared with Do Not Disturb requests.

- Wakeup requests may be entered from 5 minutes to 23 hours and 55 minutes in advance of the actual wakeup call. If the requested wakeup hour entered is 0 or from 13 to 23, the system assumes military time. If the requested wakeup hour is from 1 to 12, the system prompts the user to enter 2 (for a.m.) or 7 (for p.m.) to indicate morning or evening.

- Up to ten attendant consoles and/or front desk terminals may be in the wakeup display mode at any one time.

- The number of available speech synthesis ports is the only limit to the number of users entering wakeup call requests at the same time. If overflow occurs, such calls are routed to the attendant or to the specially administered Wakeup Messages Extension.

- Wakeup call attempts are not rerouted, forwarded, or sent to coverage.

Interactions

The following features interact with the Automatic Wakeup feature.

- Attendant or Voice Terminal Display

  If the console or terminal is in the Automatic Wakeup mode and the user presses another display mode button, the Wakeup mode is aborted and the wakeup request is not entered, changed, or deleted.

- Do Not Disturb

  If Do Not Disturb is active at a voice terminal, the Automatic Wakeup feature deactivates Do Not Disturb for that terminal, and then the system places the wakeup call.

- Property Management System (PMS) Interface

  A Check-Out request will cancel an active wakeup call request for the guest room. Also, a Room Change/Room Swap request through the PMS will cause a wakeup request to be changed or swapped.

Administration

The Automatic Wakeup feature is administered by the System Manager. The following items require administration:

- Wakeup call announcement type—Choice is one of the following: music, external recorded announcement, voice synthesis announcement, or silence.

- Length of time to leave a voice terminal connected to the announcement.
- Extension number to receive LWC messages for failed wakeup call attempts.
- Extension number to receive wakeup call attempts when voice synthesis prompting is not available.
- Automatic Wakeup Entry button (per attendant console or display-equipped terminal).
- Feature access code for voice prompting.
- Special access code for the attendant to verify that speech synthesis announcements are operating properly.
- Hospitality-Related System Parameters assignments:
  - Extension number for the wakeup log printer, if a journal printer is used
  - The time for the scheduled Wakeup Activity Report to be printed
  - The time for the scheduled Wakeup Summary Report to be printed.

Hardware and Software Requirements

If voice prompting is used, a TN725 Speech Synthesizer circuit pack is required. Each circuit pack has four ports to provide voice prompting. If speech synthesis is selected for wakeup call announcements, two ports must be reserved for wakeup announcements.

If recorded announcements are used, a model HQD614B Recorder/Announcer manufactured by the Audichron Company is required. Each Recorder/Announcer requires four auxiliary trunk ports which must be on the same TN763B circuit pack.

With hospitality features such as Automatic Wakeup, it is recommended that a journal printer be used with switch configurations larger than 800 stations. The journal printer requires an Modular Processor Data Module (MPDM) and a port on a Digital Line circuit pack or an Asynchronous Data Unit (ADU) and a port on a Data Line circuit pack.

No additional software is required.

Basic Call Management System (G1)

Description

Provides real-time and historical reports which will assist a customer in managing individual agents, Automatic Call Distribution (ACD) splits (hunt groups), and trunk groups. These reports are a subset of those available on the Call Management System (CMS) adjunct. Basic Call Management System (BCMS) reports can be accessed and displayed on the Manager I terminal or printed on demand on the printer associated with the Manager I terminal. In addition, the historical reports can be scheduled to print on the system printer.

The BCMS report feature collects and displays information pertaining to individual agents (based on the agent's extension), ACD splits, and trunk groups. Data is stored by hour or half hour for 25 time intervals (includes current time interval). Daily summary data will also be calculated and stored for 7 days.

The following reports are available with the BCMS:

- Real Time Reports
  - Split Status
  - System Status
- Historical Reports
  - Agent
  - Split
  - System
  - Trunk.

The reports can be displayed and/or printed both locally and remotely. Locally, the reports can be accessed by the ACD administrator from the Manager I terminal. Customers with multiple premises may wish to centralize the measurements data evaluation and hence may access the switch data remotely. Reports can also be scheduled to print on the Report Scheduler System Printer.

An example of each BCMS report follows, along with a brief description of the data in the report. More detailed information on these reports can be found in *DEFINITY Communications System Generic 1 and System 75—Administration and Measurement Reports,* 555-200-500, and in *DEFINITY Communications System Generic 1—Basic Call Management System Operations,* 555-204-703.

Split Status Report

The Split Status Report provides real-time status and measurement data for internally measured agents and the split they are assigned to. Figure 3-5 shows the Split Status Report.

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                                                            Page 1 of x
                       BCMS SPLIT (AGENT) STATUS

       Split: xx
  Split Name: xxxxxxxxxxxxxxx
Calls Waiting: xxx
 Oldest Call: xx:xx

xx=Staffed  xx=Avail  xx=ACD  xx=ACW  xx=AUX  xx=ExtnCalls  xx=OtherSplit

AGENT                EXT      STATE      TIME       ACD      EXTN IN   EXTN OUT
                                                    CALLS    CALLS     CALLS

xxxxxxxxxxxxxxx      xxxxx    xxxxxxx    xx:xx      xxx      xxx       xxx
agent1               12345    Avail      12:00      0        0         0
agent2               12346    ACD        12:04      1        0         0
agent3               12347    ACW        12:12      3        0         0
agent4               12348    AUX        11:30      0        0         0
agent5               12349    ExtnIn     12:08      1        2         0
agent6               12350    Extnout    12:10      0        0         1
agent7               12349    OthrSpl    11:58      0        0         0
agent8               12348    Unstaff    00:00      0        0         0


       Note:  Xs are used to show field size but are not
              displayed on the actual screen form.
```

Figure 3-5. Split Agent Status Report

Split Status Report fields are described below:

- Split—The number of the requested split.
- Split Name—The name of the requested split.
- Calls Waiting—The number of calls currently waiting in this split's queue.
- Oldest Call—The time the current oldest call has waited in the split's queue.
- Staffed—Number of agents currently logged into the split.
- Avail—Number of agents currently available to receive an ACD call in the split. If the agent is on another split's call or in the After Call Work (ACW) mode for another split, this agent is not considered available and will not be recorded here. This includes Outbound Call Management (OCM) calls that are distributed through ACD as well.

- ACD—Number of agents in the split currently on an ACD call for the split.
- ACW (S)—Number of agents in this split currently in ACW for this split.
- AUX (S)—Number of agents in this split currently in AUX work for this split.
- ExtnCalls—Number of agents in the split currently on non-ACD calls, either incoming or outgoing directly to or from their extensions.
- OtherSplit—Number of agents on another split's call or in ACW for another split.
- AGENT—Name of the agent associated with the extension.
- EXT—Extension number of the agent.
- STATE—Current state of the agent for the split. The possible states are Avail, ACD, ACW, AUX, ExtnIn, ExtnOut, OthrSpl, and Unstaff.
- TIME—The time the current state was entered (in hours and minutes).
- ACD CALLS—Number of ACD calls that the agent has completed during the current period.
- EXTN IN CALLS—Number of non-ACD calls that the agent has received and which have completed during the current period.
- EXTN OUT CALLS—Number of non-ACD calls the agent has made and completed during the current period.

System Status Report

The System Status Report provides real-time status information for internally measured splits. Figure 3-6 shows the System Status Report format. The averages are for completed calls only and will include the entire time for the completed call even if the call extends over multiple time periods.

Page 1 of x

BCMS SYSTEM STATUS

SPLIT	CALLS WAIT	OLDEST CALL	AVG ANSW SPEED	AVAIL AGENT	# ABAND	AVG ABAND TIME	# ACD	AVG TALK	AVG AFTER CALL
xxxxxxxxxxxxxxx	xxx	xx:xx	xx:xx	xxx	xxx	xx:xx	xxx	xx:xx	xx:xx
Services	3	1:03	:45	0	3	:30	20	2:30	1:25
Sales	5	:33	:15	0	11	:45	36	1:32	:35

Note: Xs are used here to show the field size but are not displayed on the actual screen form.

Figure 3-6. System Status Report

The System Status Report fields are described below:

- SPLIT—Name of the split being reported.
- CALLS WAIT—Number of calls currently waiting in the split's queue.
- OLDEST CALL—Amount of time that the oldest call has waited in queue.
- AVG ANSW SPEED—Average speed of answer for the split during the current period. This value is calculated as the sum of each call's time in queue plus time ringing at the agent's terminal divided by the total number of completed ACD calls. Calls that intraflow from other splits will not include time in queue from other splits.
- AVAIL AGENT—The number of agents in the split currently available to receive an ACD call from this split.
- # ABAND—Number of calls that have abandoned during the current period. If the queue is full, or if the call is denied because no agents are staffed, it will not count as abandoned.
- AVG ABAND TIME—Average time abandoned calls waited in queue before abandoning during the current period. This value is calculated as the sum of each abandoned call's time in queue (in minutes and seconds) divided by the total number of abandoned calls for this time interval.

- # ACD—Number of ACD calls completed by the split during the current period.
- AVG TALK—Average talk time for ACD calls completed by the split during the current period. This value is calculated as the sum of individual ACD talk time divided by the total number of ACD agents. Time ringing at the agent's voice terminal is not included.
- AVG AFTER CALL—Average ACW time for the ACD calls handled by the split during the current period. ACD calls with no ACW time are included in the average. Time spent on direct incoming or outgoing calls while in the ACW mode will not be included in the average. This value is calculated as the total ACW time minus the total ACW incoming time minus the total ACW outgoing time, divided by the total number of ACD calls.

Agent Report

The Agent Report gives traffic information for internally measured agents. The data in this report is the total for all splits the agent was logged into and can be requested by either the administered time or by daily summaries. Figure 3-7 shows the Time Report and Figure 3-8 shows the Daily Report.

```
                                                                           Page 1
                                  BCMS  AGENT  REPORT

Agent: xxxxxxxxxxxxxxx

                    #      AVG     AVG     TOTAL   TOTAL  # EXTN    AVG EXTN  TOTAL
                   ACD     TALK    AFTER   AVAIL   AUX     IN/OUT   IN/OUT    TIME
TIME              CALLS    TIME    CALL    TIME    TIME    CALLS    CALLS     STAFFED

xx:xx-xx:xx        xxx    xx:xx   xx:xx   xx:xx   xx:xx    xxx      xx:xx     xx:xx
 8:00- 9:00         10     1:15     :45   15:20   10:40     1        4:00     60:00
 9:00-10:00         18     1:40    1:00    6:20     :00     2        3:20     60:00
10:00-11:00         10     1:20     :50    8:54     :00     0         :00     30:00
--------------------------------------------------------------------------------
SUMMARY             38     1:28     :54   30:34   10:40     3        2:26    150:00


        Note:  Xs are used here to show the field size but are not
               displayed on the actual screen form.
```

Figure 3-7. Agent Time Report

Page 1

BCMS AGENT REPORT

Agent: xxxxxxxxxxxxxxx

DAY	# ACD CALLS	AVG TALK TIME	AVG AFTER CALL	TOTAL AVAIL TIME	TOTAL AUX TIME	# EXT IN/OUT CALLS	AVG EXT IN/OUT TIME	TOTAL TIME STAFFED
xx/xx/xx	xxx	xx:xx	xx:xx	xx:xx	xx:xx	xxx	xx:xx	xx:xx
10/17/88	38	1:28	:54	30:34	10:40	3	2:26	150:00
SUMMARY	38	1:28	:54	30:34	10:40	3	2:26	150:00

Note: Xs are used here to show the field size but are not displayed on the actual screen form.

Figure 3-8. Agent Daily Report

The Agent Report fields are described below:

- Agent—Name of the agent being reported.
- TIME/DAY—Time period in military time or date being reported.
- # ACD CALLS—Number of ACD calls completed by this agent for all splits during this period or day.
- AVG TALK TIME—The average time the agent spent talking on ACD calls for all splits during this period or day. This value is calculated as the total ACD time divided by total ACD calls.
- AVG AFTER CALL—Average time the agent spent in ACW for any split during this period or day. This does not include the time spent on direct incoming or outgoing calls while in the ACW mode.
- TOTAL AVAIL TIME—Total time the agent was available to receive ACD calls during the current period or day.
- TOTAL AUX TIME—Total time the agent spent in the AUX work mode for ALL splits (simultaneously) that the agent was logged into during the current period or day. If the agent is handling another split's call while in the AUX work mode, this time will not be considered in this value.
- # EXT IN/OUT CALLS—Total number of non-ACD calls completed by the agent for all splits during this period or day.

- AVG EXT IN/OUT TIME—Total time the agent spent on non-ACD calls while logged into at least one split during this period or day.
- TOTAL TIME STAFFED—Total time the agent spent logged into any split during this period or day. An agent in multiple splits is clocked for staff time as long as he/she is logged into any of the splits. Times for each split are not combined.

Split Report

The Split Report provides traffic information for internally measured splits. The information can be requested by either the administered time or daily period. Figure 3-9 shows the Time Report format and Figure 3-10 shows the Daily Report format.

```
                                                                        Page 1
                              BCMS SPLIT REPORT

Split Number: xx
  Split Name: xxxxxxxxxxxxxxx

                  AVG           AVG     AVG    AVG     #      #
               #  ANSW      #  ABAND    TALK   AFTER  FLOW   FLOW  AUX    AVG
TIME          ACD SPEED  ABAND TIME     TIME   CALL    IN    OUT   TIME  STAFF
xx:xx-xx:xx   xxx  xx:xx   xxx  xx:xx   xx:xx  xx:xx   xxx    xxx   xx:xx  xxx.x
 8:00- 9:00    32    :25     4    :32    5:15    :30     3      5    3:30   4.0
 9:00-10:00     8    :07     1    :03    3:20    :00     0      0    9:30   2.2
---------------------------------------------------------------------------
SUMMARY        40    :21     5    :26    4:52    :26     3      5   13:00   3.1


        Note:  Xs are used here to show the field size but are not
               displayed on the actual screen form.
```

Figure 3-9. Split Time Interval Report

```
                                                                  Page 1
                                  BCMS SPLIT REPORT

Split Number: xx
  Split Name: xxxxxxxxxxxxxxx

                    AVG            AVG    AVG    AVG      #      #
                #   ANSW     #     ABAND  TALK   AFTER  FLOW   FLOW   AUX    AVG
DAY            ACD  SPEED  ABAND   TIME   TIME   CALL    IN    OUT    TIME  STAFF
xx/xx/xx       xxx  xx:xx   xxx    xx:xx  xx:xx  xx:xx   xxx   xxx   xx:xx  xxx.x
10/17/88        40    :21     5      :26   4:52    :26     3     5   13:00    3.1
----------------------------------------------------------------------------
SUMMARY         40    :21     5      :26   4:52    :26     3     5   13:00    3.1


        Note:  Xs are used here to show the field size but are not
               displayed on the actual screen form.
```

Figure 3-10. Split Daily Interval Report

The Split Summary Report fields are described below:

- Split Number—The number of the split for which this report has been requested.
- Split Name—The name of the split for which this report has been requested.
- TIME/DAY—The time period, in military time, or day being reported.
- # ACD—The number of ACD calls handled by the split during the period or day.
- AVG ANSW SPEED—The average speed of answer for ACD calls handled by the split during the period or day. This value is calculated as the sum of each call's time in queue plus time ringing at the agent's terminal divided by the total number of answered ACD calls. Calls that intraflow from other splits will not include time in queue from other splits.
- # ABAND—Total number of ACD calls abandoned for this split during this period or day.
- AVG ABAND TIME—Average time for ACD calls abandoned for this split during this period or day. This does not include time spent in another split's queue before intraflowing to this split. This value is calculated as the total abandon time divided by the total number of abandoned calls.
- AVG TALK TIME—Average talk time for the ACD calls handled by this split during this period or day. Time on hold is included.
- AVG AFTER CALL—Average time agents in this split spent in ACW during the period or day. Time spent in the ACW mode while on a direct call is not included. This value is calculated as the total time in ACW divided by the total number of ACD calls.

- # FLOW IN—Total number of calls that this split accepts as a coverage point from another split/extension or that are call forwarded to this split during this period/day.
- # FLOW OUT—Total number of calls this split successfully extends to its own coverage point, call forwards out, or are answered via call pickup during this period or day.
- AUX TIME—Total time agents in the split spent in AUX for this split during this period or day. Time spent in OthrSpl is included here also.
- AVG STAFF—Average number of people staffed for this split during this period/day.

System Report

The System Report gives traffic information for all internally measured splits. The information can be requested by either the administered time period or daily. Figure 3-11 shows the Time Report and Figure 3-12 shows the Daily Report.

```
                                                                            Page 1
                                   BCMS SYSTEM REPORT

Time: xx:xx-xx:xx

                       AVG           AVG   AVG    AVG     #      #
                   #   ANSW    #     ABAND TALK   AFTER FLOW   FLOW AUX    AVG
SPLIT            ACD SPEED ABAND TIME  TIME   CALL   IN    OUT   TIME  STAFF

xxxxxxxxxx      xxxx xx:xx  xxx  xx:xx xx:xx xx:xx  xxx   xxx   xx:xx  xx.x
services          40   :21    5    :26  4:52   :15    3     5    5:50   3.1
sales              5  1:07    0    :00  8:02  2:20    0     0    9:20   2.0
--------------------------------------------------------------------------------
SUMMARY           45   :26    5    :26  5:13   :29    3     5   15:10   2.5


        Note:  Xs are used here to show the field size but are not
               displayed on the actual screen form.
```

Figure 3-11. System Time Report

```
                                                                    Page 1
                               BCMS SYSTEM REPORT

Day:   xx/xx/xx

                          AVG              AVG    AVG    AVG     #      #
                     #    ANSW     #      ABAND  TALK   AFTER  FLOW    FLOW  AUX     AVG
SPLIT               ACD  SPEED  ABAND    TIME   TIME   CALL    IN      OUT   TIME  STAFF

xxxxxxxxxxxxxx     xxxx  xx:xx   xxx     xx:xx  xx:xx  xx:xx   xxx     xxx   xx:xx  xx.x
services             40    :21     5       :26   4:52    :15     3       5    5:50   3.1
sales                 5   1:07     0       :00   8:02   2:20     0       0    9:20   2.0
-----------------------------------------------------------------------------------------
SUMMARY              45    :26     5       :26   5:13    :29     3       5   15:10   2.5


       Note:  Xs are used here to show the field size but are not
              displayed on the actual screen form.
```

Figure 3-12. System Daily Report

The System Report fields are described below:

- Page—The current page being displayed.
- Time/Day—The interval time or day for which the report was requested.
- SPLIT—Name of the split being reported.
- # ACD—Number of ACD calls handled by the split during this period or day.
- AVG ANSW SPEED—Average speed of answer for ACD calls handled by the split during this period or day. This value is calculated as the sum of each call's time in queue plus time ringing at the agent's terminal divided by the total number of answered ACD calls. Calls that intraflow from other splits will not include time in queue from other splits.
- # ABAND—Total number of ACD calls abandoned for the split during this period or day.
- AVG ABAND TIME—Average time ACD calls wait before abandoning for this split during this period or day. This value is calculated as the sum of each abandoned call's time in queue (in minutes and seconds) divided by the total number of abandoned calls for this time interval.
- AVG TALK TIME—Average talk time for the ACD calls handled by this split during this period or day.
- AVG AFTER CALL—Average time agents in the split spent in ACW during this period or day.

- # FLOW IN—Total number of calls the split accepts as a coverage point from another split/extension or that are call forwarded to this split during this period or day.
- # FLOW OUT—Total number of calls the split successfully extends to its own coverage, call forwards out, or are answered via call pickup during this period or day.
- AUX TIME—Total time agents in the split spent in AUX for this split during this period or day. Time spent in OthrSpl is included here also.
- AVG STAFF—Average number of agents staffed for the split during this period or day.

Trunk Report

The Trunk Report gives statistical information for all internally measured trunk groups. The information can be requested by either the administered time period or daily period. Figure 3-13 shows the Trunk Interval Time Report and Figure 3-14 shows the Trunk Daily Interval Report.

```
                                                                          Page 1
                                 BCMS TRUNK GROUP REPORT

      Trunk Group: xx
  Trunk Group Name:  xxxxxxxxxxxxxxx
  Number of Trunks: xx

                      INCOMING        |          OUTGOING        |% ALL % TIME
TIME          CALLS ABAND TIME    CCS | CALLS    COMP TIME   CCS | BUSY   MAINT

xx:xx          xxx  xxx   xx:xx   xx.xx    xxx    xxx  xx:xx  xx.xx   xx     xx
10:00           82    5    1:54   29.89      5      5   1:39   2.52    0      0
-----------------------------------------------------------------------------
SUMMARY         82    5    1:54   29.89      5      5   1:39   2.52    0      0


        Note:  Xs are used here to show the field size but are not
               displayed on the actual screen form.
```

Figure 3-13. Trunk Time Report

```
                                                                   Page 1

                                BCMS TRUNK GROUP REPORT

      Trunk Group: xx
  Trunk Group Name: xxxxxxxxxxxxxxx
  Number of Trunks: xx

                  INCOMING            |        OUTGOING        |% ALL % TIME
DAY         CALLS ABAND TIME     CCS  | CALLS   COMP TIME   CCS | BUSY   MAINT

xx/xx/xx     xxx  xxx   xx:xx   xx.xx    xxx     xxx  xx:xx  xx.xx  xx      xx
03/20/90      82    5    1:54   29.89      5       5   1:39   2.52   0       0
-------------------------------------------------------------------------
SUMMARY       82    5    1:54   29.89      5       5   1:39   2.52   0       0


        Note:  Xs are used here to show the field size but are not
               displayed on the actual screen form.
```

Figure 3-14. Trunk Daily Report

The Trunk Report fields are described below:

- Page—The page the user is currently on.
- Trunk Group—The trunk group number requested for this report.
- Trunk Group Name—The name assigned to the trunk group.
- Number of Trunks—The number of trunks in the trunk group being reported.
- TIME/DAY—The time, in military time, or day being reported.
- INCOMING CALLS—Number of incoming calls carried during this period or day for the trunk group.
- INCOMING ABAND—Number of incoming calls abandoned during this period/day.
- INCOMING TIME—Average holding time of incoming calls to the trunk group for the period or day. (Total time undivided by total calls in.)
- INCOMING CCS—Total holding time for incoming calls in hundred call seconds for the trunk group during this period or day.
- OUTGOING CALLS—Total number of outgoing calls for the trunk group during this period or day.
- OUTGOING COMP—Total number of outgoing calls that were answered during this period or day.

- OUTGOING TIME—Average holding time of outgoing calls in the trunk group during this period or day. (Total time out divided by total calls out.)
- OUTGOING CCS—Total holding time for outgoing calls in CCS for the trunk group during this period or day.
- % ALL BUSY—Percentage of time all the trunks in the trunk group that were busy during this period or day. (Total all busy time divided by time interval multiplied by 100.)
- % TIME MAINT—Percentage of time trunks were busied out for maintenance during this period or day. (Total maintenance busy time divided by time interval multiplied by 100.)

Commands

The following list shows the commands that may be used at the Manager I terminal to generate BCMS reports:

- monitor bcms split
- monitor bcms system
- list bcms agent
- list bcms split
- list bcms system
- list bcms trunk

The "monitor" commands display real-time status reports for agents and splits on the Manager I terminal. When a status report is displayed on the Manager I terminal it is automatically updated about every 30 seconds. An UPDATE key is also provided for updates on demand.

The "list" commands display historical information for agents, splits, and trunk groups.

Report Scheduler and System Printer

The report scheduler allows the System Manager to use the Manager I terminal to schedule BCMS reports as well as other reports, lists, etc., to be printed at specific times by an asynchronous printer. Reports are scheduled at 15-minute intervals for any combination of days of the week. Details on the report scheduler can be found in the Report Scheduler and System Printer feature description elsewhere in this chapter and in *DEFINITY Communications System Generic 1 and System 75—Administration and Measurement Reports,* 555-200-500.

Considerations

BCMS provides a set of internal switch measurement reports for telemarketing centers or customer service centers. These reports can help in managing ACD splits (hunt groups) without the need for an adjunct Call Management System (CMS).

The maximum number of measured agents for the BCMS feature is limited to 30. An agent can be a member of up to 3 splits, but will be treated as a single agent.

The maximum number of CMS measured agents (both basic and adjunct) is restricted to 400.

The maximum number of internally measured trunk groups is limited to 30.

The maximum number of internally measured splits is limited to 30. If a split is assigned more than 30 agents it cannot be measured internally.

A maximum of 25 time slices are allocated for storing data. A time slice can be either a 1 hour or a 1/2 hour interval.

A maximum of 7 summary days is stored for each historical report.

The maximum number of internally measured trunk group members is limited to 400.

The addition of an Expansion Port Network (EPN) can affect the operation of the measurements only when the EPN is unavailable. Any resource that resides in the EPN cabinet will not be available for use or for measurement data. If a remote Manager I terminal is connected to the EPN and the fiber link goes down, the Manager I terminal session will be dropped and the login prompt will appear.

Interactions

The following features interact with the Basic Call Management System feature.

- Call Coverage

  Calls extended to a BCMS measured split as a coverage point will be treated like new incoming calls to that split. These calls will increment the FLOW IN field on the Split report, providing they covered from the queue of another BCMS measured split. Calls successfully going to a coverage point from a BCMS measured split are included in the FLOW OUT field on the Split report. Again, those calls must have first been in queue for the split. Calls which cover due to the split queue being full will not cause the FLOW OUT field to be incremented.

- Call Forwarding

  Calls forwarded to a BCMS measured split will be treated like new incoming calls to that split. INFLOW and OUTFLOW counts will not be affected.

  ACD calls do not forward out of a split. Extension calls will be forwarded, but they will not be pegged in the FLOW OUT column.

- Call Pickup

  Calls answered using the call pickup feature will be treated as non-ACD calls (EXTN IN) for the agent picking up the call. Calls that are picked up for a BCMS measured agent are included in the FLOW OUT column on the Split report.

- Conference/Transfer

  When an agent receives an ACD call, the agent is put into the ACD state and the call is added to the call count for the split. Any agents conference into the call will be put in the ExtIn state and no changes will be made to the split's call count. If the call is transferred, the answering agent will be put into the ACD state and the call will be added to the answering split's call count. The transferring agent will then be returned to the Avail state.

- Hunt Groups

  The BCMS measurements are not determined in the same way as hunt group measurements although some of the information is similar. Therefore, the two reports may represent the data differently.

- Move Agents From CMS

  If agents are moved from one split to another split by the CMS adjunct, measurements will be stopped for the agent's "from" split and started for the agent's "to" split. The adjunct CMS denies agent move requests when agents are logged in (staffed). This denial is important since it eliminates measurement complications associated with move requests when the agent is on an ACD call. Move requests are also denied if the agent is being moved into an unmeasured split.

  If the adjunct CMS attempts to move an agent that is not being measured by BCMS into a split that is being measured by BCMS, and the move would exceed the maximum of 30 measured agents, the switch will reject the move. Otherwise, internal BCMS measurements will be started for the agent. If the adjunct CMS moves an agent from a split that is measured by BCMS to a split that is not BCMS measured, internal measurements for the agent will be stopped.

- Night Service

  When night service is activated for a split, new calls will go to an alternate destination. The split in night service will not consider these calls to be OUTFLOW. The calls will be treated as new incoming calls if the destination is a measured split (that is, they will not be considered INFLOW).

- System Measurements

  A DEFINITY Communications System Generic 1 can have BCMS reports, adjunct CMS reports, and switch traffic measurements simultaneously.

  The BCMS measurements are not determined in the same way as trunk group measurements although some of the information is similar. Therefore, the two reports may represent the data differently.

Administration

The Basic Call Management System is administered by the System Manager. The following items require administration:

Communication Interfaces

If the link to the adjunct CMS has been administered, CMS measurements must be busied out **(busyout mis** command) in order to add/remove adjunct CMS and BCMS measured agents or trunks at the switch. When the "busyout" has been released, adjunct CMS checks for translation changes and, if they exist, the current database is updated and measurements are restarted.

If the link has not been administered (BCMS measurements only), the **busyout mis** command will not be required to change translation data.

Hunt Group and Trunk Group

The Measured field on the Hunt Group and Trunk Group forms should be administered as one of the following:

- internal - measured by BCMS only
- external - measured by CMS adjunct only
- both - measured by both BCMS and adjunct CMS
- none - not measured (default).

If BCMS has not been administered in customer options, neither "internal" nor "both" will be allowed. If the split or trunk group is measured by BCMS only, the **busyout mis** command will not be required to make changes. Measurements can be turned off for a split while agents are logged in, but agents must be logged off to start measurements.

System Administration

The BCMS field must be set to "y" by an authorized AT&T employee.

System-Parameters

The following items require administration:

- Measurement Interval—Specifies what time interval is used for polling and reporting measurement data. The time can be specified by "hour" or "half-hour" intervals with "hour" as the default. There is a maximum of 25 time slots available for measurement intervals. If hourly is specified, an entire day of traffic information will be available for history reports; otherwise, only half a day will be available. This does not affect daily summaries as they will always reflect traffic information for the entire day. The interval may be changed at any time, but will not go into effect until the top of the hour.

- Printer Information for the System Printer—This includes the printer extension, EIA device bit rate, and lines per page.

Hardware and Software Requirements

No additional hardware is required to support the BCMS feature. However, a customer may decide to use an asynchronous system printer to obtain hard copies of BCMS history reports. The system printer can be interfaced to the switch through the EIA port on the processor board or through any of the alternate data interfaces, such as Data Modules (PDMs) connected to a digital port, or Asynchronous Data Units (ADUs) connected to a data line circuit port.

BCMS software is required.

Bridged Call Appearance—Multi-Appearance Voice Terminal

Description

The appearance of a voice terminal's primary extension number at another voice terminal is called a bridged call appearance. The Bridged Call Appearance feature is used by lifting the handset and pressing the Bridged Appearance button. The user is then bridged onto the other voice terminal's primary extension number and can handle calls on that extension number. The bridged appearance can be used to originate calls from, and answer calls to, the other voice terminal's primary extension number. The user can also bridge onto an existing call to or from the other voice terminal.

An incoming call rings the primary extension number's voice terminal and all voice terminals that have a bridged call appearance of the voice terminal's primary extension number. Each voice terminal is visually alerted for all bridged appearances on the voice terminal, but has the option of audible ringing.

A bridged call appearance can be assigned to any 2-lamp button. It does not require the use a regular call appearance. A bridged call appearance can be used just like a regular call appearance for most features. For example, the Hold, Transfer, and Priority Calling features can be used from a bridged appearance, just as they would be used from a regular call appearance.

Considerations

The Bridged Call Appearance feature allows calls to be handled from more than one voice terminal. Some practical uses of this capability are as follows:

- A secretary making or answering calls on an executive's primary extension

  These calls can be placed on hold for later retrieval by the executive, or the executive can simply bridge onto the call. In all cases, the executive handles the call as if he or she had placed or answered the call. It is never necessary to transfer the call to the executive.

- A secretary taking care of details for an executive who is already active on a call

  A secretary can bridge onto an active call and take down information such as an address or telephone number. This frees the executive for more important matters.

- Visitor telephones

  An executive may have another voice terminal in his or her office which is to be used by visitors. It may be desirable that the visitor be able to bridge onto a call which is active on the executive's primary extension number. A bridged call appearance makes this possible.

- Service environments

  It may be necessary that several people be able to handle calls to a particular extension number. For example, several users may be required to answer calls to a hot line number in addition to their normal functions. Each user may also be required to bridge onto existing hot line calls. A bridged call appearance provides this capability.

- A user frequently using voice terminals in different locations

  A user may not spend all of his or her time in the same place. For this type of user, it is convenient to have his or her extension number bridged at several different voice terminals.

A voice terminal's primary extension number can have an appearance on up to seven other voice terminals. The number of bridged call appearances allowed at each voice terminal is limited only by the number of 2-lamp buttons available on the voice terminal.

Up to six parties can be off-hook and involved in a conversation on a bridged appearance of an extension.

A maximum of 400 (V1), 500 (V2), 800 (V3), or 1600 (G1) bridged call appearances are allowed per system.

It is recommended that a bridging voice terminal have a bridged call appearance corresponding to each call appearance of the primary extension number at the bridged voice terminal. For example, if a bridged voice terminal has three call appearances of its primary extension, a bridging voice terminal should have three bridged call appearances of the extension. This allows users to refer to the individual call appearances when talking about a specific call.

Bridged call appearances may result in the reduction of available feature buttons, thereby reducing a user's capabilities. A Call Coverage module can be used to provide up to 20 bridged call appearances. This leaves the other call appearances available for use with other features. If a user's primary extension is assigned to a Call Coverage module, that call appearance cannot be bridged.

If a call terminates at a voice terminal on an extension number other than the primary extension number (for example, Terminating Extension Group, Uniform Call Distribution Group, Call Coverage Answer Group, or Direct Department Calling Group extension number), a bridged call appearance is not maintained. Therefore, it is recommended that the primary terminal not be made a member of such a group (even though administration of this is not prohibited).

With V1, V2, and V3, a single-line voice terminal cannot have any bridged call appearances and cannot bridge onto other extensions. With G1, a single-line voice terminal can have a bridged call appearance on one or more multi-appearance voice terminals.

The Bridged Call Appearance feature should not be considered as a replacement for Call Coverage.

Interactions

The following features interact with the Bridged Call Appearance—Multi-Appearance Voice Terminal feature.

- Abbreviated Dialing

  A user, accessing Abbreviated Dialing while on a bridged call appearance, accesses his or her own Abbreviated Dialing lists. The user does not access the Abbreviated Dialing lists of the primary extension associated with the bridged call appearance.

- Attendant Display and Voice Terminal Display

  A call from the primary extension number or from a bridged call appearance of the primary extension number is displayed as a call from the primary extension number (that is, the call will be displayed as coming from the primary extension number regardless of which appearance it was placed from).

- Automatic Callback

  Automatic Callback calls cannot originate from a bridged call appearance. However, when a call is originated from a primary extension number, the return call notification rings at all bridged call appearances, alerting all bridging user's that a call has been returned to the primary extension number; and all display modules (that is, primary, bridging user(s), attendant) associated with the call will show that it is a callback call.

- Call Coverage

  Coverage criteria for bridged call appearances is based entirely on the criteria of the primary extension associated with the bridged appearance. That is, a call to the primary extension that requires call coverage treatment will follow the coverage path of the primary extension and not the path of any of the bridged appearances.

  It is recommended that the primary terminal not be a member of a call coverage group, because calls to the primary terminal as a member of the group **will not be bridged.**

- Call Forwarding All Calls

  Call Forwarding All Calls can be activated or canceled for the primary extension number from any bridged call appearance of that number; then, when activated, calls to the primary extension number do not terminate at the bridged call appearances, but go to the designated forwarding destination. Bridged call appearances do not receive redirection notification of the call to the primary extension when it is forwarded.

- Call Park

  When a call is parked from a bridged call appearance, it is parked on the primary extension number.

- Call Pickup

  If a voice terminal receives ringing on a bridged call appearance, the incoming call can be picked up by members of that voice terminal's Call Pickup group. This causes all bridged call appearances to be dropped. The call is parked on the primary extension number of the answering voice terminal.

- Class of Restriction

  The Class of Restriction assigned to a voice terminal's primary extension also applies to calls originated from a bridged call appearance.

- Conference—Attendant and Conference—Terminal

  Conferences can be set up using the usual conference operations. Either a primary extension button or a bridged appearance button can be used to make the calls to be added to the conference. When using both bridged and primary appearances to form a conference, the last call should be placed on a bridged appearance so that the primary user can easily bridge onto the call and so that only one appearance will be used for the conference.

  The display will show the number of active parties in a call, including active bridged appearances.

  If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the Conference/Transfer button, must select a call appearance to be used for the conference, before dialing the number.

  If the bridging user has at least one available bridged appearance of the primary extension (other than the one he or she is currently on), the system will automatically select a bridged call appearance for the conference when the Conference/Transfer button is pressed.

- Consult

  Bridged call appearances of the primary extension do not ring on a consult call to the primary extension.

- Exclusion

  Exclusion prevents any other user from bridging onto the call. Activation of exclusion by any user (primary or bridged appearance), prior to placing a call, will prevent any other user from bridging onto the call. Activation of exclusion by any user active on a call, while the primary user and/or any other bridging users are active on the call, will drop all other users from the call (including the primary user), leaving only the activator and the calling/called party on the call.

- Hold

  Any user (primary or bridged appearance) can place an active call on hold. If only one user is active on a call and places that call on hold, then the indicator lamp at the activators appearance button shows that the call is on hold. If more than one user is bridged onto the active call, and one of the users activates Hold, the activator receives "hold" indication for the call and all other bridged users receive "active" indication for the call. Hold indications for this feature are the same as for the Conference feature.

- Hunt Group (DDC or UCD)

  Bridged call appearances cannot be used in conjunction with DDC or UCD hunt groups.

  A call to the primary terminal that is directed to a hunt group cannot be bridged onto by stations with bridged call appearances of the primary terminal.

  If a member of a UCD group is off-hook on a bridged appearance, and that user's primary extension number (which is in the UCD group) is idle, then a UCD call may terminate on that user's *primary extension number.*

- Intercom

  Intercom calls to the primary extension will not ring at the associated bridged appearances.

- Last Number Dialed

  Activation of the Last Number Dialed feature causes the last number dialed from the voice terminal to be redialed, regardless of the extension number used (primary or bridged call appearance).

- Leave Word Calling

  A Leave Word Calling message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension number assigned to the bridged call appearance.

  When a user calls a primary extension, and activates Leave Word Calling, the message is left for the primary extension, even if the call was answered at a bridged call appearance.

  Leave Word Calling messages left by the primary user can be canceled by a bridged appearance user (for example, the secretary can cancel a Leave Word Calling message left by the boss).

- Personal Central Office Line

  If a user is active on his or her primary extension number on a Personal Central Office Line (PCOL) call, bridged call appearances of that extension number cannot be used to bridge onto the call. The call can only be bridged onto if another voice terminal is a member of the same PCOL group and has a PCOL button.

- Privacy-Manual Exclusion

  When Privacy-Manual Exclusion is activated, all other users are prevented from bridging onto the active call.

- Ringback Queuing

  Ringback Queuing is not provided on calls originated from a bridged call appearance.

- Ringer Cutoff

  When activated at a multi-appearance station, Ringer Cutoff prevents any non-priority (or non-intercom) incoming call from ringing at that station, whether the call is to the station's primary extension or to any of the bridged appearances' extension(s). Manual Signaling is not affected by Ringer Cutoff.

- Terminating Extension Group (TEG)

  A call to the primary terminal that is directed to a TEG cannot be bridged onto by terminals with bridged appearances of the primary terminal. The primary terminal **should not** be assigned to a TEG.

- Transfer

  If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the Conference/Transfer button, must select a call appearance to be used for the transfer, before dialing the number.

  If the bridging user has at least one available bridged appearance of the primary extension (other than the one he or she is currently on), the system will automatically select a bridged call appearance for the transfer when the Conference/Transfer button is pressed.

- Voice Message Retrieval

  A voice message to the primary extension can be retrieved on a bridged appearance by the bridged appearance user.

- Voice Paging

  The use of Voice Paging automatically invokes exclusion; therefore, interactions for this feature are the same as for Exclusion.

Administration

The Bridged Call Appearance—Multi-Appearance Voice Terminal feature is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Bridged Appearance buttons (per voice terminal)
- Audible ringing for bridged appearances (per voice terminal).

Hardware and Software Requirements

No additional hardware or software is required. A Call Coverage module can be used to provide up to 20 bridged call appearances.

Bridged Call Appearance—Single-Line Voice Terminal (G1)

Description

Allows a multi-appearance terminal to have an appearance of a single-line voice terminal's extension number. The appearance of the single-line voice terminal's extension number at a multi-appearance terminal is called a bridged call appearance.

The bridged call appearance can be used to originate, answer, or bridge onto an existing call to or from the single-line user's extension number.

The multi-appearance terminal user can use the bridged call appearance by lifting the handset and pressing the bridged appearance button, or by pressing the bridged appearance button and lifting the handset. The user is then bridged onto the single-line voice terminal's extension number and can handle calls on that extension number.

The single-line user can also bridge onto an existing call originated, answered, or bridged onto by the associated multi-appearance terminal(s) by just going off-hook.

An incoming call will ring at the single-line voice terminal, and at all voice terminals that have a bridged call appearance of the single-line voice terminal's extension number. Each bridging voice terminal has visual alerting with the option of audible ringing for the bridged appearance of the single-line voice terminal.

When the single-line terminal user answers the call, the audible ringing stops at the single-line terminal and at all the bridging user's terminals, and the status lamps at all the bridged appearance buttons light steadily. The call can then be bridged onto by any of the bridging users.

When a bridging user answers the call, the audible ringing stops at the single-line terminal and at all of the bridging user's terminals, and the status lamps at all of the bridged appearance buttons light steadily. The call can then be bridged onto by any of the bridging user's or by the single-line voice terminal user. However, after ringing ceases at the single-line terminal, the single-line terminal user **has no indication of the call's existence.** In this case, if the single-line user did not hear the ringing, the user would not know of the existence of the call and could inadvertently pick up on an active call.

A bridged call appearance can be assigned to any 2-lamp button. It does not require the use of a regular call appearance.

A bridged call appearance can only be used to originate and/or answer calls on the single-line voice terminal's extension number or to bridge onto an active call. The bridging user **CANNOT** access a Call Waiting call or a call on hard hold. Also, the bridging user cannot access a call that has been put on soft hold by the single-line terminal user.

Because of the aforementioned restrictions, certain limitations are placed on the use of the Conference, Transfer, and Hold features for both the single-line user and the bridging users. When more than **one** user is active on a call (that is, a single-line user and one or more bridging users, or two or more bridging users, or any configuration that has more than one bridging party on an established call), attempts to use the Conference, Transfer, or Hold features are denied.

When Call Waiting and/or Priority Call Waiting is assigned to the single-line terminal, it is only active when the single-line terminal user is alone on a call; it is **NOT** active when the multi-appearance terminal user is alone on a call on the bridged appearance of the single-line terminal.

When a single-line user is alone on an active call, normal single-line Conference, Transfer, and Hold procedures apply.

When a bridging user is alone on an active call on the single-line terminal's extension number, then normal multi-appearance terminal Conference, Transfer, and Hold procedures apply.

Considerations

The bridging of a single-line terminal satisfies certain conditions that require handling a call from locations other than that of the single-line terminal. Some of these situations are as follows:

- A secretary placing calls for an executive(s)

  These calls can be placed on hold for later retrieval by the executive, or the executive can simply bridge onto the call. In all cases, the executive handles the call as if he or she had placed or answered the call. It is never necessary to transfer the call to the executive.

- A secretary taking care of details for an executive, such as a call to the finance department, traffic department, etc., (any call that requires automatic identification of the executives extension number).

  A secretary can bridge onto an active call and take down information such as an address or telephone number. This frees the executive for more important matters.

- Visitor telephones

  An executive may have another voice terminal in his or her office which is to be used by visitors. It may be desirable that the visitor be able to bridge onto a call which is active on the executive's primary extension number.

- Service environments

  It may be necessary that several people be able to handle calls to a particular extension number. For example, several users may be required to answer calls to a hot line number in addition to their normal functions. Each user may also be required to bridge onto existing hot line calls.

- A user frequently using voice terminals in different locations

  A user may not always spend time in the same place. For this type user, it is convenient to have their extension number bridged at several different voice terminals.

In the rest of this feature description, the single-line terminal and/or terminal user will be referred to as the "primary" terminal and/or terminal user, and the multi-appearance terminal and/or terminal user will be referred to as the "bridging user."

The primary terminal's extension number can have an appearance on up to seven bridging user's terminals. A bridging user cannot have more than one bridged appearance for a particular primary (analog) terminal. However, a bridging user can have appearances of more than one analog terminal on their terminal (that is, a bridging user, by use of different buttons, can bridge onto several different primary terminals).

The number of bridged appearances allowed on a bridging user's terminal is limited only by the number of 2-lamp buttons available on the terminal.

If the primary terminal (single-line terminal) is correctly administered, **but not in service,** calls can still be placed, by the bridging users, and received on the bridged appearances of the terminal. The primary terminal can be out of service for several reasons (such as an unplugged terminal, a non-existent terminal craft busyout command, etc.).

A maximum of 1600 bridged call appearances (any mix of analog and multi-appearance) is allowed per Generic 1 system.

If more than one user goes off-hook on a bridged appearance at the same time, only the user that was actually the first to go off-hook can dial.

If a bridging user **is not** active on a call, and bridges onto the appearance of an active call, then the user will be bridged onto the active call. If a bridging user **is** active on a call, and bridges onto the appearance of an active call, then the previously selected call will be dropped and the user will be bridged onto the active call.

The Exclusion feature can be activated by **the bridging user only,** while active on a call, to prevent accidental bridging of an active call.

If a call terminates at a voice terminal on an extension number other than the primary extension number (for example, Terminating Extension Group, Uniform Call Distribution Group, Call Coverage Answer Group, or Direct Department Calling Group extension number), a bridged call appearance is not maintained. Therefore, it is recommended that the primary terminal not be made a member of such a group (even though administration of this is not prohibited).

The Bridged Call Appearance feature **should not** be considered as a replacement for Call Coverage or any other similar features.

Interactions

In the following discussion of interactions, the term **"TERMINAL-BASED"** means that it does not matter to the system whether a call appearance or a bridged appearance is being used to activate/deactivate the feature. The term **"EXTENSION-BASED"** means that activation/deactivation of the feature from a bridged appearance is seen by the system as having been made by the primary (single-line) terminal.

- Abbreviated Dialing

  Abbreviated Dialing is **TERMINAL-BASED.** This means that a bridging user, accessing Abbreviated Dialing while on a bridged call appearance, accesses the Abbreviated Dialing lists of the primary terminal associated with the bridged call appearance.

- Attendant Display and Voice Terminal Display

  A call from the primary extension number or from a bridged call appearance of the primary extension number is displayed as a call from the primary extension number (that is, the call will be displayed as coming from the primary extension number regardless of which appearance it was placed from).

- Authorization (Class of Service, Class of Restriction, Facility Restriction Level)

  The Class of Service and Class of Restriction (including restrictions and Facility Restriction Level) of the primary terminal are always used when authorization checking is required, even when the call is originated from a bridged appearance button by a bridging user.

- Automatic Callback

  A bridging user that originates a call on a bridged appearance **cannot** activate Automatic Callback.

  The primary terminal user can activate Automatic Callback, then the callback call will alert the primary terminal user **and** the bridged appearances with priority alerting; if a display module is provided at the bridging user's terminal, it will show that the call is a callback call.

- Call Coverage

  It is recommended that the primary (analog) terminal not be a member of a call coverage group, because calls to the primary terminal as a member of the group **will not be bridged.**

  If the primary terminal is made a member of a coverage group, coverage criteria is based entirely on the criteria of the primary terminal. This means that a call to the primary terminal that requires call coverage treatment will follow the path of the primary terminal and not the path of any of the terminals with bridged appearances of the primary terminals. In this case, it is desirable to have the bridging user in the coverage path of the primary terminal. Then, when a call to the primary terminal requires coverage treatment, it will follow the coverage path to the bridging user's terminal, call appearances of the call will be dropped, and the call will terminate at the bridging user's terminal as a coverage call.

- Call Forwarding All Calls

  Call Forwarding All Calls is **EXTENSION-BASED;** it can be activated or canceled for the primary terminal number from the primary terminal or from any bridged call appearance of that number; then, when activated, calls to the primary extension number do not terminate at the bridged call appearances, but go to the designated forwarding destination. Bridged call appearances do not receive redirection notification of the call to the primary extension when it is forwarded.

- Call Park

  Call Park is **EXTENSION-BASED;** when a call is parked from a bridged call appearance, it is parked on the primary terminal extension number.

- Call Pickup

  Calls to the primary terminal, alerting at bridged appearances of the primary terminal, can be picked up by member's of the bridging user's Call Pickup group; this causes all bridged appearances of the call to be dropped.

  Calls ringing at a primary terminal can be picked up by member's of the primary terminal's Call Pickup group. However, if the primary terminal and the bridging user's terminal are not in the same Call Pickup group, then the bridging user cannot pick up calls to other members of the primary terminal's Call Pickup group.

  Originating on a bridged appearance and dialing the Call Pickup FAC will be interpreted as an attempt to pick up a call from the primary terminal's Call Pickup group.

  A bridging user can use Call Pickup to pick up a call that is alerting at a bridged appearance, instead of selecting the bridged appearance button. This will cause the call to terminate on the bridging user's primary extension button; and the primary terminal and all bridged appearances of the call will be dropped.

  If the bridging user has appearances of numerous single-line (primary) terminals (for example, Sales, Service, Warehouse, etc.), and it is not desired that the calls be answered by anyone other than the primary terminal user or the bridging users, then the bridging user(s) should not be assigned to a pickup group.

- Call Waiting Termination and Priority Calling

  Call Waiting Termination and Priority Calling apply only to an active call on the primary terminal which has no one else bridged on. If the primary terminal user and one or more bridging users are active on a call, Call Waiting and Priority Calling are denied.

- Conference

  A bridged call cannot be conferenced if more than one user is active on that call. This is because the bridging user has no access to the call after the primary terminal user places the call on soft hold; and the primary terminal user has no access to the bridging user's call appearance used for conference/transfer attempts.

  If a bridging user is active on a bridged call and the primary terminal user attempts a conference, the attempt is ignored. The same is true if a bridging user attempts a conference when the primary terminal user and another bridging user is active on a call.

When the primary terminal user is active on a call, and no other bridging user is active on the call, then that call can be placed on hold by the primary terminal user utilizing normal single-line conference procedures. Any attempt by a bridging user to bridge onto the call during a successful conference attempt will be denied.

A bridging user, alone on a bridged call, can conference the call utilizing the normal multi-appearance terminal conference procedures. Any attempt by the primary terminal user to bridge onto the call during a successful conference attempt will be ignored; any attempt by other bridging users will be denied (standard denial response will be returned to the bridged appearance).

If a conference is not allowed because of the preceding limitations, the user can accomplish a transfer by asking an internal non-bridged party in the connection to create the conference, or ask the remaining bridging users and/or primary user to disconnect so that the conference can be completed. At completion of the conference, the parties that left the call can reenter the call, if control of the conference remains with the primary terminal. If control of the conference does not remain with the primary terminal, the bridging user must conference the primary terminal and the bridging user back into the call as required.

If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the Conference/Transfer button, must select a call appearance to be used for the conference, before dialing the number.

If the bridging user has at least one available bridged appearance of the primary extension (other than the one he or she is currently on), the system will automatically select a bridged call appearance for the conference when the Conference/Transfer button is pressed.

- Data Privacy/Data Restriction

  When Data Privacy is activated or Data Restriction is assigned to a station involved in a bridged call and the primary terminal and/or bridging user attempts to bridge onto the call, Data Privacy and Data Restriction are automatically deactivated.

- Exclusion

  Exclusion can only be activated by a bridging user (a button is required for Exclusion). Activation of Exclusion will prohibit any further bridging onto the call. If a bridging user activates Exclusion while the primary terminal and/or other bridging users are active on the call, the primary terminal and all bridging users except the activator are dropped from the call.

- Hold

  A call cannot be put on hold if more than one user is active on that call.

  The primary terminal user, when no other bridges are active on the call, can put the call on hold, using normal single-line hold procedures. If the primary terminal user successfully soft holds the call, the status lamp at all of the bridged appearances shows the hold indication; and then the call can be put on hard hold by dialing the hard hold feature access code (FAC). The hard held call is no longer accessible to the bridging users until it is taken off hold by the primary terminal user. After the call is put on hard hold, any new call to the primary terminal is tracked by the bridged appearances.

A bridging user can place an active call on hold (if the primary terminal or any other bridges are not active on the call) by using normal multi-appearance hold procedures. Any attempt to enter the held call will return it to the status of an active call that can then be accessed using bridging procedures.

If hold is not allowed because of the preceding reasons, the user can just go on-hook and then reenter the call as required, because the call remains accessible as long as the primary terminal or any bridging user is active on it.

- Hot Line Service

  If a single-line voice terminal is administered for Hot Line Service, bridged appearances of that voice terminal's extension will also place a hot line call automatically when a user goes off-hook on that bridged appearance.

- Hunt Group (DDC or UCD)

  Bridged call appearances cannot be used in conjunction with DDC or UCD hunt groups.

  A call to the primary terminal that is directed to a hunt group cannot be bridged onto by stations with bridged call appearances of the primary terminal.

- Last Number Dialed

  Activation of the Last Number Dialed feature causes the last number dialed from the activating voice terminal to be redialed, regardless of the extension number used (primary or bridged call appearance).

- Leave Word Calling

  A Leave Word Calling message left by a user on a bridged call appearance leaves a message for the called party to call the primary terminal extension number (that is, the feature is **EXTENSION-BASED).**

  When a user calls a primary terminal, and activates Leave Word Calling, the message is left for the primary terminal, even if the call was answered at a bridged call appearance.

  Leave Word Calling messages left by the primary terminal user can be canceled by a bridged appearance user (for example, the secretary can cancel a Leave Word Calling message left by the boss).

- Personal Central Office Line

  A single-line primary terminal cannot be a member of a PCOL group.

- Preference

  Ringing Line Preference will select an alerting bridged appearance; Idle Line Preference will not.

- Priority Calling

  The primary terminal user or the bridging user can make a priority call. If a priority call is made to an idle primary terminal, the primary terminal and all bridging users will be alerted by priority alerting.

  For information on termination of a priority call to an active primary terminal, see Call Waiting Termination/Priority Calling.

- Ringer Cutoff

  Ringer Cutoff requires a button; therefore, it cannot be activated by the primary terminal user. Bridging user activation of Ringer Cutoff has no impact on the primary terminal or the other bridging users. This is a **TERMINAL-BASED** feature.

- Ringback Queuing

  Ringback Queuing is not provided on calls originated from a bridged call appearance. Ringback Queuing is automatically invoked for a single-line terminal (primary terminal).

- Service Observing

  The primary terminal user or bridging user can bridge onto a Service Observed call at any time. If the primary terminal is being Service Observed and an incoming call is answered by the bridging user, the call is not observed unless or until the primary terminal user bridges onto the call. Conversely, if the bridging user is being Service Observed and an incoming call is answered by the primary terminal user, the call is not observed unless or until the bridging user bridges onto the call.

  If the bridging user activates Service Observing, utilizing a bridged appearance, Service Observing is activated for the bridging user (that is, the feature is **TERMINAL-BASED).**

- Station Message Detail Recording (SMDR)

  If a bridging user originates and/or answers a call on a bridged appearance, the primary terminal is recorded as the calling/called terminal. A conference or transfer by a bridging user also appears as though it was performed by the primary terminal user.

- Terminating Extension Group (TEG)

  A call to the primary terminal that is directed to a TEG cannot be bridged onto by terminals with bridged appearances of the primary terminal. The primary terminal **should not** be assigned to a TEG.

- Transfer

  A call cannot be transferred if more than one user is active on that call.

  The primary terminal user, when no other bridges are active on the call, can transfer the call, using normal single-line transfer procedures. Any attempt by a bridging user to bridge onto this call during a successful transfer attempt will be denied (standard denial response will be returned to the bridged appearance).

  A bridging user, alone on a bridged call, can transfer the call, using normal multi-appearance transfer procedures. Any attempt by the primary terminal user to bridge onto this call during a successful transfer attempt will be ignored; and any attempt to bridge on by a bridging user will be denied.

  If transfer is not allowed for one of the preceding reasons, the user can ask an internal non-bridged party in the connection to transfer the call; or can ask the remaining bridging users and/or primary terminal user to disconnect so that the transfer can be completed.

  If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the Conference/Transfer button, must select a call appearance to be used for the transfer, before dialing the number.

  If the bridging user has at least one available bridged appearance of the primary extension (other than the one he or she is currently on), the system will automatically select a bridged call appearance for the transfer when the Conference/Transfer button is pressed.

- Voice Message Retrieval

  A voice message to the primary terminal can be retrieved on a bridged appearance by the bridging user. If a security code is required to retrieve the message, the bridging user must use the security code of the primary terminal.

- Voice Paging

  The use of Voice Paging automatically invokes exclusion; therefore, interactions for this feature are the same as for Exclusion.

Administration

The Bridged Call Appearance—Single-Line Voice Terminal feature is administered on a per-voice terminal basis by the system manager. The following items require administration per voice terminal:

- Analog Bridged Appearance button (abrdg-appr)—only one per single-line terminal, per bridging user terminal. A bridging user terminal can have more than one Analog Bridged Appearance button, but can only have one terminal for each single-line terminal.

- Bridged Appearance buttons (brdg-appr)—as many as required for bridging onto other multi-appearance terminals.

- Audible ringing and/or visual alerting for bridged appearances—per multi-button terminal.

The primary terminal (single-line terminal) must be translated before any bridged appearances can be translated to point to it. Also, the bridged appearances of the primary terminal must be removed before the primary terminal can be removed.

Hardware and Software Requirements

No additional hardware or software is required. A Call Coverage module can be used to provide up to 20 bridged call appearances.

Busy Verification of Terminals and Trunks (V2, V3, or G1)

Description

Allows attendants and specified multi-appearance voice terminal users to make test calls to trunks, voice terminals, and hunt groups [Direct Department Calling (DDC) and Uniform Call Distribution (UCD) groups]. These test calls check the status of an apparently busy resource.

Busy verification of voice terminal extensions, hunt group extensions, and trunks can be done by either multi-appearance voice terminal users or attendants. Feature activation is via a Busy Verify button.

An attendant or multi-appearance voice terminal user can activate Busy Verification of Terminals and Trunks by pressing the Busy Verify button. The attendant then dials an extension number if a voice terminal or hunt group is to be verified. If a trunk is to be verified, the attendant dials a trunk access code, followed by a 2-digit number (leading 0s may be required) to specify which member of the trunk group is to be verified.

After an attendant or multi-appearance voice terminal user has activated the Busy Verification of Terminals and Trunks, the system checks the validity of the entered extension number or trunk access code and member number. If the entered number is not a voice terminal extension number, a DDC/UCD group extension number, an Automatic Call Distribution (ACD) split number, or a trunk access code with a valid member number, the verification attempt is denied.

If an attendant activates Busy Verification of Terminals and Trunks for a valid voice terminal extension number, the system initiates a priority call to that extension. One of the following then occurs:

- Voice terminal is idle.

  Priority ringing is heard at the voice terminal and the voice terminal is successfully verified. The call proceeds as a normal attendant-originated call.

- Voice terminal is active on a call.

  The system first searches for an idle call appearance on the voice terminal. If one is found, that call appearance is rung. If an idle call appearance cannot be found, or if the voice terminal is a single-line voice terminal, the attendant will bridge onto the active call. All parties on the active call receive a warning tone (2-second burst of 440-Hz tone) to let them know that the attendant is bridging onto the call. A 1/2-second burst of warning tone is repeated every 15 seconds, as long as the attendant is bridged onto the call.

- Voice terminal is out of service.

  Busy verification is denied and the attendant receives reorder tone.

If an attendant activates Busy Verification of Terminals and Trunks for a valid ACD split, UCD group, or DDC group, the system initiates a priority call to that group. One of the following then occurs:

- At least one group member is available for incoming calls.

  The call rings the available group member and is treated as a priority call from an attendant to the group.

- All group members have activated the Make Busy function.

  Busy verification is denied and the attendant receives reorder tone.

- Not all group members have activated Make Busy, but no group members are available for incoming calls.

  The call will not queue if a queue is available. Busy verification is denied.

If an attendant or a multi-appearance voice terminal user activates Busy Verification of Terminals and Trunks for a valid trunk, the system checks the status of that trunk. One of the following then occurs:

- The trunk is idle.

  If the trunk is an outgoing trunk, the originator of the busy verification receives dial tone and can make a call on that trunk to verify that it is in working order. If the trunk is an incoming trunk, the originator of the busy verification receives confirmation tone as an indication that the trunk is available for use.

- The trunk is busy with an active call.

  The originator of the busy verification is bridged onto the active call. All parties on the active call receive a warning tone (2-second burst of 440-Hz tone) to let them know that the originator of the busy verification is bridging onto the call. A 1/2-second burst of warning tone repeats every 15 seconds, as long as the busy verification originator remains on the call.

- The trunk is out of service.

  The busy verification is denied. The attendant receives intercept tone.

If busy verification is denied for any other reason, intercept tone or reorder tone is returned to the user.

Considerations

Busy Verification of Terminals and Trunks provides attendants with an easy method of checking the condition of certain extensions and trunks. An attendant or multifunction voice terminal can distinguish between a voice terminal that is truly busy and one that only appears busy because of some trouble condition. Attendants or multifunction voice terminal users can also use this feature to quickly identify faulty trunks. As a result, better communications service is provided and faulty trunks can be corrected more quickly.

A busy verification can be performed on the following:

- Voice terminal extensions
- UCD and DDC hunt group extensions
- Members of the following types of trunk groups:
  - Direct Inward Dialing (DID)
  - Central Office (CO)
  - Foreign Exchange (FX)
  - Wide Area Telecommunications Service (WATS)
  - Advanced Private Line Termination (APLT)
  - Tie
  - Remote Access
  - Release Link Trunk (RLT).

The bridging capability associated with Busy Verification of Terminals and Trunks is not provided on verification attempts to UCD and DDC groups or RLTs.

Outgoing test calls cannot be made on DID trunks.

Interactions

The following features interact with the Busy Verification of Terminals and Trunks feature.

- Automatic Callback

  Once the called party in an Automatic Callback call hangs up, neither extension number can be busy verified until both the calling and called parties are connected or the callback attempt is canceled (by the activating party or by time-out of the callback interval).

- Call Coverage

  Since the busy verification call to an extension number is originated as a priority call, the call does not go to coverage.

- Call Forwarding

  A busy verification made to an extension with call forwarding activated does not busy verify the forwarded-to extension. Only the called extension is busy verified.

- Call Waiting Termination

  A busy verification cannot be made to an extension that is waiting to be answered at another extension.

- Conference—Attendant and Terminal

  If a conference call involves six parties, busy verification on any extension number in the conference is denied. If the number of parties in the conference is five or less, a busy verification can be performed on any of the associated extension numbers.

- Data Privacy

  Busy verification is denied if it results in a bridging attempt on a voice terminal that has activated Data Privacy.

- Data Restriction

  If Data Restriction is active on a call, and a busy verification bridging attempt is made on that call, the busy verification is denied.

- Hold

  A busy verification of a multi-appearance voice terminal is denied if all call appearances have calls on hold.

- Individual Attendant Access

  An attendant cannot make a busy verification of another individual attendant console or of the attendant group.

- Loudspeaker Paging Access

  If the voice terminal or trunk to be verified is connected to paging equipment, the verification attempt is denied.

- Voice Terminal Origination Restriction

  A voice terminal that is origination restricted can be assigned a Busy Verify button. However, the button cannot be used.

- Voice Terminal Termination Restriction

  Voice terminals that are termination restricted cannot be busy verified.

- Transfer

  Once the originator of the busy verification has bridged onto a call, any attempt to transfer the call is denied until the originator drops from the call.

Administration

Busy Verification of Terminals and Trunks is administered on a per-voice terminal or per-console basis by the System Manager. The only administration required is the assignment of a Busy Verify button to the desired attendant consoles and multi-appearance voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Call By Call Service Selection (G1)

Description

Allows a single Integrated Services Digital Network (lSDN) Primary Rate Increase (PRI) trunk group to carry calls to many services or facilities [such as a Software Defined Network (SDN), MEGACOM telecommunications service, MEGACOM 800 service, etc.] and/or to carry calls using different Inter-exchange Carriers.

Call By Call Service Selection uses the same routing tables and routing preferences that are used by Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), and Generalized Route Selection (GRS). The service or facility used on an outgoing Call By Call Service Selection call is determined by information assigned in the AAR/ARS/GRS routing patterns.

As an example of how Call By Call Service Selection works, see Figure 3-15. Without Call By Call Service Selection, each trunk group must be dedicated to a specific service or facility. Call By Call Service Selection eliminates this requirement by allowing a variety of services to use a single trunk group. These services are specified on a call-by-call basis. Trunking efficiency is immediately obtained with Call By Call Service Selection by distributing traffic over the total number of available trunks.

Services Used With Call By Call Service Selection

The services used on incoming and outgoing Call By Call Service Selection calls are assigned after an ISDN-PRI trunk group is assigned a service type of Call By Call Service Selection. A Call By Call Service Selection trunk group can be administered to carry calls to many services. The available services are as follows:

- ACCUNET® Digital Service—AT&T's digital network services for various high volume, high speed data transmission requirements.
- INWATS—Provides OUTWATS-like pricing and service for incoming calls.
- AT&T Long-Distance Service—A shared use, two-way, premises-to-premises service that uses the public switched network to transmit and receive voice, data, and graphics communications.
- OUTWATS Band—WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user's location to defined service areas commonly referred to as bands. Currently, the widest band is 5.
- MEGACOM—Provides an AT&T service that provides unbanded long-distance services using special access (PBX to 4 ESS™ switch) from an AT&T node.
- MEGACOM 800—Provides an AT&T service that provides unbanded 800 service using special egress (4 ESS switch to PBX) from an AT&T node.
- Network Operator—Provides access to the network operator.

- Software Defined Network (SDN)—An AT&T offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
- Presubscribed Common Carrier Operator—Provides access to the presubscribed common carrier operator.
- Maximum Banded WATS—A WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by the user.
- Other User-Defined Services—New service types can be assigned as they are developed and defined.

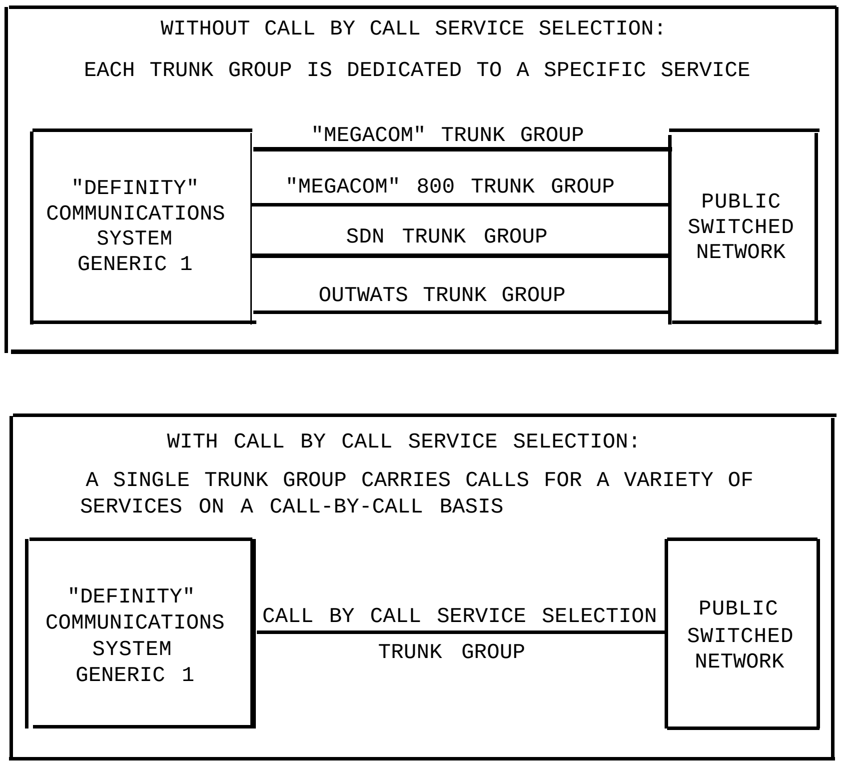

Figure 3-15. Call By Call Service Selection Example

ISDN-PRI Messages and Information Elements Used for Call By Call Service Selection

Although the technical details of ISDN-PRI messages and information elements are not critical to implementing the ISDN-PRI application, the following details may aid in the understanding of some readers and are, therefore, included in this description.

Call By Call Service Selection allows the system to specify one of the preceding service types on a call-by-call basis. This is done via a SETUP message that is present on ISDN-PRI calls. This SETUP message indicates the intent of the originating system to initiate a call using the specified service or facility. The SETUP message contains units called information elements which specify call-related information. The information elements used with Call By Call Service Selection are as follows:

- Network Specific Facility—Indicates which facilities or services are to be used to complete the call.

  The system also checks all incoming ISDN-PRI calls for the presence of a Network Specific Facility information element. If this information element is present, the system makes sure that the requested service is compatible with the administration of the trunk. If the requested service is not compatible with administration, the switch rejects the call.

  For an outgoing call on a Call By Call trunk group, the Network Specific Facility information element is constructed using the "Service/Feature" specified on the routing pattern preference selected for the call.

- Transit Network Selection—Indicates which Inter-exchange Carrier is to be used on an inter-LATA call.

  If a call requires both the Service/Feature and the Inter-exchange Carrier to be specified, the Inter-exchange Carrier information will be sent in the Network Specific Facility information element rather than the Transit Network Selection information element.

Usage Allocation Plan

Optional Usage Allocation Plans may be assigned to provide more control over a Call By Call Service Selection trunk group. Up to three Usage Allocation Plans can be assigned for each Call By Call Service Selection trunk group. A Usage Allocation Plan allows the customer to set the following:

- A maximum number of trunk group members that each specific service can use at any given time.
- A minimum number of trunk group members that will always be available for each specific service.

The sum of the allocation plan maximums may exceed the total number of trunk group members. For example, if a trunk group has 15 members and provides access to MEGACOM service, MEGACOM 800 service, and SDN, the maximum number of trunks to be used for each of these services could possibly add up to more than 15. In this case, for example, you could administer a maximum of seven MEGACOM service calls, six MEGACOM 800 service calls, and eight SDN calls. This ensures that all trunk group members are not dominated by a specific service, yet allows for periodic fluctuations in demand.

The sum of the allocation plan minimums may not exceed the total number of trunk group members. For example, if a trunk group has ten members and provides access to MEGACOM service, MEGACOM 800 service, and SDN, the minimum number of trunks to be used for each of these services cannot add up to more than ten.

If a Usage Allocation Plan has been defined for a Call By Call Service Selection trunk group, and the type of the incoming call exceeds one of the plan's limits, the system will reject the call, even if a trunk is available. If a Usage Allocation Plan has been defined for a Call By Call Service Selection trunk group, and a system user makes an outgoing call of a type that exceeds one of the plan's limits, the user will receive reorder tone unless other preferences are available.

As previously mentioned, each Call By Call Service Selection trunk group can have as many as three Usage Allocation Plans. The customer can assign either fixed or scheduled allocation plans for each Call By Call Service Selection trunk group, as described here (see Figure 3-16 for an example of the screen form used to schedule Usage Allocation Plans and Figure 3-17 for an example of the actual Usage Allocation Plan screen form):

- Fixed

  One plan applies at all times. The minimum and maximum usages specified in this plan will be in effect for the trunk group at all times.

- Scheduled

  Two or three plans can be administered to apply at different times based on the time of day and day of the week. As many as six activation times and associated plans can be assigned for each day of the week. At the specified activation time, the associated plan goes into effect for the Call By Call Service Selection trunk group.

```
          CBC SERVICE TYPE USAGE ALLOCATION PLAN ASSIGNMENT SCHEDULE
                                                         Page x of x

Usage Method:

        Fixed? y       Allocation Plan Number: 1
    Scheduled? n

       Time   #    Time   #    Time   #    Time   #    Time   #    Time   #

Sun    00:00  1    _:_    _    _:_    _    _:_    _    _:_    _    _:_    _

Mon    00:00  1    _:_    _    _:_    _    _:_    _    _:_    _    _:_    _

Tue    00:00  1    _:_    _    _:_    _    _:_    _    _:_    _    _:_    _

Wed    00:00  1    _:_    _    _:_    _    _:_    _    _:_    _    _:_    _

Thu    00:00  1    _:_    _    _:_    _    _:_    _    _:_    _    _:_    _

Fri    00:00  1    _:_    _    _:_    _    _:_    _    _:_    _    _:_    _

Sat    00:00  1    _:_    _    _:_    _    _:_    _    _:_    _    _:_    _
```

Figure 3-16. Screen Form to Schedule Usage Allocation Plans Example

```
                    CBC SERVICE TYPE USAGE ALLOCATION
                                                          Page 3 of 8

Trunk Allocation Plan 1    Trunk Allocation Plan 2    Trunk Allocation Plan 3

  Service/   Min# Max#       Service/   Min# Max#       Service/   Min# Max#
  Feature    Chan Chan       Feature    Chan Chan       Feature    Chan Chan

____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
____________   _    _      ____________   _    _      ____________   _    _
```

Figure 3-17. Screen Form Used to Assign Actual Usage Allocation Plans Example

System administration allows the customer to have anything from a simple fixed usage allocation plan to a very flexible plan with many scheduling options. The customer can even start out with no allocation plan and build the plan as the need arises. This allows the customer to respond to periodic fluctuations in the environment in a more timely manner. The customer does not have to involve the network to fine tune the trunk group administration. To ensure that administration complexity is kept to a minimum, the following steps should be followed when assigning Usage Allocation Plans.

1. Assign a Usage Allocation Plan for a Call By Call Service Selection trunk group.

2. If scheduling is desired, add one or two more Usage Allocation Plans for that trunk group.

3. Administer the scheduling information for the trunk group's Usage Allocation Plans.

Incoming Call Handling Treatment

Call By Call Service Selection provides special Incoming Call Handling Treatment for ISDN-PRI trunk groups. An incoming call on an ISDN-PRI trunk group is handled according to a treatment table that is administered for that trunk group. An example of the screen form that contains this table is shown in Figure 3-18.

```
TRUNK FEATURES                                                    Page 2 of x

         ACA Assignment? y            Long Holding Time(hours): 1

Short Holding Time(secs. ): 10        Short Holding Threshold: 15

           MIS Measured? n                       Internal Alert? n

           Used For DCS? y      PBX ID: 1    Data Restriction? n

      Maintenance Tests? y                             Send SID? n

                                                      Send Name? n
```

Service/ Feature	Called Len	Called Number	Del	Insert	SID/ANI	Night Serv
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______
______	__	______	__	______		______

Figure 3-18. Blank Screen Form to Provide Treatment of Incoming Call By Call Service Selection Calls Example

As shown in Figure 3-18, a variety of specifications can be administered in the table for the treatment of incoming calls. The table allows for as many as 12 different combinations of call treatments. Seven columns are provided for seven different specifications. These specifications are as follows:

- Selection Criteria
  - **Service/Feature**—Specifies the ISDN-PRI Services/Features that can be requested at call setup time when using this trunk group. These services/features are those described previously in the "Services Used With Call By Call Service Selection" section of this feature description. The identifier "other" can be used for all Services/Features not explicitly specified.
  - **Called Len**—Specifies the number of digits contained in the Called Party Number (the digits received for the incoming call). The number of digits contained in the Information Element (IE) must exactly match the number of digits specified in this field. Allowable entries are 0 through 16 or blank. A blank entry indicates a "wild card", meaning that a called number of any length will match.
  - **Called Number**—Specifies the leading digits contained in the Called Party Number. For this row to be chosen for a call, the data is this field must exactly match the leading digits in the IE. A blank entry indicates that the incoming digits are not significant for this entry. Allowable entries are 1 through 16 digits or blank.
- Action Taken Based on Selection Criteria
  - **Del**—Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type may be administered to be routed to a single destination by deleting all incoming digits and then administering the Insert field with the desired extension.
  - **Insert**—Specifies the digits to be prepended to the front of the Called Party Number. The new number is used to route the call. Allowable entries are up to 16 digits or blank.
  - **SID/ANI**—Specifies your preference of a SID (Station Identification) or ANI (Automatic Number Identification) request for this type of call. A blank or "none" indicates that the switch will not request either SID or ANI for any incoming calls of this type. Allowable entries are ANI only, prefer ANI but accept SID, SID only, and prefer SID but accept ANI.
  - **Night Serv**—Specifies a night service extension per Service/Feature. An entry other than blank overrides the night destination entry on page 1 of the form. Allowable entries are an assigned extension, attendant, or blank.

The treatment for an incoming call is selected based on the Service/Feature, Called Len, and Called Number fields in the table. When the attributes of an incoming call match these specifications, the call is treated according to the corresponding Del, Insert, SID/ANI, and

Night Serv specifications. If an incoming call matches more than one set of specifications, the most restrictive case is selected. The following table lists the possible cases in order of most restrictive to least restrictive:

	Service/ Feature	Called Len	Called Number
Most Restrictive	Specified	Specified	X Number Of Leading Digits Specified
	Specified	Specified	Y Number Of Leading Digits Specified, Where Y < X
	Specified	Specified	Not Specified
	Specified	Not Specified	Not Specified
	Specified as "other"	Specified	X Number Of Leading Digits Specified
	Specified as "other"	Specified	Y Number Of Leading Digits Specified, Where Y < X
	Specified as "other"	Specified	Not Specified
Least Restrictive	Specified as "other"	Not Specified	Not Specified

Considerations

Call By Call Service Selection provides the following benefits:

- Cost Reduction—Since many services share the same trunks, the total number of trunks can be reduced.
- Improved Service—Call By Call Service Selection trunks can reduce the probability of features and services being blocked.
- Simplified Networking—Network engineering is simplified because analysis of trunking needs can be done based on total traffic instead of on a per-service basis.

- The ability to respond to changes in a more timely fashion. The network does not have to be consulted because of the flexibility provided by the usage allocation plans.
- Measurement of Call By Call Service Selection calls.

Interactions

The following features interact with the Call By Call Service Selection feature.

- Automatic Alternate Routing

  Call By Call Service Selection uses the same routing tables and routing preferences that are used by Automatic Alternate Routing.

- Automatic Route Selection

  Call By Call Service Selection uses the same routing tables and routing preferences that are used by Automatic Route Selection.

- Generalized Route Selection

  Call By Call Service Selection uses the same routing tables and routing preferences that are used by Generalized Route Selection.

- Station Message Detail Recording

  On successful incoming and outgoing Call By Call Service Selection calls, the Network Specific Facility specified by the call's Network Specific Facility Information Element is recorded by Station Message Detail Recording (SMDR). SMDR refers to this information as the "INS" (ISDN Network Service).

  If an outgoing Call By Call Service Selection call uses an Inter-exchange Carrier (IXC) other than the presubscribed common carrier, SMDR will record the 3-digit IXC code.

  When a Call By Call Service Selection call is rejected because of a trunk group usage allocation plan, SMDR records the cause as an "ineffective call attempt."

- Time of Day Routing

  Any Time of Day Routing administration that affects routing preference will also affect Call By Call Service Selection.

  The Time of Day Routing feature can be used to vary the IXC based on the time of day and day of the week.

- Traffic Measurements

  The system provides traffic measurements for each individual service administered as part of the ISDN Call By Call Service Selection trunk group.

Administration

Call By Call Service Selection is administered by the System Manager on a per trunk group basis. The following items require administration:

- ISDN-PRI Trunk Group—Must be administered with a Service Type of Call By Call Service Selection. The trunk group administration also includes the following:
  - Incoming Call Handling Treatment
  - Whether or Not Usage Allocation Plans Are Required
  - Usage Allocation Plans
  - Usage Allocation Plan Assignment Schedule
  - Group Member Assignments
- AAR/ARS Routing Patterns—Routing Patterns can be administered to include a Network Specific Facility and Inter-exchange Carrier.
- Network Specific Facilities Encoding—New Network Specific Facilities can be added as needed by the craft.

Hardware and Software Requirements

A TN767 DS1 circuit pack is required for assignment of a signaling link and up to 23 ISDN-PRI Trunk Group members. The DS1 provides 24 ports. A TN741 or TN768 Tone Clock circuit pack is required to provide synchronization for the DS1 circuit pack. A TN765 Processor Interface circuit pack is required for use with the TN767 DS1 circuit pack.

Optional ARS software is required.

Call Coverage

Description

Provides automatic redirection of certain calls to alternate answering positions in a Call Coverage path.

Call Coverage Path

A Call Coverage path is a list of one, two, or three alternate answering positions (covering users) that will be accessed, in sequence, when the called individual or group (principal) is not available to answer the call. Any of the following can be assigned a Call Coverage path, and are thus eligible to have their calls redirected to coverage.

- Voice terminal
- Uniform Call Distribution (UCD) group
- Direct Department Calling (DDC) group
- Terminating Extension Group (TEG)
- Personal Central Office Line (PCOL) group
- Automatic Call Distribution (ACD) split (V3 or G1).

The System Manager establishes the coverage paths and sets the redirection criteria at the time the system is implemented. These paths and criteria can be changed at later dates. If a coverage path is not assigned to a particular facility, calls will not be redirected from that facility, unless another feature such as Call Forwarding All Calls is assigned. A coverage path can include any of the following:

- Voice terminal.
- Attendant group.
- UCD group.
- DDC group.
- ACD split.
- Coverage Answer group, which is a group of up to eight voice terminals specifically established to answer redirected calls. All group members are rung simultaneously. Any group member can answer the call.
- Message Center (if an Applications Processor is provided with the system).
- Audio Information Exchange (AUDIX).

Multiple Coverage Paths (V3 or G1)

With Version 3 or DEFINITY Generic 1 systems, a principal can be assigned multiple coverage paths. Each extension is assigned a coverage path. That coverage path, in turn, can be linked to up to three other coverage paths. This makes a total of four coverage paths that can be assigned to each extension. If a call gets redirected from each of the points in the first coverage path, the call will then go to the next coverage path. If the call is not answered by a point in that coverage path, is goes to the next path, and so on.

Redirection Criteria

The redirection criteria determine the conditions under which a call redirects from the principal (called) extension number to the first position in the coverage path. The criteria and conditions that apply are as follows:

- Active

  Redirects calls to call coverage immediately when the principal is active on at least one call appearance. For a voice terminal with only one appearance or a single-line extension, the Busy criterion (described below) should be assigned instead of the Active criterion.

- Busy

  Redirects calls to coverage when all available call appearances at the principal extension are in use. For multi-appearance voice terminals, one call appearance is reserved for outgoing calls or incoming priority calls (described later). The remaining assigned call appearances are available for other incoming calls. An incoming call (other than a priority call) will redirect to coverage only when all of these unreserved call appearances are in use. If at least one unreserved call appearance is idle at the principal extension, the call will remain at that idle appearance.

  A TEG is considered busy if any voice terminal in the group is active on a call.

  For a UCD or DDC group, each voice terminal in the group must be active on at least one call appearance in order for the call to be redirected to coverage. If any voice terminal in the group is idle (not active on any call appearance) the call directs to that voice terminal. If no voice terminal is available, the call can queue if queuing is provided. If queuing is not provided or if the queue is full, the call routes to coverage. Queued calls will remain in queue for a time interval equal to the Don’t Answer Interval (described next).

- Don’t Answer

  Redirects calls to coverage if unanswered during the assigned Don’t Answer Interval (1 to 9 ringing cycles [V1 and V2] or 1 to 99 ringing cycles [V3 and G1]). A call will ring for the assigned Don’t Answer Interval and then redirect to coverage.

- Cover All Calls

  Redirects all incoming calls to coverage. This criterion has precedence over any other criterion previously assigned.

- Send All Calls/Go To Cover (V3 or G1)

  Allows users to activate Send All Calls or Go To Cover as an overriding coverage criteria. This redirection criteria must be assigned before a user can activate the Send All Calls or Go To Cover features (described later).

- No Coverage

  Occurs when none of the above criteria have been assigned. Calls are only redirected to coverage when the principal has activated Send All Calls or the caller has activated Go to Cover. Both of these overriding criteria are described later.

Redirection criteria can be assigned in combinations; Active/Don't Answer and Busy/Don't Answer can be useful. Other combinations are not possible or do not provide any useful function. For example, Active/Busy does not accomplish anything. A busy voice terminal is always active.

Redirection criteria is assigned separately for internal and external calls. Thus, Busy/Don't Answer can be assigned for internal calls and Active can be assigned for external calls. Similarly, Busy/Don't Answer could apply for external calls and No Coverage could apply for internal calls. In the latter case, internal calls remain directed to the called terminal or group.

Certain overriding criteria are possible. These criteria, of course, are checked before the redirection criteria are checked. The overriding criteria are:

- Go to Cover

  Allows users, when making a call to another internal extension, to send the call directly to coverage. This is optionally assigned to a button on a voice terminal and is activated by the internal calling party. Use of Go to Cover is described later.

- Send All Calls

  Allows principals to temporarily direct all incoming calls to coverage regardless of the assigned redirection criteria. For example, if the redirection criteria are administered so that no calls redirect, all incoming calls will terminate at the principal's voice terminal unless Send All Calls is activated. Also, activating Send All Calls allows covering users to temporarily remove their voice terminals from the coverage path.

  Send All Calls is activated by pressing the Send All Calls button or by dialing the Send All Calls access code. The option is deactivated by pressing the button a second time or by dialing the deactivate code.

  If a user is not assigned a coverage path with Send All Calls or Cover All Calls redirection criteria, that user cannot activate Send All Calls. An activation attempt under this condition is denied for both button and dial access.

Send All Calls is similar to Cover All Calls, described previously. However, Cover All Calls is set by the System Manager and would be used for screening the principal's call. The principal may or may not be rung on an incoming call, depending on how this function is assigned. Send All Calls is controlled by the principal and is normally used when the principal will be away temporarily.

TEG calls are not affected by the activation of Send All Calls.

If a user has activated Send All Calls and only has one coverage point, and receives a call from that coverage point, the call will ring silently at the user's voice terminal, because the coverage point is already on the call.

- Send Term

  This is the same function as Send All Calls, except Send Term is for a TEG. Since a TEG cannot be in a coverage path, this function only applies to a directly called TEG.

- Call Forwarding All Calls

  Call Forwarding provides a temporary override of the redirection criteria. The call attempts to complete to the forwarded-to extension number before redirecting to coverage. If the principal's redirection criteria are met at the forwarded-to extension, then the call is redirected to the principal's coverage path.

Attendant-extended calls that redirect to Call Coverage use the type of redirection criteria (external or internal) applicable for the calling party. In other words, the call is treated as if no attendant is involved.

Call Coverage provides redirection of calls from the called principal or group to alternate answering positions when certain criteria are met. Yet the call is intended for the called principal or group. Certain provisions allow calls to direct to and/or be answered by the principal even though the redirection or overriding criteria are met. These provisions are:

- If no answering positions are available in the Coverage Path, the call rings the called voice terminal, if possible; otherwise, the calling party receives busy tone. This applies even if the Cover All Calls redirection criterion or the Send All Calls overriding criterion is active.

- Similarly, calls directed to a UCD or DDC group will queue, if queuing is available, when no group members are available to answer the call. The call remains in queue for a time interval equivalent to the Don't Answer Interval before routing according to the Coverage Path. If no points on the path are available, the call remains in queue. The worst case is when group queuing and the coverage points are both unavailable. In this case, the caller receives busy tone.

- If the redirection criterion is Active or Cover All Calls, a called principal can receive a redirection notification signal (a short burst of ringing) when the call routes to coverage. (Redirection Notification is optional on a per-terminal basis.) Note that in the Active, Cover All Calls, and Don't Answer cases, the principal could answer the call. Busy means no call appearances are available to answer the call. Redirected calls maintain an appearance on the called voice terminal, if possible. The call appearance status lamp flashes to indicate an incoming call before the call redirects. When the call does redirect, the status lamp lights steadily. The user can answer the call by pressing the call appearance button. If the call has already been answered, the principal is bridged onto the call. This provision is called Temporary Bridged Appearance.

- Priority Calling, Dial Intercom, and Automatic Intercom Calls always route directly to the principal's voice terminal until the calling party activates Go to Cover. These calls take precedence over the redirection criteria and seize the call appearance normally reserved for outgoing calls, if no other call appearances are available.

An internal calling party is informed that a call is redirecting to coverage by a single, short burst of ringing, called a Call Coverage tone. This tone is followed by an optional period of silence, called a Caller Response Interval. This interval allows the calling party time to decide what to do: hang up or activate Leave Word Calling, Automatic Callback, or Go to Cover. Activating Go to Cover cancels the remaining interval.

Covering User Options

For specific Call Coverage needs, the following options are available to voice terminal users:

- Consult

  Allows the covering user, by first pressing the Conference or Transfer button and then the Consult button, to call the principal (called party) for private consultation. These two actions place the calling party on hold and establish a connection between the principal and the covering user. If the principal wishes, the covering user can complete the conference and add the calling party to the conversation. Similarly, the call can be transferred to the principal. Consult calls use the Temporary Bridged Appearance maintained on the call, if there is one. If not, the Consult call seizes any idle call appearance. If there is no idle call appearance, the Consult call is denied.

- Coverage Callback (Implied Principal Addressing)

  Allows a covering user, by pressing the Cover Callback button, to leave a message for the principal to call the calling party. The calling party must be an internal caller. The principal receives no indication that the covering user handled the call.

  Alternatively, if the covering user presses the Leave Word Calling button, a "call me" message is left for the pincipal. The principal calls the covering user to get the message. This method is used when an external call is received or when an internal caller wants to leave a message but will not be available for a return call.

- Coverage Answer Group

  A Coverage Answer Group can have up to eight members. When a call is redirected to a Coverage Answer Group, all voice terminals in the group ring simultaneously. Anyone in the group can answer the call. A Coverage Answer Group member already handling a group call is rung when another call is redirected to that Coverage Answer Group. If a Coverage Answer Group member is also a member of another Coverage Answer Group, he or she can also receive calls for the other group. A second call directed to a Coverage Answer Group lights a Coverage Incoming Call Identification lamp.

- Coverage Incoming Call Identification (ICI)

  A Coverage ICI button can be assigned to multi-appearance voice terminal users without a display in a Coverage Answer Group.

  The Coverage ICI status lamp simply identifies a call incoming to that Coverage Answer Group. If a Coverage Answer Group is assigned to more than one Call Coverage path, the path number cannot be identified. Likewise, if a given path is assigned to more than one principal, the individual principals cannot be identified. To provide unique path and principal identification, the System Manager must establish a unique path for each principal and a unique Coverage Answer Group to be included in the path. A second coverage call takes control of the Coverage ICI lamp and does not return control to the previous call when the second call is released.

What Happens When a Call Goes to Coverage

When a call meets the redirection criteria of the principal, the call attempts to route to one of up to three points in the coverage path, beginning with point 1. If no coverage points are available, the call may revert to the called principal or group. If any point in the path is available, the call either rings the individual voice terminal or member of a group specified for that point or queues on the group. Once a call is ringing or queued at any point in a coverage path, the call never reverts to the called principal or group, or to the previous point. A call remains at a coverage point for a time equal to the Don't Answer Interval for Subsequent Redirection (1 to 9 ringing cycles [V1 and V2] or 1 to 99 ringing cycles [V3 and G1]). At the end of this time, the call attempts to route to any remaining points in the coverage path. If no other point is available to accept the call, the call will remain queued or continue ringing the current coverage point.

Typical Call Coverage Arrangements

Call Coverage is an extremely flexible feature and allows various combinations of coverage points. To illustrate the usefulness of Call Coverage, three typical coverage arrangements are given here as an example.

- Executive Coverage

  Provides a principal with call redirection to covering users having a close working relationship with the principal. Because of the status of the principal, personalized answering should be provided. Also, the principal may or may not choose to answer his or her own calls.

  A typical example of this form of coverage is when a principal's calls are redirected to a secretary. The secretary would be informed of the principal's daily schedule and other pertinent facts such as the importance of certain calls. The secretary could provide personalized answering by answering calls with the principal's name.

If the secretary is unavailable to answer the coverage call for the principal, the call redirects to a backup answering position. Personalized answering should also be provided at the backup position.

- Middle Manager Coverage

  Provides a group of principals with call redirection to one or more covering users (such as a secretary). The secretary should have some knowledge of the principal's daily schedule. A backup answering position should be provided in case the secretary is unavailable.

- General User Coverage

  Provides less-personal coverage for a broader spectrum of users. Covering users typically consist of a group or pooled answering arrangement. With this type of arrangement, coverage calls may be distributed among the members of the answering group.

As an example of how to provide a particular cover arrangement, the following provisions for the Executive Coverage arrangement are given.

- Determine if the secretary and backup position have a call display capability.
  - If so, Coverage Answer Groups are not required.
  - If not, establish a unique Coverage Answer Group for each one without a display. Specify only the applicable extension number. The Coverage Answer Group will contain only one member. Establish two groups, if required. Note that if the secretary and/or the backup answering position are in a Coverage Answer Group, each will receive only one redirected call for the executive at any given time. Calls do not ring a Coverage Answer Group member already busy on a call to the group. For frequently called executives, it is desirable that the secretary and possibly the backup answering position have a digital display capability.
- Establish a unique Call Coverage Path for the executive.
  - If the secretary will screen the calls, specify Cover All Calls as the redirection criteria.
  - If the executive will answer calls, specify Active, Busy, Don't Answer, Active/Don't Answer, or Busy/Don't Answer as desired.
  - Specify the secretary and the backup position [or the Coverage Answer Group(s) containing the secretary's and backup position's extension numbers] as the coverage points in the path.
- Optionally, specify a Send All Calls button on the executive's voice terminal. If someone else answers the executive's calls, the button is not needed.

- Specify a Send All Calls button and a Consult button on the secretary's voice terminal. Specify a Coverage ICI button if the secretary doesn't have a call display capability. Send All Calls is needed if the secretary will be unavailable for a period of time. Consult is needed to enable private consultation with the executive during an established call. Coverage ICI is needed to identify the call as a call to the executive rather than a personal call to the secretary.

- Specify a Consult button and a Coverage ICI button on the backup position's voice terminal for the same reasons these buttons were specified for the secretary.

Considerations

Call Coverage provides the means to redirect calls to alternate answering positions. The feature is versatile enough to permit suitable alternate answering arrangements for virtually every level of employee. Special functions, such as Send All Calls and Consult, accommodate the day-to-day variations that occur in an employee's work schedule. Call Coverage was designed on the premise that incoming calls are intended for the called party, but suitable alternatives must be available if the called party cannot, or does not wish to, answer his or her own calls.

The system allows for as many as 100 (V1) or 200 (V2, V3, or G1) Coverage Answer groups with up to 8 voice terminals in each group.

Up to 200 (V1), 400 (V2 or V3), or 600 (G1) coverage paths can be established. Each coverage path can have one, two, or three coverage points.

With V2, V3, or G1 systems, incoming tie trunk calls can be administered as either internal or external and are redirected to Call Coverage accordingly.

Interactions

The following features interact with the Call Coverage feature.

- Agent Call Handling (V3 or G1)

  Cover All Calls should not be assigned to agents with the Automatic Answer option. Any call (ACD or non-ACD) to an extension that has Automatic Answer enabled and has its coverage redirection criteria administered as Cover All Calls will not go to coverage, but will go to the called extension. Cover All Calls redirection criteria has no effect on an incoming call when a user is in the Auto-In mode.

- Attendant Display and Voice Terminal Display

  These features provide call identification for the covering user.

- Automatic Callback and Ringback Queuing

  Callback calls do not redirect to coverage. The caller can activate Automatic Callback when ringing, redirection notification signal, or busy signal is heard.

- Automatic Intercom, Dial Intercom, and Priority Calling

  Calls using these features will not redirect to coverage unless the caller presses the Go to Cover button.

- Bridged Call Appearance

  Coverage criteria for bridged call appearances is based entirely on the criteria of the primary extension associated with the bridged call appearance.

  If a voice terminal user has activated Send All Calls on its primary extension, incoming calls will still ring bridged call appearances of that extension as long as a Temporary Bridged Appearance of the call is maintained at the primary extension.

- Call Forwarding All Calls

  Call Forwarding provides a temporary override of the redirection criteria. Normally, calls forward instead of redirecting to coverage. However, if a forwarding extension number's redirection criteria are met at the designated (forwarded-to) extension number, the forwarded call is handled as if Call Forwarding has not been activated. When the forwarded call goes to coverage, however, a Temporary Bridged Appearance remains at the forwarded-to voice terminal until the call is answered and released.

  If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding is active at a voice terminal, incoming Priority Calling calls forward to the designated extension number.

  The Redirection Notification Signal applies to both Call Coverage and Call Forwarding.

  If an extension has both Send All Calls and Call Forwarding All Calls activated, calls to that extension that can immediately redirect to coverage will do so. However, other calls, such as priority calls, will forward to the designated extension.

  Activation of Send All Calls at the forwarded-to extension does not affect calls forwarded to that extension.

- Call Pickup

  Any call redirected to a covering user who is a member of a Call Pickup group can be answered by other members of the Call Pickup Group

- Centralized Attendant Service (CAS)

  If an incoming CAS call is directed to a hunt group, the call will not redirect to the hunt group's coverage path.

- Class of Restriction and Controlled Restrictions

  Users who may normally be restricted from receiving calls can still receive calls directed to them via Call Coverage.

- Hold

  If a covering user puts a call on hold, and the principal picks up on the call, the coverage appearance may or may not be dropped, depending on administration.

- Leave Word Calling

  Call Coverage can be used with or without Leave Word Calling. However, the two features complement each other. When a covering user activates Leave Word Calling during a coverage call, a message is left for the principal to call the covering user. When a covering user activates Coverage Callback during a coverage call, a message is left for the principal to call the internal caller.

- Night Service—Night Station Service (V1)

  A call routed to the DID LDN (Direct Inward Dialing Listed Directory Number) night extension via Night Station Service does not go to coverage, even if the coverage criteria of the DID LDN night extension is met.

  Calls routed to the attendant via Call Coverage or Call Forwarding do not route to the DID LDN night extension.

- Temporary Bridged Appearance

  Calls redirected to coverage maintain an appearance on the called voice terminal if a call appearance is available to handle the call. The called party can bridge onto the call at any time. With V3 and G1, the system can be administered to allow a temporary bridged appearance of the call to either remain at or be removed from the covering voice terminal after the principal bridges onto the call. With V1 and V2, a temporary bridged appearance automatically remains at the covering voice terminal after the principal bridges onto the call.

  Consult calls use the Temporary Bridged Appearance maintained on the call. At the conclusion of a consult call, the bridged appearance is no longer maintained. If the principal chooses not to talk with the calling party, the principal cannot bridge onto the call later.

  If a call has, or has had, a Temporary Bridged Appearance, is conferenced or transferred, and redirects to coverage again, a Temporary Bridged Appearance is not maintained at the conferenced-to or transferred-to extension.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  If a user has a Auxiliary Work button, and activates or deactivates Send All Calls, the Auxiliary Work function associated with DDC or UCD is activated or deactivated simultaneously.

  If a user has no Auxiliary Work button, activating or deactivating Send All Calls still makes the user available or unavailable for DDC or UCD calls, but Auxiliary Work is not activated or deactivated. The Auxiliary Work activate or deactivate code and the DDC or UCD extension must be dialed to activate the Auxiliary Work function.

  Activating or deactivating the Auxiliary Work function does not activate or deactivate Send All Calls.

Administration

Call Coverage is administered by the System Manager. The following items require administration:

- Coverage Paths

  The same coverage path can be used for as many voice terminal users as desired.

- Coverage Path Lists (per voice terminal) (V3 or G1)

- Cover Answer Groups

- Don't Answer Interval and Don't Answer Interval for Subsequent Redirection

  The Don't Answer Interval specifies the number of ringing cycles heard at the principal's terminal before the call is redirected to the first coverage point. This interval is recommended to be two rings, but can be administered from one to nine rings. All principals with the same coverage path are assigned the same Don't Answer Interval.

  The Don't Answer Interval for Subsequent Redirection specifies the number of rings at a covering terminal before the call attempts to redirect to the next coverage point. This interval is recommended to be two rings, but can be administered from one to nine rings. This interval is administered as a system parameter.

- Caller Response Interval

  This interval can be from 0 to 10 seconds. If 0 is administered, the Caller Response interval does not apply.

- Redirection Notification Signal

  This signal is administered on a per-terminal basis. If administered, the signal also applies to forwarded calls. With Call Coverage, the signal indicates to the caller that the call is being redirected to coverage because of the Active or Cover All Calls redirection criteria.

- Feature Access Codes for Activation and Deactivation of Send All Calls
- Whether incoming tie trunk calls are treated as internal or external calls (V3 or G1)
- Whether or not a temporary bridged appearance is maintained by the covering user after the principal bridges onto the call (Keep Held SBA at Coverage Point field on Feature-Related System Parameters screen form) (V3 or G1).
- Buttons on Multi-Appearance Voice Terminals, as desired:
  - Consult
  - Coverage Callback
  - Go to Cover
  - Coverage ICI
  - Send All Calls.

Hardware and Software Requirements

No additional hardware or software is required.

Call Forwarding All Calls (V1)

Description

Allows all calls to an extension number to be forwarded to a selected internal extension number or the attendant. This feature is activated or deactivated by dial access code.

Call Forwarding All Calls can be activated or deactivated by voice terminal users.

Voice terminal users activate Call Forwarding All Calls by dialing a feature access code followed by the designated (forwarded-to) extension number. The feature is deactivated by dialing a different feature access code.

Calls can be forwarded only once. Calls forwarded to a designated (forwarded-to) extension number do not forward again. These calls ring the designated extension number, if possible; redirect if the forwarding party's Call Coverage criteria are met; or return busy tone to the calling party.

When Call Forwarding All Calls is activated at a voice terminal and a call for that terminal is forwarded, the terminal can (if administered to do so) receive a redirection notification signal that a call is being forwarded.

Call Forwarding All Calls can also be activated or deactivated at data terminals.

Considerations

With Call Forwarding All Calls, voice terminal users can have their incoming calls forwarded to another extension number. This allows voice terminal users to have their calls follow them when they know they will be temporarily near another extension. There is no limit to the number of calls that can be forwarded simultaneously.

Calls to attendants cannot be forwarded. However, calls can be forwarded to the attendant group.

Interactions

The following features interact with the Call Forwarding All Calls feature.

- Automatic Callback and Ringback Queuing

  Callback calls do not forward.

- Bridged Call Appearance

  Call Forwarding All Calls can be activated or canceled for the primary extension number from a bridged call appearance of that number. When activated, calls to the primary extension number do not terminate at the bridged call appearances. Bridged call appearances do not receive redirection notification when a call to the primary extension is forwarded.

- Call Coverage

  If the principal's (forwarding extension number) redirection criteria are met at the designated (forwarded-to) extension number, the forwarded call is redirected to the principal's coverage path and the designated extension gets a temporary bridged appearance until the call is answered or the caller hangs up.

  If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding is active at a voice terminal, incoming Priority Calls forward to the designated extension number and all other calls redirect according to the Call Coverage path.

  When a covering user has activated Call Forwarding, a coverage redirected call does not forward to the designated extension number. Instead, the call is redirected to the next point in the principal's coverage path, if available. If no other coverage point is available, the call will remain at the principal's voice terminal.

- Class of Restriction and Controlled Restrictions

  If a call would normally be restricted between the forwarding and forwarded-to extensions, Call Forwarding activation is denied. However, if restrictions are assigned after Call Forwarding is activated, any termination restrictions are ignored.

- Code Calling Access and Call Park

  Calls using these features override Call Forwarding. Code Calling Access and Call Park calls complete to the called extension number even if Call Forwarding is active.

- Direct Inward Dialing (DID)

  If an incoming DID call is forwarded to another extension and answered by the forwarded-to extension, any other calls to the same DID extension within the next 30 seconds will receive busy tone.

- Hot Line Service and Manual Originating Line Service

  Voice terminals assigned these features cannot activate Call Forwarding. However, calls can be forwarded to these terminals.

- Night Service—Night Station Service

  A call routed to the Direct Inward Dialing Listed Directory Number (DID LDN) night extension via Night Station Service does not forward to another extension, even if Call Forwarding All Calls has been activated at the DID LDN night extension.

- Personal Central Office Line (PCOL)

  PCOL calls are not forwarded.

- Send All Calls

  If an extension has both Send All Calls and Call Forwarding All Calls activated, calls to that extension that can immediately redirect to coverage will do so. However, other calls, such as priority calls, will forward to the designated extension.

  Activation of Send All Calls at the forwarded-to extension does not affect calls forwarded to that extension.

Administration

Call Forwarding All Calls is assigned on a per-extension number basis by the Class of Service. The following items require administration by the System Manager:

- Voice Terminals
  - Class of Service
  - Redirection Notification
- Feature Access Codes for Activation and Deactivation of Call Forwarding All Calls.

Hardware and Software Requirements

No additional hardware or software is required.

Call Forwarding All Calls (V2, V3, or G1)

Description

Allows all calls to an extension number to be forwarded to a selected internal extension number, external (off-premises) number, the attendant group, or a specific attendant. This feature is activated or deactivated by dial access code or by a Call Forwarding button.

Call Forwarding All Calls can be activated or deactivated by voice terminal users and data terminal users. Also, an attendant or voice terminal user with console permission (V3 or G1) can activate or deactivate the feature for a particular extension number, Terminating Extension Group, Direct Department Calling group, Uniform Call Distribution group, or Automatic Call Distribution (ACD) split (V3).

Voice terminal users activate Call Forwarding All Calls by dialing a feature access code or pressing a Call Forwarding button, and then dialing the designated (forwarded-to) number. The feature is deactivated by dialing a different feature access code or pressing the Call Forwarding button again.

An attendant activates Call Forwarding All Calls by dialing a feature access code, followed by the forwarding extension number plus the forwarded-to number. The attendant deactivates the feature by dialing a different access code, followed by the extension number for which the feature is to be canceled. The attendant cannot have a Call Forwarding button assigned to the console.

A voice terminal user with console permission (V3 or G1) activates Call Forwarding All Calls for another user by dialing a feature access code, followed by the forwarding extension number plus the forwarded-to number. The attendant or voice terminal user with console permission (V3 or G1) deactivates the feature for another user by dialing a different access code, followed by the extension number for which the feature is to be canceled. A voice terminal user with console permission (V3 or G1) can also activate Call Forwarding All Calls for himself or herself by dialing the feature access code or pressing the Call Forwarding button.

When a Call Forwarding button is used to activate the feature, the status lamp associated with the button remains lighted until the feature is deactivated.

Calls can be forwarded only once. Calls forwarded to a designated (forwarded-to) number do not forward again. These calls ring the designated number, if possible; redirect if the forwarding party's Call Coverage criteria are met; or return busy tone to the calling party.

When Call Forwarding All Calls is activated at a voice terminal and a call for that terminal is forwarded, the terminal can (if administered to do so) receive a redirection notification signal that a call is being forwarded.

Considerations

With Call Forwarding All Calls, users can have their incoming calls forwarded to another extension number. This allows users to have their calls follow them when they know they temporarily will be near another extension. A user can also forward calls to an outside number when temporarily at an off-premises location. There is no maximum number of calls that can be forwarded simultaneously.

For Terminating Extension Groups, Uniform Call Distribution groups, and Direct Department Calling groups, Call Forwarding All Calls can only be activated by the attendant or voice terminal user with console permission (V3 or G1).

If an incoming call on a CO trunk is forwarded to an external extension and answered by the forwarded-to extension, any other calls to the same extension within the next 30 seconds will receive busy tone or redirect to coverage, if Send All Calls is assigned.

When a call is forwarded to an off-premises location, the forwarding-to number can have a maximum of 16 digits.

Calls to attendants cannot be forwarded. However, calls can be forwarded to the attendant group.

A voice terminal user with console permission cannot activate Call Forwarding All Calls for a data module.

If a user attempts to make a call to an extension that he or she is restricted from calling, and the called extension has activated Call Forwarding All Calls, the call will not complete.

A user cannot forward calls to an extension that he or she is normally restricted from calling.

Interactions

The following features interact with the Call Forwarding All Calls feature.

- Automatic Callback and Ringback Queuing

  Automatic Callback cannot be activated toward a voice terminal that has Call Forwarding activated. However, with V2, if Automatic Callback was activated before the called voice terminal user activated Call Forwarding, the callback call attempt is made toward the called party and is not redirected. With V3 and G1, if Automatic Callback was activated before the called voice terminal user activated Call Forwarding, the callback call attempt is redirected to the forwarded-to party.

- Call Coverage

  If the principal's (forwarding extension number) redirection criteria are met at the designated (forwarded-to) extension number, the forwarded call is redirected to the principal's coverage path and the designated extension gets a temporary bridged appearance until the call is answered or the caller hangs up.

If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding is active at a voice terminal, incoming Priority Calls forward to the designated extension number and all other calls redirect according to the Call Coverage path.

When a covering user has activated Call Forwarding, a coverage redirected call does not forward to the designated extension number. Instead, the call is redirected to the next point in the principal's coverage path, if available. If no other coverage point is available, the call will remain at the principal's voice terminal.

- Code Calling Access and Call Park

  Calls using these features override Call Forwarding. Code Calling Access and Call Park calls complete to the called extension number even if Call Forwarding is active.

- Direct Inward Dialing (DID)

  If an incoming DID call is forwarded to another extension and answered by the forwarded-to extension, any other calls to the same DID extension within the next 30 seconds will receive busy tone or redirect to coverage, if assigned.

- Hot Line Service and Manual Originating Line Service

  Voice terminals assigned these features cannot activate Call Forwarding. However, calls can be forwarded to these terminals.

- Interflow (V3 or G1)

  The Interflow feature allows ACD calls to be redirected from one split to a split on another switch or to another external location. This is accomplished by forwarding calls that are directed to the split extension to an off-premises location via the Call Forwarding All Calls feature. For details on the Interflow feature, see the Intraflow and Interflow feature description elsewhere in this chapter.

- Personal Central Office Line (PCOL)

  PCOL calls cannot be forwarded.

- Priority Calling

  Priority calls cannot be forwarded to an off-premises extension.

- Send All Calls

  If an extension has both Send All Calls and Call Forwarding All Calls activated, calls to that extension that can immediately redirect to coverage will do so. However, other calls, such as priority calls, will forward to the designated extension.

  Activation of Send All Calls at the forwarded-to extension does not affect calls forwarded to that extension.

- Station Message Detail Recording (SMDR)

  When a call is forwarded to an off-premises number, the call is recorded in SMDR records as a call from the forwarding station.

- SMDR Account Code Dialing

  If forced entry of account codes is required, calls cannot be forwarded to off-premises destinations.

Administration

Call Forwarding All Calls is assigned on a per-extension number basis by the Class of Service. The following items require administration by the System Manager:

- Voice Terminals
  - Class of Service
  - Call Forwarding Buttons
  - Redirection Notification
- Feature Access Codes for Activation and Deactivation of Call Forwarding All Calls.

Hardware and Software Requirements

No additional hardware or software is required.

Call Park

Description

Allows users to put a call on hold and then retrieve the call from any other voice terminal within the system.

When a voice terminal user, active on a call, needs to go to another location for information, the call can be placed in Call Park and retrieved at the other location.

Conference calls can also be placed in Call Park.

Call Park can be activated by any of the following:

- A single-line voice terminal user—Flash the switchhook, dial the Call Park access code, and hang up. The call is parked on the user's extension number.
- A multi-appearance voice terminal user—Press the Transfer or Conference button, dial the Call Park access code, and press the Transfer or Conference button again, or simply press the Call Park button (if assigned). The call is parked on the user’s extension number.
- An attendant—Press Start, dial the Call Park access code followed by any extension number, and press Release. The call will be parked on the number dialed. An attendant can use the Direct Extension Selection With Busy Lamp Field feature instead of dialing the extension number.
- The system—When Code Calling Access is used, the call is automatically parked on the paged party’s extension number.

Calls are retrieved by dialing the Call Park Answer Back access code and the extension number where the call is parked, or by pressing the same Call Park button used to park the call.

A systemwide expiration interval can be set for parked calls. When the interval expires, the parked call will redirect to an attendant console (or the parking user if administered to do so [G1 only]) and will no longer be parked on the extension number. However, if the parked call has already been retrieved when this interval expires, the call will not redirect. If two parties are connected on a parked call, a third party can also answer the call before the interval expires, creating a 3-way conference. With V2, V3, or G1, if no attendant (this includes Centralized Attendant Service, local attendants, and Individual Attendant Access) or night service extension is administered, and if Night Service—Trunk Answer From any Station is not administered, the expiration interval is ignored and the call will remain parked.

The attendant console group can have up to ten common shared extension numbers used exclusively for Call Park. These extension numbers are not assigned to a voice terminal, but are stored in system translations and used to park a call. These extension numbers are particularly useful when one party is paged at the request of another party. The calling party is parked and the extension number is announced. Common shared extensions should be assigned to the optional selector console in the 00 through 09 block (bottom row) in any hundreds group that the attendant can easily identify. The lamp associated with the extension number will identify call parked or no call parked (instead of active or idle status).

Considerations

Call Park can be used whenever a voice terminal user who is on a call needs to go elsewhere and obtain information, and wishes to complete the call at another extension. Call Park also allows users to answer a call from any station after being paged by a voice terminal user or an attendant.

Only one call per extension number can be parked at a time, even if the extension number has multiple appearances. However, a conference call with five parties can be parked. The sixth conferee will be the retrieving party.

Calls cannot be parked on a group extension number. If a group member places a call in Call Park, the call will be parked on the member extension number. Group members include the following:

- A Coverage Answer Group member
- A Direct Department Calling group member
- A Terminating Extension Group member
- A Trunk Answer From Any Station answering user
- A Uniform Call Distribution group member.

Interactions

The following features interact with the Call Park feature.

- Abbreviated Dialing

  An Abbreviated Dialing button can be assigned so that parking calls or retrieving parked calls can be done by pressing a button, instead of using the buttons and access codes normally used. This operation reduces the number of steps required to park a call or retrieve parked calls.

- Bridged Call Appearance

  If a user, active on a bridged call appearance, activates Call Park, the call is parked on the primary extension associated with the bridged call appearance.

- Data Privacy and Data Restriction

  These features are automatically deactivated when a call is parked.

- Loudspeaker Paging Access

  Calls to paging zones cannot be parked.

- Loudspeaker Paging Access—Deluxe (G1)

  If the system is administered to have Deluxe Paging, parked calls are redirected to the parking user when the Call Park Timeout Interval expires.

- Music-on-Hold

  Music can be provided to one party on a parked call. However, music cannot be provided to a multiple-party (conference) parked call.

  If Music-on-Hold is provided, the user activating Call Park also hears music after the call is parked and confirmation tone is heard.

- Remote Access

  A Remote Access caller cannot park a call. However, the Code Calling Access feature, an answering attendant, or a voice terminal user can park an incoming Remote Access call.

Administration

Call Park is administered on a per-system basis by the System Manager. The following items require administration:

- Call Park access code
- Answer Back access code
- Call Park Timeout Interval (from 1 to 90 minutes in intervals of 5 seconds)
- Call Park button (multi-appearance voice terminals only). A Call Park button should have a lamp so that the voice terminal user can tell when a call is parked on his or her extension.
- Common shared extension numbers for the attendant group (from 1 to 10).
- With G1, the Deluxe Paging and Call Park Timeout to Originator field on the Feature Related System Parameters form must be administered as "yes" to have parked calls return to the parking user (the originator) when the Call Park Timeout interval expires. If this field is administered as "no", parked calls go to the attendant when the Call Park Timeout interval expires.

Hardware and Software Requirements

No additional hardware or software is required.

Call Pickup

Description

Allows voice terminal users to answer calls to other extension numbers within the user's specified Call Pickup group.

Call Pickup groups are established so that when one member of a group is away other members of the group can answer that member's calls. A Call Pickup group usually consists of users who are located in the same area or have similar functions.

When a member of a Call Pickup group is away and receives an incoming call, any member of the Call Pickup group can answer the call. A member simply goes off-hook and dials the Call Pickup access code or presses a Call Pickup button. That group member is then connected to the calling party.

A Temporary Bridged Appearance is maintained at the called voice terminal. This allows the called party to bridge onto the call after it has been picked up by another member of the Call Pickup group.

Considerations

With Call Pickup, users do not have to leave their own voice terminal in order to answer a call at a nearby voice terminal. Instead, a user simply lifts the handset and dials an access code or presses a Call Pickup button. This allows unanswered calls to be handled more quickly and efficiently.

Up to 200 (V1), 400 (V2 or V3), or 800 (G1) Call Pickup groups can be established. Each group can have up to 25 (V1) or 50 (V2, V3, or G1) members. However, a voice terminal can be a member of only one Call Pickup group.

A user cannot activate Call Pickup while already active on a call (V1 only).

Interactions

The following features interact with the Call Pickup feature.

- Automatic Callback and Ringback Queuing

  Callback calls cannot be answered by Call Pickup group members.

- Bridged Call Appearance

  Activating Call Pickup while on a bridged call appearance will pick up a call in the Call Pickup group of the bridged extension.

  If a voice terminal receives ringing on a bridged call appearance, the incoming call can be picked up by members of that voice terminal's Call Pickup group. This causes all bridged call appearances to be dropped. The call is parked on the primary extension number of the answering voice terminal.

- Call Forwarding All Calls

  A forwarded call cannot be picked up at the forwarded-to voice terminal unless the forwarding and forwarded-to voice terminals are in the same pickup group.

- Call Waiting Termination

  A Call Waiting call cannot be picked up by a Call Pickup group member.

- Hold

  A call, picked up and placed on hold at an extension, remains on that extension, even if the called party answers the call.

- Hot Line Service and Manual Originating Line Service

  Voice terminals assigned these features can be Call Pickup group members so their incoming calls can be answered. However, voice terminal users with these features assigned cannot answer calls for other group members.

- Intercom—Automatic

  Call Pickup can be used to answer an Automatic Intercom call.

Administration

Call Pickup is administered by the System Manager. The following items require administration:

- Call Pickup group number
- Members (extension numbers) of each Call Pickup group
- Call Pickup access code
- Call Pickup buttons.

Hardware and Software Requirements

No additional hardware or software is required.

Call Waiting Termination

Description

Provides for calls to busy single-line voice terminals to wait and sends a distinctive call waiting tone to the called party.

The called party hears one quick burst of tone when a call from another voice terminal user is waiting, two quick bursts of tone when an attendant-handled or an outside call is waiting, or three quick bursts of tone when a Priority Call is waiting. The called party hangs up on the current call and immediately receives ringing from the waiting call.

With V1 systems, the call in progress at the voice terminal cannot be placed on hold. It must be terminated. With V2, V3, or G1 systems, the call in progress at the voice terminal can be placed on hold in order to answer the waiting call. After answering the waiting call, the voice terminal user can return to the held call or toggle back and forth between the two calls. The single-line voice terminal user can only be connected to one call at a time.

The calling party hears special audible ringback tone if the call is allowed to wait. If Call Waiting is denied, the calling party hears busy tone. Only one call can wait at a time.

The burst(s) of tone heard by the called voice terminal user is not heard by other parties on the call.

An internal caller can activate Leave Word Calling or Automatic Callback after Call Waiting has been activated.

A Priority Call and an attendant-handled call can wait for the voice terminal to become idle even if the Call Waiting Termination feature is not assigned.

Calls to a Direct Department Calling or Uniform Call Distribution group voice terminal cannot wait. However, such calls can enter the group queue (if provided) unless the queue is full.

Considerations

With Call Waiting Termination, the party who calls a busy single-line voice terminal does not have to hang up and try the call again later. Instead, the call will wait at the called voice terminal until the called party hangs up on the current call.

Call Waiting Termination applies only to busy single-line voice terminals. Calls to multi-appearance voice terminals are routed to an idle call appearance and do not wait.

An analog voice terminal user must place the active call on "soft" hold (see Hold feature) and dial the Answer Hold-Unhold feature access code to answer the waiting call. The soft held call at that time becomes a "hard" held call.

Interactions

Call Waiting is denied when the following features are activated at the single-line voice terminal:

- Automatic Callback (to or from the voice terminal)
- Data Privacy
- Data Restriction
- Another Call Waiting Call.

A Call Waiting call cannot be picked up by a Call Pickup group member.

Administration

Call Waiting Termination is assigned on a per-voice terminal basis by the System Manager. The only administration required is the assignment of Call Waiting Termination to the desired voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Centralized Attendant Service (V2, V3, or G1)

Description

Allows services performed by attendants in a private network of switching systems to be concentrated at a central, or main, location. Each branch in a Centralized Attendant Service (CAS) has its own listed directory number (LDN) or other type of access from the public network. Incoming trunk calls to the branch, as well as attendant-seeking voice terminal calls, are routed to the centralized attendants over release link trunks (RLTs).

The CAS attendants are located at the main location. The main location can be a System 85, a DIMENSION PBX, a System 75 (V3), a DEFINITY Communications System Generic 1, or a DEFINITY Communications System Generic 2.1.

The CAS main PBX operates independently of the CAS branch PBXs. The operation for CAS main PBX traffic is identical to a stand-alone PBX.

Each branch in a network with CAS is connected to the main by way of Release Link Trunks (RLTs). These trunks serve three basic functions:

- Paths for sending incoming attendant-seeking trunk calls at the branch to the centralized attendant to be processed and extended back to their destinations at the branch (both parts of a call use the same trunk)
- Paths for returning timed-out waiting and held calls from the branch to the main
- Paths for routing calls from voice terminals in the branch to the centralized attendant at the main.

RLTs can be seized only from the branch switch and are used only for CAS calls and CAS signaling. After processing by a centralized attendant, CAS calls are extended back over the same RLT to, for example, the requested extension number or outgoing trunk. The RLT is then dropped and becomes available for other calls toward the centralized attendants.

Two queues are associated with CAS calls, one at the main and one at the branch. When idle RLTs are available from the branch to the main, RLTs are seized and CAS calls are placed in the attendant queue at the main along with other attendant-seeking calls. If all RLTs are in use, however, the branch switch puts calls to the centralized attendant in a RLT trunk queue at the branch. The length of the RLT trunk queue can vary from 1 to 100 and is set during administration of the RLT group.

Backup service provides for all CAS calls to be sent to a backup extension in the local branch if all RLTs are maintenance busy or out of service, or if a Backup button is pressed while not lighted. The backup extension can be assigned a Backup button and associated status lamp to activate the feature and provide notification that backup service is in effect. The status lamp remains lighted as long as backup service is in effect. If the Backup button is pressed while the status lamp is lighted, calls will not be sent to the backup extension unless all RLTs are maintenance busy or out of service.

A CAS call from a branch can be put on Remote Hold by the CAS attendant. The branch holds the call and drops the RLT. After a time-out (same as the timed reminder for an attendant-held call), the branch automatically attempts to route the call back to the CAS attendant. It is possible for the returning call to be queued for the RLT. It is recommended that when CAS attendants use Remote Hold when they have to put a call on hold. This keeps RLTs from being tied up unnecessarily.

The branch in a CAS network generates call identification tones and transmits them to the CAS attendant by way of the RLT. These tones indicate to the attendant the type of call coming from the branch or the status of a call extended to or held at the branch. The attendant hears these tones in the console handset prior to actually being connected to the caller.

- Incoming trunk call: 480 Hz (100 ms), 440 Hz (100 ms), 480 Hz (100 ms) in sequence; heard immediately after attendant lifts handset
- Call from branch terminal to attendant or transferred by branch terminal to attendant: 440 Hz (100 ms), silence (100 ms), 440 Hz (100 ms) in sequence; heard immediately after attendant lifts handset
- Call extended to idle station or recall on don't answer: ringback tone for 300 ms followed by connection to normal ringing cycle
- Call extended to busy terminal—automatically waiting or recall on attendant call waiting: 440 Hz (100 ms)
- Call extended to busy terminal—waiting denied or not provided: busy tone
- Remote hold or remote hold recall: a series of four to six cycles of 440 Hz (50 ms), silence (50 ms)
- Recall on don't answer: 300 ms burst of ringback, then connection to normal ringback at any point in its cycle
- Recall from a call on remote hold: a series of four to six cycles of 440 Hz for 50 ms silence for 50 ms
- Recall from a call waiting at a single-line terminal: 100 ms burst of 440 Hz.

The centralized attendant at the main has access, through RLTs, to all outgoing trunk facilities at the branches in a CAS network. The attendant can extend an incoming LDN call to an outgoing trunk at a branch by dialing the access code and allowing the caller to dial the rest of the number or by dialing the complete outgoing number. Automatic Route Selection (ARS) is also available to the attendant in establishing outgoing calls.

Calls extended to busy single-line voice terminals at the branch wait automatically. When any waiting extended call is not answered within an administered interval, the branch switch attempts to return the call to the centralized attendant. The Call Waiting feature does not apply to multi-appearance terminals; if no appearances are available, busy tone is sent to the attendant, who tells the caller that the line is busy.

Calls from voice terminals at the branch to a centralized attendant are also routed over RLTs seized by the branch switch. A branch caller reaches the attendant simply by dialing the attendant code, "0." The conversation between the branch caller and the attendant ties up the seized RLT, but calls of this type are usually short.

Considerations

CAS reduces the number of attendants required at each branch location. More efficient call handling is provided by letting one group of centralized attendants handle calls for the individual branches. For example, a chain of department stores can have a centralized attendant location at the main store. The centralized attendant can then handle calls for the individual stores.

In a CAS network, System 75s and DEFINITY Generic 1s can function as branches or as the main (V3 and G1 only); the main location, where the centralized attendants reside, must be a system capable of providing attendant concentration.

A system can be a branch to only one main location.

A system serving as a main location can have 99 CAS branches, 200 RLTs, and a total of 7 consoles (including a night console).

A network with CAS can also be a Distributed Communications System, but this association is not required.

A branch can have a local attendant. Access to the local attendant must be by way of an individual attendant extension. Incoming trunk calls in a CAS network may bypass local attendants but can be routed back to them by the centralized attendant.

The CAS branch calls are terminated on the CAS main PBX based on the incoming RLT trunk group day-destination or the night-service destination. A CAS call may also be answered by the Night Service—Trunk Answer Any Station feature. With these considerations, an attendant console will not always be answering/extending the incoming CAS calls. If a non-attendant answers a CAS call, the call may be extended back to the branch through use of the FLASH button on a multi-appearance voice terminal or a switchhook flash on a single-line voice terminal. The branch reaction to Flash Signals and the branch application of tones is the same whether an attendant or non-attendant answers/extends the call.

If an extended call returns to the CAS main attendant because it was unanswered, the called party at the branch is not dropped but continues to be alerted until the caller is released. This allows the attendant to talk to the caller, then extend the call again, if the caller wishes, without redialing the number.

If the recall time-out occurs for an extended CAS call which has gone to Coverage and no one has answered, then the branch leaves the extended-to party ringing and drops coverage from the call.

When an analog station's call goes to coverage, the analog station is dropped from that call. This is the exception to the branch leaving the extended-to party ringing. Therefore, if the CAS main attendant extends a call to an analog station and that call goes to coverage and later returns to the CAS main attendant, then this call is treated as an incoming LDN call and the attendant must re-extend the call, if requested by the user.

On an incoming CAS call to the main attendant, the Name field from the trunk group form for that RLT is displayed to the attendant. It is recommended that the Name field in the Trunk Group form provide the attendant with CAS branch identification information.

If the Music-on-Hold feature is provided in a CAS branch, it will be applied to two stages of LDN calls. During the brief period in which the attendant is extending a call, the caller (who is on "soft hold" at the branch) receives music. Music-on-Hold is also connected to callers on Remote Hold.

Interactions

The following features interact with the Centralized Attendant Service feature.

- Attendant Control of Trunk Group Access

  If a local attendant has control of the outgoing RLT trunk group, when CAS is in effect, new attendant-seeking calls are routed to the local attendant.

- Abbreviated Dialing

  The main attendant may use an Abbreviated Dialing button to extend CAS calls after obtaining branch dial tone.

- Attendant Auto-Manual Splitting

  The SPLIT lamp and button do not function on CAS main calls. Therefore, attendant conference does not function on CAS calls.

- Busy Indicator Buttons

  Busy Indicators can be used to identify incoming calls over an RLT. Busy Indicators may also be used to dial after the attendant has started to extend the call.

- Call Coverage

  Calls can be redirected to a centralized attendant by Call Coverage. Calls to a CAS backup extension for backup service should not be redirected via Send All Calls to the backup extension's coverage path.

- Call Forwarding

  Calls to a CAS backup extension should not be forwarded and do not terminate at the backup extension.

- Call Park

  If a CAS Attendant parks a call and the call returns to the CAS attendant after the Call Park expiration interval, the CAS attendant hears incoming trunk call notification.

- DXS and DTGS Buttons

  DXS and DTGS buttons at the main attendant console can be used with CAS operation. However, when a DXS button is used to make a CAS call, it takes a few seconds before the attendant hears ringback tone.

- DCS Operation

  If the RLT trunk group is administered as a DCS trunk, the following interaction applies: On an incoming CAS call to the attendant, the DCS message is displayed instead of the name of the incoming RLT trunk group. Upon answering the call, the attendant hears the call identification tones. Receipt of these tones indicates to the attendant that the call is a CAS call. In this situation, a Trunk-Name button may be used to obtain the name of the RLT trunk group.

- Emergency Access to the Attendant

  CAS Branch Emergency Access calls generated by Feature Access Code or Off-hook Alert are routed to the branch's local attendant group. If there is no attendant in the branch PBX, the emergency call will be routed to the branch's administered Emergency Access Redirection Extension. When the branch PBX is in CAS Backup Service, the emergency calls are routed to the backup station and the call is treated as a normal call.

- Hunt Groups

  If an incoming CAS call is directed to a hunt group, the call will not redirect to the hunt group's coverage path.

- Night Service—Night Console Service

  When the CAS main enters night service, CAS calls terminate at the CAS main night service destination. Calls do not go to CAS attendants when Night Service has been activated at the branch.

- Night Service—Trunk Answer From Any Station

  In a multi-switch DCS environment with CAS, the result of transferring incoming trunk calls via the Night Service Extension or the Trunk Answer From Any Station feature varies depending on the home switch of the transferred-to station, the home switch of the connected trunk, and the type of night service function chosen (Night Service Extension, Trunk Answer From Any Station, or both).

- Non-Attendant Console Handling of CAS Calls

  The CAS branch calls are terminated on the CAS main PBX based on the incoming RLT trunk group day destination or the night service destination. A CAS call may also be answered by the Night Service—Trunk Answer From Any Station feature.

  Normally, a non-attendant extends a CAS call by using the Flash button. However, if the non-attendant does not have a Flash button, the call can be extended as follows:

  — Multi-appearance voice terminal users can extend a CAS call by pressing the Conference or Transfer button and then dialing the extension of the party the call is being extended to. To complete the call, the user must then drop the call. To drop the extended-to party, the user must press the Conference or Transfer button again.

  — Single-line voice terminal users can extend a CAS call by flashing the switchhook and then dialing the extension of the party the call is being extended to. To complete the call, the user must then drop the call. To drop the extended-to party, the user must flash the switchhook again.

- Non-Attendant Console Extends Call

  To extend a CAS call back over the same RLT, the non-attendant presses the Flash button or activates switchhook flash (depending on terminal type), which sends the flash signal over the RLT. The branch PBX allows the call to be extended. Analog voice must be assigned permission to use switchhook flash.

- Non-Attendant Console Releases Call

  The non-attendant can drop the RLT by going on-hook, using the DISCONNECT or DROP button, or by selecting another call appearance.

- Non-Attendant Console Holds Call

  A multi-function non-attendant may hold a CAS call by pressing the hold button. An analog non-attendant may not hold a CAS call.

- Non-Attendant—Display Trunk Name

  If the non-attendant with a display presses the Trunk-Name button while active on a trunk call, then the PBX displays the name field from the trunk group form.

- Station Message Detail Recording (SMDR)

  If the CAS main RLT trunk has the SMDR option selected, SMDR records will be generated for the incoming CAS calls.

- Timed Reminder

  The timer value used for recalling held calls at the attendant console is a parameter that can be set on the console form.

If an attendant at the main location transfers a call from a branch location to an extension at the main location, the timed reminder does not apply and the call will not return to the attendant if unanswered.

- Trunk-Name Button

  The Trunk-Name button can be used when an outgoing call has been made over a trunk which has been administered to have no outgoing display.

Administration

CAS is administered by the System Manager. The following items require administration:

- Access to CAS (branch, main, or none)
- Branch attendant individual extension number
- RLT group (outgoing for branch) (incoming for main—V3 and G1)
- RLT group queue length (this must be greater than 0)
- CAS backup extension
- CAS Backup buttons (used to activate/deactivate CAS Backup)
- Extension permitted to put system into night service if the system has no local attendant
- Recall time-out values
- Remote hold access code
- Trunk-Name button, if DCS is provided
- Flash button for a multi-appearance voice terminal if a non-attendant will answer CAS calls.

Hardware and Software Requirements

Requires a TN760B Tie Trunk circuit pack. The TN760B will serve all other tie trunk applications in addition to CAS. As an alternative, the TN722 DS1 Tie Trunk circuit pack can be used for the release link trunks of the CAS network.

CAS software is required.

Class of Restriction

Description

Defines up to 64 different classes of call origination and termination privileges. Systems may have only a single Class of Restriction (COR), one with no restrictions, or may have as many CORs (up to 64) as necessary to effect the desired restrictions.

A COR is assigned to each of the following:

- Trunk group
- Voice terminal
- Data module
- Loudspeaker Paging Access zone
- Code Calling Access zone
- Remote Access barrier code
- Attendant consoles (as a group)
- Individual Attendant Consoles
- Terminating Extension Group
- Uniform Call Distribution group
- Direct Department Calling group
- Automatic Call Distribution split.

Use of CORs can be categorized as follows:

- Calling party restrictions
- Called party restrictions
- Forced entry of account codes (V2, V3, or G1)
- Partitioned Group Number (V3 or G1)
- Service Observing (V3 or G1)
- Priority Queuing (V3 or G1)
- Miscellaneous restriction groups
- Selective denial of public network calling through a Common Control Switching Arrangement (CCSA) or Enhanced Private Switched Communications Service (EPSCS) network

- An Automatic Route Selection (ARS) or Automatic Alternate Routing (AAR) (V2, V3, or G1) Facility Restriction Level (FRL) for control of call routing.

Features assignable as calling party restrictions are as follows:

- Code Restriction
- Origination Restriction
- Outward Restriction
- Toll Restriction.

Features assignable as called party restrictions are as follows:

- Inward Restriction
- Manual Terminating Line Restriction
- Termination Restriction.

Use of CORs

CORs can be used to assign a variety of restrictions to a variety of facilities. The types of restrictions which can be assigned are described in the following paragraphs. As an aid to understanding CORs, the screen form used to administer CORs is shown in Figure 3-19. The values shown on the form are the default values. However, these values can be changed to implement the desired restrictions. The Forced Entry of Account Codes field applies only to Version 2, Version 3, and DEFINITY Generic 1. This field does not appear on Version 1 forms. The Partitioned Group Number, Service Observing, and Priority Queuing fields only apply to Version 3 and DEFINITY Generic 1. These fields do not appear on Version 1 and 2 forms.

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                                                        Page 1 of 1

                              CLASS  OF  RESTRICTION

              COR Number: ___                          FRL: 7

                 APLT? y         Calling  Party  Restriction:  ____none_

(V3 or G1) Partioned Group Number: 1     Called  Party  Restriction:  ____none_

(V3 or G1) Service Observing? n

(V2, V3, or G1) Forced Entry of Account Codes? n

(V3 or G1) Priority Queuing? n

CALLING PERMISSION ( Enter "y" to grant permission to call specified COR )

  0? y      8? y     16? y     24? y     32? y     40? y     48? y     56? y
  1? y      9? y     17? y     25? y     33? y     41? y     49? y     57? y
  2? y     10? y     18? y     26? y     34? y     42? y     50? y     58? y
  3? y     11? y     19? y     27? y     35? y     43? y     51? y     59? y
  4? y     12? y     20? y     28? y     36? y     44? y     52? y     60? y
  5? y     13? y     21? y     29? y     37? y     45? y     53? y     61? y
  6? y     14? y     22? y     30? y     38? y     46? y     54? y     62? y
  7? y     15? y     23? y     31? y     39? y     47? y     55? y     63? y
```

Figure 3-19. Example of Screen Form Used for Implementing CORs

Calling Party and Called Party Restrictions

Calling party restrictions prevent specified users from placing certain calls or accessing certain features. Features assignable as calling party restrictions are Restriction—Toll/Code, Restriction—Voice Terminal—Origination, and Restriction—Voice Terminal Outward. These individual features are described fully elsewhere in this chapter. A brief description is given here:

- Code Restriction—Denies the specified voice terminal completion of outgoing calls to selected office and area codes.
- Outward Restriction—Prevents callers at specified voice terminals from activating the Public Network Access feature. Calls can be placed to other voice terminal users, to an attendant, and to tie trunks.
- Origination Restriction—Prevents callers at specified voice terminals from originating calls. Voice terminal users can, however, receive calls.

- Toll Restriction—Prevents callers at specified voice terminals from placing certain calls with a 0 or 1 as the first or second digit, unless the called office code, area code, or service code is on an allowed calls list. This list can contain up to ten entries. (In areas where area codes can also serve as office codes, the system requires the prefix 1 on area code calls to differentiate them from local calls. In this case, local calls with a 0 or 1 as the second digit are not subject to toll restriction.)

Called party restrictions prevent specified users from receiving certain calls. Features assignable as called party restrictions are Inward Restriction, Restriction—Voice Terminal—Manual Terminating Line, and Restriction—Voice Terminal—Termination. These individual features are described fully elsewhere in this chapter. A brief description is given here:

- Inward Restriction—Restricts users at specified voice terminals from receiving public network, attendant-originated, and attendant-extended calls.

- Manual Terminating Line Restriction—Restricts users at specified voice terminals from receiving calls other than those from an attendant.

- Termination Restriction—Restricts users of specified voice terminals from receiving any calls.

Looking at the screen form used to administer CORs (see Figure 3-19), the Calling Party Restriction field and the Called Party Restriction field are both administered as "none." However, the Calling Party Restriction field can be administered as Code, Origination, Outward, or Toll. Likewise, the Called Party Restriction field can be administered as Inward, Manual (Manual Terminating Line), or Termination. Including "none" as a choice of restrictions, as many as 20 combinations of calling and called party restrictions are possible. However, it is unlikely that all 20 combinations will be needed in any one situation. Therefore, only the required ones should be established.

Calling and called party restrictions are the basis for all CORs. In cases where no restrictions are needed, a single COR could be assigned with calling and called party restrictions of "none." This same COR could be used for unrestricted voice terminals, trunk groups, terminating extension group, UCD groups, DDC groups, data modules, the attendant group, and individual attendant extensions (V2, V3, or G1).

The following are typical examples of calling and called party restrictions which may be assigned to a COR:

- Long-distance calling is to be limited by the Code Restriction feature, but there will be no restrictions on incoming calls.

    — Calling party restriction=Code

    — Called party restriction=None.

- A voice terminal in a storeroom should not be used for outside calling. Also, all incoming calls should be from internal callers.
  - — Calling party restriction=Outward
  - — Called party restriction=Inward.
- A voice terminal in a certain department cannot be used for outside calling. Incoming calls must be from the attendant (assuming that department cannot be dialed directly from the outside).
  - — Calling party restriction=Outward
  - — Called party restriction=Manual Terminating Line.
- Certain voice terminals are to be included in a Uniform Call Distribution group for answering business calls only. These terminals are not to be used individually.
  - — Calling party restriction=Origination
  - — Called party restriction=Termination.

The called party restriction is checked only at the called terminal, module, attendant console, zone, or group. For example, if a call redirects from one voice terminal to another, as through the Call Coverage feature, the called party restriction of the called (redirected from) voice terminal is the only one checked.

Each COR is established as needed and is arbitrarily identified by a number, 0 through 63. For example, if the Class of Restriction for the storeroom is 12, the storeroom voice terminal(s) is assigned COR 12.

Forced Entry of Account Codes (V2, V3, or G1)

Account Codes are used to associate calling information with specific projects or account numbers. This is accomplished by dialing a specific account code before making an outgoing call. With V2, V3, or G1, account code dialing can be optional or mandatory (forced) on a per-COR basis.

Looking at the screen form used to administer CORs (Figure 3-19), the Forced Entry of Account Codes field is preset as "n." This means that account code dialing is optional. A "y" in the field would indicate that account code dialing is mandatory on calls that use trunk access codes to place a call on a trunk group with this COR.

Partitioned Group Number (V3 or G1)

When Automatic Alternate Routing (AAR) and Automatic Route selection (ARS) services are to be partitioned among different groups of users within a single system, all users in a specific group must share the same Partition Group Number (PGN). A PGN is assigned to each COR. The PGN is not a restriction, but a means used to indicate the choice of route tables to be used on a particular call.

Service Observing (V3 or G1)

Service Observing allows a specified user, such as a supervisors, to observe a call that involves other users while the call is in progress. While observing a call, the supervisor can toggle between a listen-only and a listen/talk connection to the call. If the person whose calls are to be observed has a COR that does not permit Service Observing, the supervisor cannot observe that user's calls. If the person whose calls are to be observed has a COR that does permit Service Observing, the supervisor can observe that user's calls. For more information on Service Observing, see the Service Observing feature description elsewhere in this chapter.

Priority Queuing (V3 or G1)

Priority Queuing allows calls with increased priority to be queued (at hunt groups) ahead of calls with normal priority. If a COR is administered as having Priority Queuing, calls made by a user with that COR are queued ahead of non-priority calls and are answered sooner.

For intraflowed calls from one hunt group to another to be given priority queuing treatment, ACD software must be activated.

Selective Denial of Public Network Calling Through a Common Control Switching Arrangement (CCSA) or Enhanced Private Switched Communications Services (EPSCS) Network (APLT Field)

Public network calling via the private CCSA or EPSCS network (commonly referred to as off-network calling) is optional on a per-private network basis. If off-network calling is not provided, then the APLT field can be ignored. If off-network calling is provided, then permission or denial to access the off-network capability is set via the APLT field. Users assigned a COR that has APLT set to "n" (no) can use off-network calling. Users assigned a COR that has APLT set to "y" (yes) cannot. If there is a need for both yes and no choices in a system, separate CORs must be assigned to reflect this.

Looking at the screen form used to administer CORs (Figure 3-19), the APLT field is preset as "y." This means that a facility with this COR is not allowed to access CCSA or EPSCS off-network capabilities for public network calling. An "n" in this field indicates that the facility can access CCSA or EPSCS off-network capabilities.

Automatic Route Selection (ARS)/Automatic Alternate Routing (AAR) Facilities Restriction Level (FRL) for Control of Call Routing

If the system does not use AAR or ARS to determine the most-preferred routing of calls, then the FRL field can be ignored. If AAR or ARS is used, then an FRL is used to either allow or deny access to certain routes. The FRL for the outgoing (trunk) side of the call is provided in the AAR/ARS Routing Pattern. Although each outgoing trunk group has a COR and each COR has an FRL, this FRL is not used. Call routing is determined by a comparison of the FRLs in the AAR/ARS Routing Pattern and the FRL in the COR of the call originator (typically, a voice terminal user).

The FRL field (see Figure 3-19) is preset to "7". However, this field can have a value of 0 through 7. An originating FRL of 0 has the least calling privileges, whereas an originating FRL of 7 has the most calling privileges. Each of the up to 6 routes in each of the up to 16

(V1) or 254 (V2, V3, or G1) ARS Routing Patterns also has an FRL. These route FRLs can also have a value of 0 through 7. A route FRL of 0 is the least restrictive, whereas a route FRL of 7 is the most restrictive. An FRL of 0 will be checked before the other routes in a given ARS routing pattern. To access a route, the originating FRL must be greater than or equal to the route FRL. Determination of appropriate FRL values must be made with respect to the outgoing routes from a specific system and the desired levels of calling privileges. This is part of ARS customization. The FRL of the call originator is contained in the COR assigned. The FRL field in a COR assigned to an outgoing trunk group is never checked. and should be ignored.

Assuming AAR and/or ARS have/has been customized for a system, the System Manager must establish unique CORs for each of the up to eight levels of ARS calling privileges that will be used in the system. However, these CORs must maintain the desired restrictions dictated by the other fields on the screen form. The simplest case is a COR specifying no restriction. Ordinarily, this COR can be assigned to all unrestricted users. However, if some subset(s) of these users requires different FRLs, separate CORs must be established for each different FRL required.

For a detailed description of Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), and Facility Restriction Levels (FRLs), refer to the individual feature descriptions given elsewhere in this chapter.

Miscellaneous Restriction Groups

Miscellaneous Trunk and Miscellaneous Terminal Restriction groups restrict access to a terminal, module, zone, attendant console, or group. This is accomplished via the COR assigned to the calling and the called facilities. When a COR is administered, access by that COR to each of the 64 CORs is either allowed or denied. Since a given COR can be assigned to both calling and called facilities, calling to one's own COR can be restricted. This is fully explained in the following paragraphs.

The simplest way to understand miscellaneous restrictions is to look at the screen form used during implementation (see Figure 3-19). When a COR is established, the assigned number is entered in the COR Number field. If this COR is assigned to a facility that originates a call, such as a voice terminal, the calls to CORs associated with terminating facilities can be prohibited. The Miscellaneous Restriction group information is found in the CALLING PERMISSION field. A "y" entry in this field indicates that the COR specified at the top of the form can call the COR numbers that contain a "y." If an "n" is entered, the specified COR cannot be called by the COR number at the top of the form. On the screen form in Figure 3-19, no restrictions apply because all 64 CORs are specified as "y."

The V1 screen form in Figure 3-20 gives an example of Miscellaneous Restriction groups. This form is for COR 6 as is indicated in the COR Number field. The 64 COR numbers in the CALLING PERMISSION field relate which CORs can or cannot receive a call from a facility with a COR of 6. In the example shown:

- A facility with a COR of 3, 7, or 10 cannot be called by a facility with a COR of 6.
- A facility with any COR other than 3, 7, or 10 can be called by a facility with a COR of 6.

Miscellaneous Restriction groups apply on a per-COR basis. However, the same COR can be assigned to more than one facility. Facilities with the same COR may be like facilities (such as two voice terminals) or different facilities (such as a voice terminal and a trunk group). In either case, the same restrictions apply to both facilities.

Certain facilities, such as voice terminals, can originate and receive calls. Call origination and termination restrictions are specified via a single COR. Miscellaneous Restrictions can prevent calling any COR, including one's own COR. If in Figure 3-20, COR 6 in the CALLING PERMISSION field is set to "n," then an originating facility with a COR of 6 cannot call any facility with a COR of 6. This means that two voice terminals, each with a COR of 6, cannot call each other.

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                                                                Page 1 of 1

                                  CLASS OF RESTRICTION

                  COR Number: 6                              FRL: 7

                        APLT? y          Calling Party Restriction:  none

 (V3 or G1) Partioned Group Number: 1     Called Party Restriction:   none

(V3 or G1) Service Observing? n

 (V2, V3, or G1) Forced Entry of Account Codes? n

(V3 or G1) Priority Queuing? n

CALLING PERMISSION ( Enter "y" to grant permission to call specified COR )

  0? y      8? y     16? y     24? y     32? y     40? y     48? y     56? y
  1? y      9? y     17? y     25? y     33? y     41? y     49? y     57? y
  2? y     10? n     18? y     26? y     34? y     42? y     50? y     58? y
  3? n     11? y     19? y     27? y     35? y     43? y     51? y     59? y
  4? y     12? y     20? y     28? y     36? y     44? y     52? y     60? y
  5? y     13? y     21? y     29? y     37? y     45? y     53? y     61? y
  6? y     14? y     22? y     30? y     38? y     46? y     54? y     62? y
  7? n     15? y     23? y     31? y     39? y     47? y     55? y     63? y
```

Figure 3-20. Screen Form Used to Explain Miscellaneous Restriction Groups

When a COR is administered, the allowance or denial of access from that COR to each of the 64 CORs applies only to the COR being administered. For example, if COR 6 is administered with access denied to COR 3, this only specifies that COR 3 cannot receive a call from COR 6. Whether or not COR 3 can be accessed by any other COR (for example, COR 7) is determined when that COR (COR 7) is administered. From this, it follows that a single COR cannot be used to provide both unrestricted service and miscellaneous restrictions.

Class of Restriction Examples

The examples given here are designed to help in the understanding of CORs and to illustrate some of the practical aspects of CORs. These are, however, only examples. In reality, each system must be administered to meet its individual needs.

Example Using Miscellaneous Restrictions

As an illustration of miscellaneous restrictions, assume a system installation provides the following:

- Central office trunks
- WATS
- FX trunks
- Data modules
- Attendant service
- Voice terminals
- Direct Inward Dialing (DID) trunks
- Remote Access.

In an unrestricted environment, each of the above facilities could have the same COR. However, suppose the following requirements exist:

- Attendants cannot make data calls.
- Remote Access can be used for data calls only.
- DID cannot be used for data calls except through Remote Access. (A dedicated Remote Access trunk group is not required, although one or more could be provided. This example assumes all Remote Access is via DID.)
- There are three classes of voice terminals:
  - — Those that can call anywhere, any time
  - — Those that can place local central office and in-house calls only
  - — Those that can place local central office, FX, and in-house calls only.

To implement the above requirements, a COR must be assigned to each facility or group of facilities. For simplicity, each can have a unique COR. The CORs are arbitrarily assigned as follows:

- COR 30—Local central office trunks
- COR 31—WATS trunks
- COR 32—FX trunks
- COR 33—Data modules
- COR 34—Attendant group
- COR 35—Unrestricted voice terminals
- COR 36—Voice terminals that can place in-house and local central office calls only (no FX or WATS calls)
- COR 37—Voice terminals that can place in-house, local central office, and FX calls only (no WATS calls)
- COR 38—DID trunk group
- COR 39—One of the remote access barrier codes (can be up to ten).

With the CORs defined, it should be individually determined which CORs cannot call other CORs. This is done as follows:

- COR 30 (local central office trunks)—No restrictions were specified for these trunks. The default values on the screen form (see Figure 3-19) are sufficient. No action is required, except to specify a COR number of 30.
- COR 31 (WATS)—CORs that cannot use WATS are specified as they are encountered. WATS itself is an outgoing service without any calling capabilities. Thus, Miscellaneous Restrictions are not specified on this form. The Calling Party Restriction should be "none" (although this restriction does not really have any meaning for an outgoing facility). Similarly, the Called Party Restriction applies to facilities capable of answering a call. Since this is not the case with WATS, "none" should be specified. Again, the default values are sufficient, so only the COR number needs to be specified.
- COR 32 (FX)—According to the requirements for this example, no restrictions apply. Reasons are the same as for WATS. Only the COR number needs to be specified.
- COR 33 (data modules)—No restrictions apply for reasons similar to the reasons why no restrictions were assigned for WATS. Only the COR number needs to be specified.
- COR 34 (attendant group)—The attendant group cannot call COR 33 (data modules). Specify an "n" beside COR 33 in the CALLING PERMISSION field. Specify 34 in the COR Number field.
- COR 35 (unrestricted voice terminals)—Since no restrictions were specified, only the COR number needs to be entered.

- COR 36 (no FX or WATS calls)—This COR cannot call COR 32 (FX) or COR 31 (WATS). Specify an "n" beside CORs 32 and 31 in the CALLING PERMISSION field. Specify 36 in the COR Number field.
- COR 37 (no WATS calls)—This COR cannot call COR 31 (WATS). Specify an "n" beside COR 31 in the CALLING PERMISSION field. Specify 37 in the COR Number field.
- COR 38 (DID)—This COR cannot call COR 33 (data modules). Specify "n" beside COR 33 in the CALLING PERMISSION field. Enter 38 in the COR Number field.
- COR 39 (Remote Access barrier code)—This COR can be used for data calls only. Thus, this COR can call COR 33, but not CORs 30 (local central office), 31 (WATS), 32 (FX), 34 (attendant group), 35, 36, or 37 (voice terminals). Specify an "n" beside CORs 30, 31, 32, 34, 35, 36, and 37 in the CALLING PERMISSION field. Enter 39 in the COR Number field. (The CORs listed in the CALLING PERMISSION field can be viewed as terminating or screening CORs that can or cannot be called by the originating COR. Since COR 38 [DID] is neither a terminating nor a screening COR, it does not have to be considered when assigning the barrier code COR.)

Example Using Calling Party Restrictions, Called Party Restrictions, and Miscellaneous Restrictions

To illustrate the use of both Calling and Called Party restrictions, and Miscellaneous restrictions, assume a system installation provides the following:

- Central office trunks (outgoing)
- WATS
- FX trunks (outgoing)
- Voice terminals
- Data modules
- Terminating Extension Groups
- Loudspeaker Paging.

Suppose that the following requirements exist:

- Only the attendant can access loudspeaker paging.
- Terminating Extension Groups can only accept calls from internal voice terminals.
- There are six classes of voice terminals:
  - Those that are toll restricted
  - Those that cannot call outside to a public network (outward restricted)

— Those that can receive calls only from an attendant

— Those that can call anywhere, any time

— Those that cannot place FX or WATS calls

— Those that cannot place WATS calls.

To implement the above requirements, a COR must be assigned to each facility or group of facilities. For simplicity, each can have a unique COR. The CORs are arbitrarily assigned as follows:

- COR 40—Local central office trunks
- COR 41—WATS trunks
- COR 42—FX trunks
- COR 43—Attendant group
- COR 44—Data modules
- COR 45—Terminating Extension Groups
- COR 46—Loudspeaker Paging Access Zones
- COR 47—Unrestricted voice terminals
- COR 48—Voice terminals that are toll restricted
- COR 49—Voice terminals that are outward restricted
- COR 50—Voice terminals that can only receive calls from an attendant
- COR 51—Voice terminals that cannot place FX or WATS calls
- COR 52—Voice terminals that cannot place WATS calls.

With the CORs defined, it should be determined individually which CORs cannot call other CORs. This is done as follows:

- COR 40 (local central office trunks)—Restrictions that prohibit access to this COR are assigned when the originating CORs are considered. Only the COR number has to be specified on this form.
- COR 41 (WATS)—This is the same case as described in the previous configuration example. Only the COR number needs to be specified.
- COR 42 (FX)—Again, only the COR number needs to be specified.
- COR 43 (attendant group)—No restrictions were stated, so only the COR number needs to be specified.

- COR 44 (data modules)—No restrictions were stated, so only the COR number needs to be specified.
- COR 45 (Terminating Extension Group)—This COR can receive internal voice terminal-originated calls only. Since no tie trunks are specified for this example, the Inward Restriction feature can provide the desired restriction. Specify "inward" as the Called Party Restriction. If dial repeating tie trunks are provided, Miscellaneous Restrictions could be used to deny trunk access to the group. Also, specify 45 as the COR number.
- COR 46 (Loudspeaker Paging Access zones)—Since this COR can be accessed by an attendant only, the Manual Terminating Line feature can provide the restriction. Specify "manual" as the Called Party Restriction. Specify 46 as the COR number.
- COR 47 (unrestricted voice terminals)—No restrictions were stated, so only the COR number needs to be specified.
- COR 48 (toll restricted voice terminals)—Specify "toll" as the Calling Party Restriction. Specify 48 as the COR number.
- COR 49 (outward restricted voice terminals)—Specify "outward" as the Calling Party Restriction. Specify 49 as the COR number.
- COR 50 (voice terminals that can only receive calls from an attendant)—Specify "manual" as the Called Party Restriction. Specify 50 as the COR number.
- COR 51 (voice terminals that cannot place WATS or FX calls)—None of the Calling Party Restrictions uniquely prohibit WATS and FX calls, so Miscellaneous Restrictions are used. Enter an "n" beside COR 41 (WATS) and COR 42 (FX) in the CALLING PERMISSION field. Leave the Calling Party Restriction as "none" and specify 51 as the COR number.
- COR 52 (voice terminals that cannot place WATS calls)—Enter an "n" beside COR 41 (WATS) in the CALLING PERMISSION field. Leave the Calling Party Restriction as "none" and specify 52 as the COR number.

Another method to determine COR assignment is to consider the restrictions to be assigned. The requirements given for this example were as follows:

- Only the attendant can access loudspeaker paging.
- Terminating Extension Groups can only accept calls from internal voice terminals.
- The six classes of voice terminals are:
  - — Those that are toll restricted
  - — Those that cannot call outside to a public network (outward restricted)
  - — Those that can receive calls only from an attendant

— Those that can call anywhere, any time

— Those that cannot place FX or WATS calls

— Those that cannot place WATS calls.

Assignments for these requirements could be made as follows:

- COR 20—Manual Terminating Line Restriction
- COR 21—Inward Restriction
- COR 22—Toll Restriction
- COR 23—Outward Restriction

  Note: A new Manual Terminating Line Restriction for voice terminals was not established. COR 20, above, can be assigned.

- COR 24—Unrestricted
- COR 25—COR for WATS
- COR 26—COR for FX
- COR 27—Provides Miscellaneous Restrictions for WATS and FX. Enter an "n" beside COR 25 and COR 26 on the form for COR 27.
- COR 28—Provides Miscellaneous Restriction for WATS. Enter an "n" beside COR 25 on the form for COR 28.

Now assign the appropriate COR to each physical or screening facility:

- Central office trunks—COR 24 (unrestricted)
- WATS—COR 25 (WATS COR)
- FX—COR 26 (FX COR)
- Attendant group—COR 24 (unrestricted)
- Voice terminals—COR 22 (toll), COR 23 (outward), COR 20 (manual), COR 24 (unrestricted), COR 27 (WATS and FX miscellaneous), or COR 28 (WATS miscellaneous), as required
- Data Modules—COR 24 (unrestricted)
- Terminating Extension Group—COR 21 (inward)
- Loudspeaker Paging trunks—COR 20 (manual).

This latter method is probably more difficult to use, but it minimizes the number of CORs established. This method required 9 CORs to effect the same restrictions as 13 CORs with the previous method.

Considerations

COR provides the means to consolidate assignment and administration of the various restriction features available with the system.

All items associated with a COR are distinct and separate. A unique COR must exist for each needed combination of FRLs, CCSA/EPSCS off-network restrictions, calling party restrictions, called party restrictions, and miscellaneous restrictions. Up to 64 CORs can be established, as required, to provide the needed combinations.

Interactions

The following features interact with the Class of Restriction feature.

- AAR/ARS Partitioning

  Partition Group Numbers are assigned via a COR.

- ARS/AAR

  Originating FRLs are assigned via a COR. Termination Restrictions do not apply to ARS/AAR calls.

- Bridged Call Appearance

  The Class of Restriction assigned to a voice terminal's primary extension.

- Call Coverage

  Users who may normally be restricted from calls can still receive calls directed to them via Call Coverage.

- Call Forwarding All Calls

  If a call would normally be restricted between the forwarding and forwarded-to extensions, Call Forwarding activation is denied. However, if restrictions are assigned after Call Forwarding is activated, any termination restrictions are ignored.

- Code Restriction

  This feature is assigned to an originating facility via a COR. (Code Restriction is assigned to an outgoing trunk group on the trunk group form.)

- Controlled Restriction

  Restrictions assigned via the Controlled Restriction feature override the calling and called party restrictions via a COR.

- Emergency Access to Attendant (V3 or G1)

  Emergency Access to Attendant calls are not restricted by COR.

- Forced Entry of Account Codes

  This feature can be assigned via a COR.

- Inward Restriction

  This feature is assigned via a COR.

- Manual Terminating Line Restriction

  This feature is assigned via a COR.

- Origination Restriction

  This feature is assigned via a COR.

- Outward Restriction

  This feature is assigned via a COR.

- Private Network Access and Public Network Access

  Access to the public network via the private network is allowed or denied via a COR (assuming the private network provides the capability to access the public network).

- Termination Restriction

  This feature is assigned via a COR.

- Toll Restriction

  This feature is assigned to an originating facility via a COR. (Toll Restriction is assigned to an outgoing trunk group on the trunk group form.)

Administration

Class of Restriction (COR) is administered by the System Manager. For each COR which is assigned, the following items must be administered:

- COR Number

- Facility Restriction Level (FRL)
- Permission to access EPSCS or CCSA off-net facilities
- Calling Party Restriction
- Called Party Restriction
- Permission to call other CORs
- Forced Entry of account codes for SMDR (V2, V3, or G1) (yes or no)
- Partitioned Group Number (V3 or G1)
- Priority Queuing (V3 or G1) (yes or no)
- Service Observing (V3 or G1) (yes or no).

Assignment of Restrictions

A COR is assigned to each of the following:

Voice Terminals

All voice terminals must be assigned a COR. The same COR may be assigned to all voice terminals or a unique COR may be assigned to a particular voice terminal or group of voice terminals. This COR applies individually to each voice terminal and is independent of all other COR applications. such as Miscellaneous Restriction groups or Uniform Call Distribution groups.

The main items of concern for individual voice terminals are calling, party restrictions and called party restrictions (described previously in "Use of CORs"). If no restrictions are needed for a certain group of voice terminals, "none" can be specified for both calling party and called party restrictions. If it is desired to restrict a group of voice terminals from making outside calls, a COR specifying a calling party restriction of "outward" should be established.

Additionally, miscellaneous restrictions, restrictions to CCSA and EPSCS off-network calling capabilities, and FRLs also apply. A separate COR must be established for each unique set of restrictions.

Trunk Groups

Each trunk group is assigned a COR. Trunk groups are assigned CORs mainly for the use of miscellaneous restrictions. For example, in Figure 3-20, access to trunk groups with a COR of 3, 7, or 10 is denied to facilities with a COR of 6.

Calling party and called party restrictions should be "none." Whether or not a central office or FX trunk group is toll or code restricted is specified on the trunk group form used during implementation.

For Toll Restriction to apply on a call, either the trunk group or the originating facility must specify "toll." For Code Restriction to apply on a call, both the trunk group and the originating facility must specify "code." The originating facility can be specified as "code" or "toll" via the Calling Party Restriction field of the COR assigned to the facility. If the originating facility is specified as something other than "code" or "toll," and the trunk group is specified as "code" or "toll," the call will be neither toll nor code restricted. This paragraph is summarized as follows:

Originating Facility Restriction	Trunk Group Restriction	Restriction Applied On Call
toll	toll	toll
toll	code	toll
code	toll	toll
code	code	code
other	code	none
other	toll	toll

Attendant Consoles (as a group) and Individual Attendant Extensions

Attendants are normally allowed full access to the system's capabilities. Therefore, calling and called party restrictions will usually be "none." Also, access to the attendant is normally allowed to all CORs. This is accomplished via a "y" (yes) for the attendant's COR in the Calling Permission field on the screen form for each assigned COR.

Data Module, Loudspeaker Paging Access Zone, Code Calling Access Zone, and Remote Access Barrier Code

Each data module, Loudspeaker Paging Access zone, Code Calling Access zone, and Remote Access barrier code is assigned a COR. Through Miscellaneous Restriction groups certain users are allowed access to certain facilities, while other users are denied access. For example (looking at Figure 3-20), if a Loudspeaker Paging Access zone has a COR of 3, 7, or 10, then a voice terminal with a COR of 6 cannot access that Loudspeaker Paging Access zone.

Terminating Extension Group, Automatic Call Distribution Split (V3 or G1), Uniform Call Distribution Group, and Direct Department Calling Group

These groups are set up to receive calls. A COR is assigned to each group. This COR is distinct and separate from CORs assigned to the individual group members. The group COR allows or denies calls to the group. Since Miscellaneous Restriction groups are normally used to restrict calling, called party restrictions should be specified as "none." Since a group cannot originate a call, calling party restrictions do not apply. However, for simplicity, "none" is normally specified. For calls by group members or calls to individual group members, the COR assigned to the voice terminal applies. The group COR has no effect on calls directly to or from a group member.

The important aspect of these CORs is that they allow the called party restrictions of the group (normally none) to be different from the called party restrictions of the individual group members (Inward, Manual Terminating Line, or Termination).

Hardware and Software Requirements

No additional hardware or software is required.

Class of Service

Description

Defines whether or not voice terminal users may access the following features and functions:

- Automatic Callback
- Call Forwarding All Calls
- Data Privacy
- Priority Calling
- Off-Hook Alert (V3 or G1)
- Console Permission (V3 or G1)
- Client Room (V3 or G1).

With Versions 1 and 2, only the first four features in the above list are administered using Class of Service. The last three functions in the list are not available with Versions 1 and 2. With Version 3 or DEFINITY Generic 1, all of the features and functions listed above are administered using Class of Service.

There are only two choices for each feature: a voice terminal user or individual attendant **can** or **cannot** access the feature.

With Versions 1 and 2, four features with two choices each yields 16 possible combinations. A Class of Service (COS) parameter is preassigned for each of these 16 combinations. Although the system does allow changing these parameters, there is no need to do so. Any change will result in unneeded duplication. Which COS numbers represent which combination of allowed/denied features are given in *AT&T System 75—Implementation, Release 1 Version 1,* 555-200-650, *AT&T System 75 and System 75 XE—Implementation, Release 1 Version 2,* 555-200-651, *AT&T System 75—Implementation, Release 1 Version 3,* 555-200-652, and *DEFINITY Communications System Generic 1—Implementation,* 555-204-654.

To assign a COS, simply choose the COS number, 0 through 15, that represents the desired allowed/denied combination of features and indicate that number when implementing voice terminals.

With Version 3 or DEFINITY Generic 1, there are still 16 possible COSs. However, each COS is used to allow or deny access to seven features and functions. The parameters can be changed to meet individual COS needs. To assign a COS, administer the desired allowed/desired combination of features and functions for one of the 16 COSs, and indicate that COS number when implementing voice terminals.

The following functions are those that are available only with Version 3 or DEFINITY Generic 1 systems.

- Off-Hook Alert

  This function can be administered only if the optional Emergency Access to the Attendant (V3 or G1) feature is provided. The Off-Hook Alert function lets the customer administer yes/no to each of the 16 established COS parameters according to the allowed/denied capability to access this feature.

- Console Permission

  This function allows multi-appearance voice terminal users to control the same features that the attendant controls. This feature is usually available to front desk personnel in a hotel/motel. With console permission, you can do the following:

  — Activate Automatic Wakeup for another extension.

  — Activate and deactivate controlled restrictions for another extension or group of extensions.

  — Activate and deactivate Do Not Disturb for another extension or group of extensions.

  — Activate Call Forwarding All Calls for another extension.

- Client Room

  This function can be administered when Hospitality Services (V3 or G1) are provided. This function allows the Check-in, Check-out, Room Change/Swap, and Maid Status features. In addition, it is required at consoles or terminals that are to receive Message Waiting Notification.

Other than to allow/deny access to the described features, COS has no other use in the system. Restriction groups and call origination/reception privileges are defined and assigned by a Class of Restriction (COR), not a COS.

Considerations

COS is used to assign four features (V1 and V2) or as many as seven features (V3 or G1). Each voice terminal and individual attendant is assigned one of 16 COSs to determine whether or not it will have any or all of these features. COS serves no other purpose than to assign these features.

Interactions

None.

Administration

COS is administered by the System Manager. The parameters for each COS can be changed. However, with Versions 1 and 2 it is not necessary to change these parameters because the preassigned values include all possible combinations. The only other administration required is the assignment of a COS to each individual attendant (V2, V3, or G1) and voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

Code Calling Access

Description

Allows attendants, voice terminal users, and tie trunk users to page with coded chime signals.

As many as nine individual paging zones can be provided. (A zone is the location of the loudspeakers, for example, conference rooms, warehouses, etc.) In addition, one zone can be provided to activate all zones simultaneously. Each paging zone requires a separate Code Calling Access code.

A paging party dials the Code Calling Access code and the extension number assigned to the person to be paged. The paging party is automatically parked (through the Call Park feature) on the paged party's extension number. The system translates the number to a chime code and then plays the code over loudspeakers. The paged party, recognizing the chime code, can answer the call from any voice terminal within the system by dialing the Call Park Answer Back access code and his or her own extension number.

Considerations

With Code Calling Access, users do not have to be at their own voice terminal in order to answer calls. Users who are frequently away from their voice terminal or at a location where a ringing voice terminal might be disturbing can be assigned a chime code. When a user's chime code is heard, that user can answer the parked call from a nearby voice terminal.

The system can have up to nine individual zones plus one zone to activate all zones simultaneously.

As many as 125 three-digit chime codes can be provided. Only one extension number can be assigned to each chime code.

Interactions

The following features interact with the Code Calling Access feature.

- Abbreviated Dialing

  If Abbreviated Dialing is used for Code Calling Access, special characters should not be used. If they are used, the call will be denied.

- Call Park

  This feature is automatically provided with Code Calling Access.

- Conference—Attendant

  A call cannot be conference while accessing paging equipment. The attendant can, however, release the call after paging the called party.

- Conference—Terminal

  A call cannot be conference while accessing paging equipment.

- Controlled Restriction

  Controlled Total restriction prohibits use of Code Calling Access.

- Loudspeaker Paging Access

  It is not possible to use a PagePac* paging system for Code Calling Access when multi-zone paging is desired. The PagePac paging systems expect a 2-digit code to access a particular zone. The system, however, immediately plays the chime code once a connection is established.

- Miscellaneous Trunk Restriction

  Voice terminals and tie trunks with this restriction cannot use Code Calling Access.

- Origination Restriction

  This restriction prohibits use of Code Calling Access.

- Transfer

  A call cannot be transferred while accessing paging equipment.

Administration

Code Calling Access is administered by the System Manager. The following items can be administered:

- Trunk access code and Class of Restriction for each of the nine individual paging zones and for the zone used to activate all zones simultaneously.
- Number of times (1 to 3) the chime code will play. If the chime code is set to play more than once, the paging party must remain on the call until the chime code is repeated the desired number of times.

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- Loudspeaker locations (name of zone)
- 3-Digit chime codes for extensions. The codes are combinations of the digits 1 through 5.

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack for each individual zone. (These hardware requirements can be shared with the Loudspeaker Paging Access feature. Activation of each feature is by the assigned trunk access code.)

No additional software is required.

Conference—Attendant

Description

Allows the attendant to set up a conference call for as many as six conferees, including the attendant. Conferees from inside and outside the system can be added to a conference call.

Considerations

Whenever an attendant needs to talk with more than one party at the same time, the Attendant Conference feature can be used. An attendant can also establish a conference call for other voice terminal users or parties outside the system.

The attendant can set up only one conference call at a time. The attendant can hold a conference call on the console or release from it.

The attendant cannot handle any other calls while setting up a conference call.

Once an attendant adds a party to a conference call (whether the call was established by an attendant or other voice terminal user), only the attendant can add another party to the call.

An attempt to add a third outside trunk party to a conference call is denied.

Interactions

The following features interact with the Conference—Attendant feature.

- Bridged Call Appearance

  A Bridged Appearance button can be used to make conference calls.

- Trunk-to-Trunk Transfer

  If Trunk-to-Trunk Transfer is disabled and the attendant releases from a conference call involving only trunk conferees, the trunks are also disconnected.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Conference—Terminal

Description

Allows multi-appearance voice terminal users to set up 6-party conference calls without attendant assistance. Single-line voice terminal users can set up 3-party conference calls without attendant assistance.

Considerations

With the Conference—Terminal feature, voice terminal users can set up their own conference calls without assistance from an attendant.

With assistance from other users, a single-line voice terminal can have more than three parties on a conference call. For example, one user can add a party, who can add another party, and so on.

If a voice terminal user releases from a conference call involving only trunk conferees, the trunks are also disconnected if trunk-to-trunk connections are disallowed through administration.

Interactions

A Bridged Appearance button can be used to make conference calls.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Consult

Description

Allows a covering user, after answering a coverage call, to call the principal (called party) for private consultation.

Consult is activated by first pressing the Conference or Transfer button followed by the Consult button to call the principal. This places the calling party on hold and establishes a connection between the principal and the covering user. The covering user can then add the calling party to the conversation, transfer the call to the principal, or return to the calling party.

Details of how Consult is used in conjunction with Call Coverage are given in the Call Coverage feature description elsewhere in this chapter.

Considerations

Consult can be used to let a covering user consult with the principal, to determine whether he or she wishes to speak with the called party (for example, an executive's secretary may wish to consult the executive on an established call).

Interactions

Consult calls use the Temporary Bridged Appearance maintained on the call. At the conclusion of a Consult call, the bridged appearance is no longer maintained.

Bridged Call Appearances of the principal's extension are not alerted on a Consult call to the principal extension.

Consult is only used in conjunction with the Call Coverage feature.

A Consult call acts as a priority call and will wait at an single-line voice terminal, even if the single-line voice terminal does not have Call Waiting Indication assigned.

Administration

Consult is administered by the System Manager on a per-voice terminal basis. The only administration required is the assignment of a Consult button.

Hardware and Software Requirements

No additional hardware or software is required.

Coverage Callback

Description

Allows a covering user to leave a message for the principal (called party) to call the calling party.

Coverage Callback is activated by pressing the Cover Callback button after answering a coverage call.

Details of how Coverage Callback is used in conjunction with Call Coverage are given in the Call Coverage feature description elsewhere in this chapter.

Considerations

Coverage Callback is useful whenever it is necessary to let the principal know that a call has been received from a certain party.

Interactions

Coverage Callback is only used in conjunction with the Call Coverage feature.

Administration

Coverage Callback is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Cover Callback button.

Hardware and Software Requirements

No additional hardware or software is required.

Coverage Incoming Call Identification

Description

Allows multi-appearance voice terminal users without a display in a Coverage Answer Group to identify an incoming call to that group.

When an incoming call is directed to a Coverage Answer Group, the status lamp associated with the Coverage Answer Group button lights at group member’s voice terminal.

Details of how Coverage Incoming Call Identification (ICI) is used in conjunction with Call Coverage are given in the Call Coverage feature description elsewhere in this chapter.

Considerations

With Coverage ICI, members of Coverage Answer Groups do not have to have a display in order to identify incoming calls to the group.

A second coverage call takes control of the Coverage ICI lamp and does not return control to the previous call when the second call is released.

Interactions

Coverage ICI is used only in conjunction with the Call Coverage feature.

Administration

Coverage ICI is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Coverage Answer Group button.

Hardware and Software Requirements

No additional hardware or software is required.

Customer-Provided Equipment (CPE) Alarm (XEV2, XEV3, or G1)

Description

Provides the customer with an indication that a system alarm has occurred and that the system has attempted to contact a preassigned service organization about the problem. A customer-provided device, such as a lamp or a bell, is used to indicate the alarm situation.

The system can be administered so that the CPE Alarm will be activated during certain alarm levels. Only one of these levels may be administered. The CPE Alarm will be activated when an alarm occurs which corresponds to, or is more severe than, the administered alarm activation level. The levels for which the CPE Alarm can be activated are listed below in descending order, beginning with the most severe.

- Major Alarm—This alarm is the most severe system alarm. A major alarm indicates that a vital system hardware component, which will seriously affect overall service, has failed.
- Minor Alarm—This alarm indicates that a hardware component, which may affect service on a limited scale, has failed.
- Warning Alarm—This alarm indicates that a problem may exist with a hardware component, but the problem does not affect service.

The system can also be administered so that the CPE Alarm is not activated under any of the previously listed alarm levels.

The CPE Alarm is also activated during a Power Failure Transfer (see the Power Failure Transfer feature description elsewhere in this chapter), regardless of the administered alarm activation level. Even if the system is administered so that the CPE Alarm is not activated at any alarm level, it will be activated during a Power Failure Transfer.

The CPE Alarm is deactivated when the problem that caused the alarm is resolved. If there are multiple problems, the CPE Alarm will not be deactivated until all problems, at or above the administered alarm activation level, are resolved.

For more information, see *AT&T System 75, Installation and Test,* 555-200-104, and *DEFINITY Communications System, Generic 1, Installation and Test,* 555-204-104.

Considerations

The CPE Alarm feature lets customers use their own equipment to indicate an alarm condition. This indication lets the customer know when there is a problem with the system and when the problem has been resolved.

Interactions

The following feature interacts with the Customer-Provided Equipment (CPE) Alarm feature.

- Power Failure Transfer

  The CPE Alarm is always activated during a Power Failure Transfer, regardless of the administered alarm activation level.

Administration

The CPE Alarm feature is administered on a per-system basis by the System Manager.

Hardware and Software Requirements

The only hardware required is the actual CPE Alarm device (lamp, bell, etc.). This device must be customer-provided and customer-installed.

No additional software is required.

Data Call Setup

Description

Provides three methods to set up a data call: Data Terminal (keyboard) Dialing, Voice Terminal Dialing, or dedicating a voice terminal for data calls. Typically, when a data terminal is available, keyboard dialing is more convenient and requires less steps; therefore, it should be used whenever possible.

In addition to data terminal dialing and voice terminal dialing, the system accepts calls from other devices, such as a Modular Processor Data Module (MPDM) equipped with an Automatic Calling Unit (ACU) interface module. An analog modem interfaced with an ACU can also be used to provide dialing capability for a host computer.

Voice Terminal Dialing

Voice terminal dialing must be used under the following conditions:

- The Data Terminal is connected to an analog modem.
- The Data Terminal is not accessible for dialing.

Allows voice terminal users to originate and control data calls from the voice terminal. The Transfer feature functions the same for data calls as it does for voice calls. The feature permits a user to set up a call using any unrestricted voice terminal and then transferring the call to a data endpoint. However, the primary way to establish data calls is with the multi-appearance voice terminal Data Extension button(s). Any administrable feature button can be assigned as a Data Extension button in system administration.

The voice terminal Data Extension buttons control the One-Button Transfer to Data, Return-to-Voice, and Data Call Preindication operations for the associated data module. These operations are described below. Multiple Data Extension buttons can be assigned to a multi-appearance voice terminal, and that voice terminal can set up data calls for other data terminals. Also, a single data module can be controlled from Data Extension buttons on up to four voice terminals.

Voice terminal dialing has the advantage that the user may hear the different types of network tones (particularly for V1).

For off-premises dialing, particularly for toll calls, the user may opt for voice terminal dialing, instead of keyboard dialing (in V1 it is the only way to detect tones; in V2, V3, or G1 the user may prefer to hear than to see text).

The following options, either alone or combined, permit flexible procedures for establishing data calls:

- One-Button Transfer to Data

  Allows a user to transfer the call to the associated data module simply by pressing the Data Extension button after the called data endpoint answers. This method is recommended for voice terminal data call setup.

- Return-to-Voice

  Allows a user to return the data connection to the voice terminal. The user simply presses the Data Extension button associated with the busy data module. If the user hangs up following the return, the call is disconnected. If Return-to-Voice is affected by two voice terminal users, each through use of the Data Extension button associated with the two data endpoints of the call, then a voice call is established. Return of a data call to the voice terminal implies that the same (data) call will be continued in the voice mode, or transferred to another data endpoint.

- Data Call Preindication

  Allows the user, before dialing the distant data endpoint, to reserve the associated data module by pressing the Data Extension button. This ensures that a conversion resource, if needed, and the data module are reserved for the call. Use of Data Call Preindication before one-button transfer to data is recommended when establishing data calls that use toll network facilities. Needed conversion resources are reserved before any toll charges are incurred.

Data Terminal (Keyboard) Dialing

Allows a user to set up and disconnect data calls directly from a data terminal. A voice terminal is not needed. The voice terminal functions of switchhook and the audible call progress tones are replaced with keyboard dialing and text known as call progress messages. The message "DIAL:" prompts the user to enter the called data number manually from the keyboard, and "RINGING" informs the user the called data number is being rung. With Version 2, Version 3, or DEFINITY Generic 1, if the data call is placed in queue, the message "WAIT, xx IN QUEUE" is received (xx represents queue position). This queue number is updated by the system as the call moves up in the queue. Table 3-B lists the call progress messages.

To originate and disconnect a call using Data Terminal Dialing, the user presses the BREAK key on the terminal. (This is equivalent to a voice terminal user lifting the handset [call origination] or hanging up [call disconnect].) If the terminal being used does not generate a 2-second continuous break signal, the user can press the ORIGINATE/DISCONNECT button on the data module. Then, the data terminal allows the user to enter digits from the data terminal keyboard, after the message "DIAL:" (which is the equivalent of dial tone on a voice terminal).

In addition to the numeral, #, and * characters found on a touch-tone pad, the dialing information may contain the following special characters:

- SPACE, —, (, and #) may be used to improve legibility. These characters are ignored by the system during dialing.
- + (V2, V3, or G1) character (wait) may be used to "interrupt" or "suspend" dialing until dial tone is received from the distant switch. The + is ignored in V1.
- , (V2, V3, or G1) (pause) character may be used to place a 1.5-second pause in dialing (multiples of the , can be used). The , should not be used in V1.

- % (V2, V3, or G1) (mark) character may be used to indicate the following digits are for end-to-end signaling (touch-tone). This is required when the trunk is rotary. It is not required when the trunk is "touch-tone." The % should not be used in V1.
- UNDERLINE or BACKSPACE characters may be used to correct previously typed characters on the same line.
- @ character may be used to delete the entire line and start over with a new DIAL: prompt.

Each line of dialing information may contain up to 36 characters.

Examples of dialing are as follows:

- DIAL: 3478
- DIAL: 9+(201) 555-1212
- DIAL: 8, 555-2368
- DIAL: 9+555-2368+%9999+123 (remote access).

Single-Line Dialing

All of the dialing information, including pauses and ignored characters, are typed on a single line, The line with the DIAL: prompt must be complete; that is, the dialing information must specify a complete call before the carriage return or line feed.

Single-line dialing is recommended if all dialing information can be entered on one line.

Multiple-Line Dialing (Version 2, Version 3, or DEFINITY Generic 1)

Automatically invoked when a single line of dialing information is incomplete. Multiple-line dialing is used only with off-premises calling.

In multiple line dialing, the DIAL: prompt follows on the next line when all of the dialing information of the previous line has been sent and dial tone has occurred; additional dialing information is requested.

This is a typical off-premises dialing sequence:

- DIAL: 9
- DIAL: (201) 555-2368
- RINGING
- ANSWERED.

Table 3-B. Call Progress Messages for Keyboard Dialing

Displayed Message	Application	Meaning
DIAL:	Placing a call	Equivalent to dial tone. Enter the desired number or feature access code followed by a carriage return or a line feed.
RINGING	Placing a call	Equivalent to ringing tone. Called terminal (far-end) is ringing.
BUSY	Placing a call	Equivalent to busy tone. Called number is in use, or out of service.
ANSWERED	Placing or receiving a call	Notifies calling and called users that call has been answered.
OUTGOING TRKS	Placing a call	Notifies calling user that outpulsing is complete.
ANSWERED - NOT DATA(V2, V3, or G1)	Placing a call	Notifies calling and called users that call has been answered and a modem answer tone has not been detected.
TRY AGAIN	Placing a call	Equivalent to reorder tone. System facilities are currently not available.
DENIED	Placing a call	Equivalent to intercept tone. Call cannot be placed as dialed.
ABANDONED (V2, V3, or G1)	Receiving a call	Notifies called user that the calling user abandoned the call.
NO TONE (V2, V3, or G1)	Placing a call	Notifies user that tone was not detected.
CHECK OPTIONS	Placing a call	Notifies calling terminal that data module options are incompatible.
XX IN QUEUE (V2, V3, or G1)	Call in queue	Current position of the user in queue. XX-indicates position.
PROCESSING* (V2, V3, or G1)	Call in queue	Notifies user when out of queue. Facility is available.
TIMEOUT* (V2, V3, or G1)	Call in queue	Notifies user when time has been exceeded. Call will be terminated.
FORWARDED*	Receiving a call	Equivalent to redirection notification signal. Called terminal has activated Call Forwarding and received a call, and call has been forwarded.

* Bell sounds when message is displayed.

Table 3-B. Call Progress Messages for Keyboard Dialing (Contd)

Displayed Message	Application	Meaning
INCOMING CALL-*	Receiving a call	Equivalent to ringing.
PLEASE ANS-	Receiving a call	Originating voice terminal user has transferred call to data module using One-Button Transfer to Data.
-TRANSFER	Call is transferred to voice	Notifies calling terminal when Data Call Return-to-Voice occurs.
CONFIRMED	Activating or deactivating a feature	Equivalent to confirmation tone. Feature request is accepted, or call has gone to a local coverage point.
-OTHER END (V2, V3, or G1)	During a call	Notifies user that the other end terminated the call.
DISCONNECTED*	Call is terminated	Call or call attempt is disconnected from system
WAIT	Placing a call	Notifies user that normal processing is continuing.
WAIT, XX IN QUEUE(V2, V3, or G1)	Placing a call	Notifies user that call entered a local hunt group queue. XX indicates position.

* Bell sounds when message is displayed.

Considerations

All systems have Data Call Setup capability. This facilitates data calling by eliminating the need to dedicate a voice terminal for data calls. Version 2, Version 3, or DEFINITY Generic 1 offers the enhancement of off-premises Multiple Line Dialing and additional call progress messages.

When a voice terminal user places a data call to a digital data endpoint, and does not transfer the call to another digital data endpoint but uses a modem or acoustically coupled modem, the user must dial the Data Origination access code assigned in the system before dialing the distant endpoint.

Data Call Preindication is activated by pressing a Data Extension button before dialing the distant data endpoint. Preindication is in effect until the associated Data Extension button is pressed again for a one-button transfer; there is no time-out.

The number of assigned Data Extension buttons per voice terminal is not limited. However, only four (V1, V2, and V3) or one (G1) voice terminal(s) can be assigned buttons that access the same data module.

When multiple Data Extension buttons control a single data module, the control is shared except for Data Call Preindication. The module is reserved for the preindicating user while Preindication is in effect. After a data call is established, any of the users with an associated Data Extension button can disconnect the call by using Data Call Return-to-Voice.

When placing outgoing or off-premises calls via keyboard dialing, the call progress message "WAIT" indicates recognition of the nature of the call and acceptance of the call. The "ANSWERED" text indicates completion of outpulsing over the selected trunk. Since no tone detection or analysis is done for these calls, no further messages are given to the user. The best success ratio (V1 only) for placing outgoing calls is achieved by placing the call from a voice terminal and using One-Button Transfer to Data.

Interactions

The following features interact with the Data Call Setup feature.

- Abbreviated Dialing

  This feature can be used by voice terminal or Data Terminal (Keyboard) Dialing users on calls to data endpoints. However, with V2, V3, or G1, only 22 of the 24 available digits in an abbreviated dialing number can be used for keyboard dialing. The remaining 2 digits must contain the "wait" indicator for tone detection.

- Data Call Hot Line

  Upon going off-hook for origination, the system automatically places a call to a predesignated local or off-premises destination.

- Call Forwarding All Calls

  Calls incoming to a data module can be forwarded. That is, calls can be redirected to another endpoint. This feature is activated using Data Terminal (Keyboard) Dialing. If the forwarded-to endpoint is an analog data endpoint, and the calling user is a digital endpoint, modem pooling is activated automatically.

- Modem Pooling

  This feature is automatically available on data calls when the system ascertains the need for a conversion resource. The system automatically inserts the conversion resource. Data Call Preindication (optionally) or Data Origination is used to indicate that conversion resource is needed.

- Uniform Call Distribution (UCD)

  UCD can provide a group of data modules or analog modems for answering calls to facilities, such as computer ports, connected to the data modules or modems.

- Information Systems Network (ISN) Interface

  ISN consists of packet data switches which support data calls between data endpoints and the system. The physical connection to the system is via the Data Line Circuit (DLC) board. The DLC provides eight ports for connection with asynchronous Electronics Industries Association (EIA) RS-232C compatible Data Terminal Equipment.

  Data Terminal (Keyboard) Dialing is used to access ISN endpoints.

Administration

Data Call Setup does not require assignment as such; however, the following related items require administration by the System Manager:

- Data Origination Access Code—Allow users to indicate a need for a conversion resource on an analog to digital data call origination.
- Port Assignments—Assign the data modules, BCTs, DLCs, 7404D, analog modems.
- Modem Pooling—Assign Circuit Packs or ports.
- Data Extension buttons—Assign Data Extension buttons to multi-appearance voice terminals.

Hardware and Software Requirements

Data Call Setup is a means of using data equipment to establish data calls. Requirements for data modules, 510D or 515 BCT voice terminals, and modems are as follows:

- ***Data Modules:*** Each data module requires one port on a TN754 or TN784 Digital Line circuit pack. A Digital Terminal Data Module (DTDM) shares the port with its associated voice terminal.
- ***510D or 515 BCT:*** Each 510D or 515 BCT requires one port on a TN754 or TN784 Digital Line circuit pack for shared use of voice and data.
- ***7400D Series Voice Terminals or CALLMASTER™ Data Communications Terminal:*** Each Voice Terminal requires one port on a TN754 or TN784 (G1) Digital Line circuit pack for shared use of voice and data. The 7403D and 7405D voice terminals require an optional Digital Terminal Data Module. The 7404D requires an optional messaging cartridge, the 7406D requires an optional 703A Data Stand, and the 7407D requires an optional 702A Data Services Unit (DSU) for connection to associated data terminals. In addition, with G1, a 7400B Data Module may be used to provide all 7400D series voice terminals a connection to data terminals and a common Digital Communications Protocol (DCP) interface to the switch.

- ***Modems:*** Each modem requires one port on a TN742 Analog Line circuit pack. (Administration designates the modem as a 2500-series voice terminal and assigns an extension number. A modem is connected to the port instead of a voice terminal. Access is through the assigned extension number.)

- ***Modem Pooling:*** Version 1 requires a TN758 Modem Pool circuit pack (two conversion resources per pack). Version 2, Version 3, or DEFINITY Generic 1 requires either a TN758 Modem Pool circuit pack, or one digital port associated with a Trunk Data Module (either TDM or MTDM) and one analog port with analog modem for each conversion resource. With G1, a 7400A Data Module may be used in place of the TDM or MTDM.

Keyboard Dialing to off-premises data endpoints requires the use of a TN748C Tone Detector circuit pack. Extensive use of features and services using tone detection may necessitate adding additional TN748C circuit packs (several features also use a TN748C).

A TN726 Data Line circuit pack can be used to provide direct access for data terminal users.

No additional software is required.

Data Hot Line (V2, V3, or G1)

Description

Provides for automatic nondial placement of a data call to an endpoint when the originator goes off-hook.

Data Hot Line calls are automatically placed, by the system, from specified digital data endpoints to preassigned extension numbers or off-premises numbers. Hot Line originating endpoints are destinations that are associated with Data Communications Equipment (DCE), Data Terminal Equipment (DTE) connected to the system by a data module, or other devices such as Digital Terminal Data Module (DTDM), Processor Data Module (PDM), 515 Business Communications Terminal (BCT), or a port of a Data Line Circuit (DLC). The destination number is stored in the Abbreviated Dialing List.

Considerations

Data Hot Line offers fast and accurate call placement to commonly called data endpoints. Voice and data terminal users that constantly call the same destination number can use Data Hot Line to automatically place the call by simply lifting the handset or going off-hook.

The number of terminals that can be assigned Data Hot Line is not limited, and the number of terminals that can be assigned the same destination number is not limited. The only limit, if any, would be on the number of entries stored in the Abbreviated Dialing List.

Interactions

The following features interact with the Data Hot Line feature.

- Call Forwarding—All Calls

  A Hot Line originator cannot activate Call Forwarding, since an off-hook intended to dial the Call Forwarding Feature Access Code will cause activation of the Data Hot Line feature instead.

- Data Terminal (Keyboard) Dialing (V2, V3, or G1)

  Any Terminal Dialing text may occur when a Data Hot Line call is being established, with the exception of the inhibition of the initial dial text message prompt normally given on off-hook for origination.

- System/ISN Access

  A data call to an Information Systems Network (ISN) data endpoint from a system digital data endpoint requires a 2-stage dialing. A Hot Line Destination may be an extension number for an outgoing ISN group only, the Hot Line originator must then interact with ISN and manually enter the second address (data endpoint).

Administration

Data Hot Line is administered on per-voice or data terminal basis by the System Manager. The following item requires administration:

- Hot Line Destination Number—Assign the destination number, for that station, in the Abbreviated Dialing List.

Hardware and Software Requirements

No additional hardware or software is required.

Data-Only Off-Premises Extensions

Description

Allows users to establish data calls involving Data Communications Equipment (DCE) or Data Terminal Equipment (DTE) that is located remotely from the System 75 site using DATAPHONE® digital service or other private line data facilities. A Data-Only Off-Premises Extension uses a Modular Trunk Data Module located on-premises. Communication with the remote data equipment is accomplished through the private line facility linking the on-premises Modular Trunk Data Module and the remote data equipment.

The Trunk Data Module and DCE or DTE constitute a digital data endpoint. Data calls to this type of data endpoint can be placed using Voice Terminal Dialing or Data Terminal (Keyboard) Dialing. Since there is no voice terminal at the remote site, data calls can be originated from the remote data terminal using Keyboard Dialing only. If computer-generated dialing is used on calls, it must follow the Keyboard Dialing protocol.

Considerations

All systems have the capability for Data-Only Off-Premises Extensions to allow for data calls to remote DCE using DATAPHONE digital service or other private line data facilities.

Data-Only Off-Premises Extensions provides digital data endpoints located off-premises through a Trunk Data Module located on-premises. Communications to or from this Trunk Data Module (and the associated off-premises equipment) must be through an on-premises Processor Data Module or Digital Terminal Data Module. Communications between Trunk Data Modules are not supported. Likewise, Modem Pooling, which is conceptually similar to a Trunk Data Module, cannot be used on calls to or from a Data-Only Off-Premises Extension.

Interactions

The following features interact with the Data-Only Off-Premises Extensions feature.

- Voice Terminal Dialing

  An on-premises multi-appearance voice terminal may have a Data Extension button associated with the Trunk Data Module used for a Data-Only Off-Premises Extension. The voice terminal user and the remote data equipment user share control of the data module. Actions of the user at the voice terminal may affect the remote user.

  — One-Button Transfer to Data

    The on-premises voice terminal user can transfer a call to the Data-Only Off-Premises Extension. The Data Extension button on the voice terminal lights and the Call in Progress lamp on the data module lights during an established data call.

— Return-to-Voice

If a data call has already been established, the voice terminal user may press the associated busy Data Extension button to transfer the call to the voice terminal. The data module associated with the Data Extension button is disconnected from the call. The Call in Progress lamp on the data module goes dark.

— Data Call Preindication

The multi-appearance voice terminal user presses the idle associated Data Extension button to reserve the data module. The data module is then busy to all users except the Preindicating user, including the remote user. When the data module is reserved, the lamp associated with the Data Extension button winks at the preindicator’s voice terminal and lights at any other associated voice terminals. A remote user receives the BUSY message when attempting to originate a call.

Administration

Data-Only Off-Premises Extensions is assigned on a per-line basis by the System Manager. The following item requires administration:

- Digital Line Cirduit Pack—Assign the associated data module to a vacant port.

Hardware and Software Requirements

Requires a Trunk Data Module and one port on a TN754 or TN784 (G1) Digital Line circuit pack. No additional software is required.

Data Privacy

Description

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data privacy, when activated by a user, denies the system the ability to gain access to, or to superimpose tones onto, the protected call.

To activate this feature, the user dials the activation code at the beginning of the call.

Considerations

All systems have the capability for Data Privacy to provide interruption protection by denying the system the ability to gain access to the protected, analog data call.

Connections involving one or more digital data endpoints (data module) are automatically protected from receiving system-generated tones. In this case, the Data Privacy feature is not needed.

Data Privacy, when activated, applies to both voice and data calls. The feature can be activated on Remote Access calls, but not on incoming trunk calls. Data Privacy is canceled if the call is transferred, added to a conference call, bridged onto, or disconnected by the activating user. Data Privacy can be activated on calls originated from attendant consoles.

Interactions

The following features interact with the Data Privacy feature.

- Attendant Call Waiting and Call Waiting Termination

  If Data Privacy is activated, Call Waiting is denied.

- Intercom—Automatic and Dial

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. Intercept tone is received when the ICOM button is pressed under this condition.

- Modem Pooling

  With V1, V2, and V3, when a call using a modem pool is made, Data Privacy and/or Data Restriction is turned off. With G1, the insertion of a modem pool does not turn off Data Privacy and/or Data Restriction.

- Music-on-Hold Access

  If a call with Data Privacy activated is placed on hold, Music-on-Hold access is withheld to prevent the transmission of some musical tone which a connected data service might falsely interpret as a data transmission.

- Priority Calls

  If Data Privacy is activated, Priority Calls to the activating extension number are denied.

- Busy Verification cannot be done when data privacy is active.

Administration

Data Privacy is activated by a system code, administered by the System Manager. The feature is assigned on a per class-of-service basis.

Hardware and Software Requirements

No additional hardware or software is required.

Data Restriction

Description

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data Restriction, when administered to an extension number or trunk group, denies the system the ability to gain access to, or to superimpose tones onto, the protected call.

This feature is administered at the system level to selected analog and multi-appearance voice terminals and trunk groups. Once administered, the feature is active on all calls to or from the associated terminal or trunk group.

Considerations

All systems have the capability for Data Restriction to prevent overriding or ringing features from interrupting the voice or data call.

Connections involving one or more digital data endpoints (data modules) are automatically protected from receiving system-generated tones. In this case, the Data Restriction feature is not needed.

Data Restriction applies to both voice and data calls. Also, Data Restriction cannot be assigned to attendant consoles. Data Restriction is removed from the current call if it is transferred, added to a conference call, bridged onto, or disconnected from by the restricted extension.

Interactions

The following features interact with the Data Restriction feature.

- Attendant Call Waiting and Call Waiting Termination

  If Data Restriction is activated, Call Waiting is denied.

- Intercom—Automatic and Dial

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. Intercept tone is received when the ICOM button is pressed under this condition.

- Modem Pooling

  With V1, V2, and V3, when a call using a modem pool is made, Data Privacy and/or Data Restriction is turned off. With G1, the insertion of a modem pool does not turn off Data Privacy and/or Data Restriction.

- Music-on-Hold Access

  If a call with Data Restriction activated is placed on hold, Music-on-Hold access is withheld to prevent the transmission of some musical tone which a connected data service might falsely interpret as a data transmission.

- Priority Calls

  Priority Calls to a data-restricted extension number are denied.

- Busy Verification cannot be done when Data Restriction is assigned.

Administration

Data Restriction is assigned on a per-line or trunk basis by the System Manager.

Hardware and Software Requirements

No additional hardware or software is required.

DCS Alphanumeric Display for Terminals (V2, V3, or G1)

Description

Allows calls to or from terminals equipped with alphanumeric displays to have transparency with respect to the display of call-related information.

Calling Name Display is the presentation, on the *called* terminal's alphanumeric display, of the name of the party who originated the call. Called Name Display is the presentation on the *originating* terminal's display of the name of the party to whom the call is directed. Both displays provide more useful and precise information than such general identifiers as a trunk group name or an extension number.

The transparency allows calling and called name information, plus miscellaneous identifiers (ids) to be sent from a terminal on one node to a terminal on another node. Transparency in this area is limited by the type of systems at the endpoint nodes and at the intermediate node, if any.

Considerations

DCS Alphanumeric Display for Terminals gives the user considerable call handling capabilities by displaying call related information on calls to and from other Distributed Communications System (DCS) nodes.

Calls to and from a DEFINITY Generic 1 in a DCS network have Calling/Called Name Display transparency under the following conditions:

- The other party is at another DEFINITY Generic 1 and the tandem node is a System 75 Version 3 or later, DEFINITY Generic 1, System 85 Release 2 Version 2 or later, or a DEFINITY Generic 2.1.

- The other party is at a System 85 Release 2 Version 2 or later, or a DEFINITY Generic 2.1.

- The call is not routed through a tandem System 85 Release 2 Version 1 or Enhanced DIMENSION PBX node. (Such calls will display only the extension number of the calling or called party.)

On outgoing DCS calls, display of the called name may be delayed for a few seconds until the required information arrives from the distant node. The called name display only works between System 75s and DEFINITY Generic 1s.

Within the same System 75 or DEFINITY Generic 1 node in a DCS, complete transparency of Calling and Called Name Display exists.

Interactions

The following DCS configurations provide transparency of alphanumeric display information:

- Networks of two or more System 75s or DEFINITY Generic 1s with a System 75 Version 3 or later, DEFINITY Generic 1, System 85 Release 2 Version 2 or later, or a DEFINITY Generic 2.1 as an intermediate node

- A System 75 or DEFINITY Generic 1 connected to a System 85 Release 2 Version 2 or later, or a DEFINITY Generic 2.1.

Configurations in which System 75s or DEFINITY Generic 1s are connected to or through a System 85 Release 2 Version 1 or an Enhanced DIMENSION PBX are not covered because these nodes do not provide display transparency.

If both DCS and Integrated Services Digital Network—Primary Rate Interface (ISDN-PRI) features are provided with a system, the ISDN-PRI display information is displayed in DCS format.

The following features have transparency with respect to Calling and Called Name Display and miscellaneous id. If the display for a DCS call differs at all from the display for a call between terminals at the same system, the difference is noted. Refer to *DEFINITY Communications System Generic 1 and System 75—Voice Termial Operations,* 555-200-701 for detailed descriptions of call information displays.

- Automatic Callback

  Complete display transparency.

- Call Coverage

  At the calling terminal, the miscellaneous id "cover" is not displayed.

- Call Forwarding

  When a system user calls a party on a different node in the DCS and the call is forwarded, the miscellaneous id "forward" is not displayed. At the covering (forwarded-to) user's terminal, only the calling party's name is shown; the called party's name is not displayed.

- Call Park

  When a DCS call between a local system user and a user on another node is parked by the remote user, the miscellaneous id "park" is not displayed at the local terminal.

- Call Pickup

  When a DCS call from a system user to another node is answered by way of Call Pickup, the miscellaneous id "cover" is not displayed at the caller's terminal.

- Call Waiting

  When a DCS call from a system user to another node is waiting at the called terminal, the miscellaneous id "wait" is not displayed at the caller's terminal.

- Centralized Attendant Service

  When a user dials the extension for Centralized Attendant Service, a Release Link Trunk (RLT) is seized or the caller is queued for an RLT. The caller's terminal will display the trunk group identifier, such as OPERATOR.

- Conference

  When a DCS call is conferenced either at a remote node or at the local system, all DCS Calling and Called Name Display transparency is lost to local system users. If all parties drop out except for a local user and another DCS user, the local user's terminal will display the trunk group identifier.

- Direct Department Calling (DDC)/Uniform Call Distribution (UCD)

  Complete display transparency.

- Internal Terminal-to-Terminal Calling

  Complete display transparency.

- Transfer

  When a DCS call is transferred at a remote node to a user on any node, all DCS Calling and Called Name Display transparency is lost to users on the local system.

Administration

DCS tie trunk groups between nodes must be administered by the System Manager with the Outgoing Display disabled. This enables the called party's name to be displayed at the calling terminal.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Attendant Control of Trunk Group Access (V2, V3, or G1)

Description

Allows an attendant at any node in the DCS to exercise control over an outgoing trunk group at a different node in the cluster.

Each attendant console has 12 Trunk Group Select buttons to be used with the Attendant Direct Trunk Group selection feature. Each button allows the attendant direct access to an outgoing trunk group by merely pressing the button assigned to that trunk group. Each of the 12 buttons has a Busy lamp which lights when all trunks in the associated trunk group are busy. On a basic console, six of these buttons have two additional lamps that are used for Attendant Control of Trunk Group Access. On an enhanced console, all 12 buttons have the additional lamps, but a G1 system is needed to use Attendant Control of Trunk Group Access on more than the first six Trunk Group Select buttons. The two additional lamps are as follows:

- Warn (warning) lamp

  Lights when a preset number of trunks are busy in the associated trunk group.

- Cont (control) lamp

  Lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group.

Attendant control of a remote trunk group in the Distributed Communications System (DCS) network is activated by pressing the Cont Act button followed by the desired Remote Trunk Group Select button. Then the initiating node sends a message to the remote node where the trunk group to be controlled resides. The message indicates that control of that trunk group has been initiated.

When the remote node receives the control activation message from the initiating node, it has 4 seconds to send a reply message back to the initiating node if control of the remote trunk group can be activated. A confirmation message will be sent to the initiating node and the Cont lamp at the corresponding Trunk Group Select button is lighted at the remote node if control of the remote trunk group can be activated. An error message is sent to the attendant at the initiating node if the trunk access code is invalid, if the trunk group is already controlled, or if the remote node is a System 85 or Enhanced DIMENSION PBX and the attendant does not have a Trunk Group Select button with Cont lamp for that trunk group.

When a trunk group is controlled in a DCS environment, calls to the trunk group by anyone other than an attendant are routed to the local attendant at the node where the trunk group resides. If that node does not have an attendant, the call is routed to a Centralized Attendant Service (CAS) main attendant or an attendant at a location arranged for Inter-PBX Attendant Calls. However, if CAS or the Inter-PBX Attendant Calls feature is not provided, the party attempting to call on the controlled trunk receives intercept tone.

A detailed description of CAS and Inter-PBX Attendant Calls is given elsewhere in this chapter.

Considerations

DCS Attendant Control of Trunk Group Access allows attendants to obtain control of access to specific trunk groups at any node in the DCS network. This allows the attendant to monitor the use of the controlled trunk group.

There must be direct DCS tie trunk connections between the initiating node and the remote node where the trunk group to be controlled originates. Otherwise, control of remote trunk groups is denied.

If the remote node (where the trunk group to be controlled resides) is a System 75 or DEFINITY Generic 1, it is not necessary for that node to have an attendant console with corresponding 3-lamp Trunk Group Select button. However, if the remote node is a System 85, DEFINITY Generic 2.1, or Enhanced DIMENSION PBX, control of the trunk group is not allowed unless an attendant at that node has a corresponding 3-lamp Trunk Group Select button.

The attendant must use the Remote Trunk Group Select button to directly access the controlled remote trunk group. If an attendant controls a remote trunk group, and that attendant dials the trunk access codes of the DCS tie trunk and the controlled remote trunk group, the call is routed to the attendant at the node where the trunk group resides.

If Attendant Control of Trunk Group Access is activated, and no attendant is assigned, or the attendant is later removed, calls to a controlled trunk group route to the attendant queue.

Interactions

The following features interact with the DCS Attendant Control of Trunk Group Access feature.

- DCS Attendant Display

  When a user attempts to access a controlled trunk group and is routed to the local attendant, the display shows the reason the call was redirected. If the call is routed via CAS or the Inter-PBX Attendant Calls feature, the display does not show the reason the call was redirected.

- Uniform Dial Plan

  DCS tie trunks should not be attendant controlled. This would result in all Uniform Dial Plan calls on the controlled tie trunk being routed to the controlling attendant instead of to the desired destination.

Administration

The ability of an attendant to control access to a remote trunk group is dependent on the administration by the System Manager of Trunk Group Select buttons for remote trunk groups in the DCS.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Attendant Direct Trunk Group Selection (V2, V3, or G1)

Description

Allows attendants at one node to have direct access to an idle outgoing trunk at a different node in the DCS.

A Trunk Group Select button can be assigned to access a trunk group at the local node or a trunk group at a remote node. A Trunk Group Select button assigned to access a remote node is referred to as a remote Trunk Group Select button. Pressing a remote Trunk Group Select button has the same affect as dialing the tie trunk group access code for the remote node and the trunk access code of the selected trunk.

DCS Attendant Direct Trunk Group Selection functions the same as the regular Attendant Direct Trunk Group Selection feature (fully described elsewhere in this chapter). The only difference is an attendant can access a trunk group at a remote node.

Considerations

With DCS Attendant Direct Trunk Group Selection an attendant can have faster access to trunk groups at remote nodes. There is no need to look up trunk access codes, because the press of a button connects the attendant to the desired trunk group.

There must be a direct DCS tie trunk connection between the initializing node and the remote node where the trunk group to be accessed originates. Otherwise, access to the remote trunk group is denied.

Interactions

None.

Administration

A remote Trunk Group Select button must be assigned both the tie trunk access code to the remote node and the trunk access code of the remote trunk group to be selected. These assignments are made by the System Manager.

With G1, feature buttons may be assigned remote Trunk Group Select buttons, in addition to the 12 fixed Trunk Group Select buttons on each attendant console.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Attendant Display (V2, V3, or G1)

Description

Provides some transparency with respect to the display of call-related information.

Calls to and from a System 75 or DEFINITY Generic 1 in a Distributed Communications System (DCS) environment have Calling/Called Party Identification transparency under the following conditions:

- The other party is at another System 75 or DEFINITY Generic 1 and the intermediate node is a System 75 Version 3 or later, DEFINITY Generic 1, System 85 Release 2 Version 2 or later, or a DEFINITY Generic 2.1.
- The other party is at a System 85 Release 2 Version 2 or later, or a DEFINITY Generic 2.1.
- The call is not routed through an intermediate System 85 Release 2 Version 1 or Enhanced DIMENSION PBX node. (Such calls will display only the extension number of the calling or called party.)

A detailed description of the Attendant Display feature is given elsewhere in this chapter.

Considerations

DCS Attendant Display gives the attendant considerable call handling capabilities by displaying call related information on calls to and from both local and remote nodes. This detailed information can be very useful in processing calls.

System 75 or DEFINITY Generic 1 Classes of Restriction (CORs) may not correspond to those used by an Enhanced DIMENSION PBX, System 85, or DEFINITY Generic 2.1. Therefore, if the DCS network contains nodes other than System 75s, or DEFINITY Generic 1s, the display CORs may be misinterpreted. If it is important that certain CORs between various systems correspond with each other, those CORs should be administered accordingly.

On outgoing calls, the display of called party information may be delayed a few seconds until the required information arrives from the remote node. The called party information is displayed only if both nodes are System 75s or DEFINITY Generic 1s.

DCS tie trunks between nodes must be administered with the Outgoing Display disabled. This enables the called party's name to be displayed at the calling attendant's display.

Interactions

When both Integrated Services Digital Network (ISDN) and DCS display information, or only DCS display information, are received, the switch will display the DCS display information in the DCS format. If ISDN display information is received, and no DCS display information is received, then the ISDN display information is displayed in the ISDN formats.

Administration

The administration required for DCS Attendant Display is the same as that required for the Attendant Display feature that is described elsewhere in this chapter.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Automatic Callback (V2, V3, or G1)

Description

Allows a user at one node to make an automatic callback call to a user at another node in the Distributed Communications System (DCS).

A DCS Automatic Callback call can be activated from a voice terminal at one node to a voice terminal at another node, in the same way as if at a local node under the following conditions:

- If the called party is at a System 85, DEFINITY Generic 2.1, or Enhanced DIMENSION PBX node, the callback call can only be activated if the called node is returning busy tone or special audible ringback.
- If the called party is at a System 75 or DEFINITY Generic 1 node, the callback call can be activated if the called node is returning busy tone, Call Waiting ringback tone, or ringback tone.
- The calling party must disconnect within 6 seconds after hearing the confirmation tone for Automatic Callback activation.

The callback of the calling or called parties is as follows when a callback call has been made to a user at another node:

- When the calling party answers the callback call, and no tie trunk to the called party's node is available, Automatic Callback is reactivated toward the called party. The calling party hears confirmation tone instead of ringback when this occurs.
- If the calling party is on a System 85, DEFINITY Generic 2.1, or Enhanced DIMENSION PBX node and is unable to receive the callback call (for example, a busy single-line voice terminal without Call Waiting), Automatic Callback is reactivated by the calling party's node. If the calling party is on a System 75 or DEFINITY Generic 1 node and is unable to receive the callback call, the callback call is canceled.
- If the calling party is on a System 85, DEFINITY Generic 2.1, or Enhanced DIMENSION PBX node, the callback call will be canceled if the calling party calls the called party, or vice versa. If the calling party is on a System 75 node, the callback call is not canceled when one party calls the other.

A detailed description of the Automatic Callback feature is given elsewhere in this chapter.

Considerations

DCS Automatic Callback eliminates the need for voice terminal users to continuously redial busy or unanswered calls to voice terminals within the DCS network.

An Automatic Callback request is canceled automatically if the called party does not become available within 40 minutes, or if the calling party does not hang up within 6 seconds after activating Automatic Callback.

DCS Automatic Callback does not work on the last trunk between nodes. Thus, if *"n"* trunks are provided, there can be up to *"n - 1"* Automatic Callback calls.

Interactions

The following features interact with the DCS Automatic Callback feature.

- Attendant Control of Trunk Group Access and DCS Attendant Control of Trunk Group Access

  Automatic Callback cannot be activated if the call uses a controlled trunk group.

- Call Forwarding All Calls and DCS Call Forwarding All Calls

  Automatic Callback call cannot be activated toward a voice terminal at a System 75 or DEFINITY Generic 1 node that has Call Forwarding activated.

Administration

The administration required for DCS Automatic Callback is the same as that required for the Automatic Callback feature. This information is given elsewhere in this chapter.

Hardware and Software Requirements

AP/DCS interface and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Automatic Circuit Assurance (V2, V3, or G1)

Description

Allows a voice terminal user or attendant at a System 75 node to activate or deactivate Automatic Circuit Assurance (ACA) referral calls for the entire Distributed Communications System (DCS) network. This transparency also allows the referral calls to be generated at a node other than the node that detects the problem.

If referral calls are generated at a System 75 or DEFINITY Generic 1 node for one or more remote nodes, the remote nodes are notified when ACA referral is activated or deactivated at the System 75 or DEFINITY Generic 1 node.

If referral calls are generated at a remote node for a System 75 or DEFINITY Generic 1 node, the System 75 or DEFINITY Generic 1 node is notified when ACA referral is activated or deactivated at the remote node. This notification is accomplished via the ACA button located on the attendant console or voice terminal at the System 75 or DEFINITY Generic 1 node. The lamp associated with the ACA button lights when ACA referral is activated and goes dark when ACA referral is deactivated. The ACA button serves no other purpose when a remote node generates the System 75 or DEFINITY Generic 1 referral calls.

A detailed description of the Automatic Circuit Assurance feature is given elsewhere in this chapter.

Considerations

The DCS Automatic Circuit Assurance feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time.

Interactions

None.

Administration

DCS Automatic Circuit Assurance requires that the System Manager administer whether ACA referral calls are to be "local," "remote," or "primary":

- If administered as local, referral calls are generated at the System 75 or DEFINITY Generic 1 node for the System 75 or DEFINITY Generic 1 node.
- If administered as remote, referral calls are generated at a remote node for the System 75 or DEFINITY Generic 1 node. In this case, the remote node identification must also be entered.

- If administered as primary, referral calls are made at the System 75 or DEFINITY Generic 1 node for a remote node.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Busy Verification of Terminals and Trunks (V2, V3, or G1)

Description

Allows attendants and multi-appearance voice terminal users to make test calls to voice terminals and trunk groups that are located at other nodes within the Distributed Communications System (DCS).

Attendants and multi-appearance voice terminal users can busy verify voice terminals at a remote location. This is done by first pressing the Verify button and then entering the desired Uniform Dial Plan extension number. The verification then continues the same as if the voice terminal being verified is on the same node.

Multi-appearance voice terminal users can busy verify a trunk at a remote location. This is done by first pressing the Verify button, then dialing the trunk access code of the tie trunk group to the remote node, pressing the Verify button a second time, and then entering the desired trunk access code and the trunk group member number to be verified. The verification of the trunk then continues as if the trunk being verified is on the same node.

Attendant operation is the same except a Trunk Group Select button can be used to access the tie trunk to the remote node. A detailed description of the Busy Verification of Terminals and Trunks feature is given elsewhere in this chapter.

Considerations

DCS Busy Verification of Terminals and Trunks provides an easy method of checking the working condition of extensions and trunks at remote nodes.

Interactions

If the Trunk Identification by Attendant feature is used during busy verification of a trunk (Trunk ID button is pressed), the trunk access code and trunk group member number of the DCS tie trunk being used is displayed.

DCS Busy Verification of Terminals and Trunks transparency is lost if the routing pattern is administered to not delete the RNX and the AAR prefix is inserted on the terminating switch trunk group. The voice terminal display at the terminating switch displays only "a=station name". The extension field is left blank.

Administration

The administration for DCS Busy Verification of Terminals and Trunks is the same as that for the Busy Verification of Terminals and Trunks feature which is described fully elsewhere in this chapter.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Call Forwarding All Calls (V2, V3, or G1)

Description

Allows all calls to an extension number to be forwarded to a selected extension number within the Distributed Communications System (DCS) network or to an external (off-premises) number. This feature is activated or deactivated by dial access code or by a Call Forwarding button. The feature can be activated or deactivated only by voice terminal users within the DCS.

Activation and deactivation of the feature is the same as described for the Call Forwarding All Calls feature described in detail elsewhere in this chapter.

Considerations

With DCS Call Forwarding All Calls, voice terminal users can have their calls follow them to any location within the DCS network or outside the DCS network.

Calls to an attendant cannot be forwarded. However, an attendant can activate or deactivate the feature for other extension numbers within the DCS.

Interactions

If the forwarding extension and the designated extension are at different nodes, and the designated extension's coverage criteria are met on a forwarded call, the call is redirected to a point in the designated extension's coverage path.

If the forwarding extension and the designated extension are at different nodes, Leave Word Calling and Coverage Callback cannot be activated at the designated extension for a forwarded call.

Administration

The administration for DCS Call Forwarding All Calls is the same as that for the Call Forwarding All Calls feature.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Call Waiting (V2, V3, or G1)

Description

Allows calls from one node to busy single-line voice terminals at another node to wait until the called party is available to accept the call.

DCS Call Waiting includes the following features:

- Attendant Call Waiting
- Call Waiting—Origination (not a System 75 or DEFINITY Generic 1 feature)
- Call Waiting—Termination
- Priority Calling.

Attendant Call Waiting, Call Waiting Termination, and Priority Calling function the same between System 75 or DEFINITY Generic 1 nodes in a Distributed Communications System (DCS) as they do from a single System 75 or DEFINITY Generic 1. These features are described fully elsewhere in this chapter.

Call Waiting—Origination is not a feature of System 75 or DEFINITY Generic 1, but is supported in a DCS for calls into a System 75 or DEFINITY Generic 1 from nodes that do provide it. When activated before a call, this feature rings an idle single-line terminal with 3-burst priority ringing. If the called party is busy, the call waits and the busy party hears 3-burst call waiting tone through the handset. The waiting party hears Call Waiting ringback tone.

Call Waiting—Origination can also be activated *after* the caller receives busy tone. After activation, the call waits, the busy party hears 3-burst call waiting tone through the handset, and busy tone changes to Call Waiting ringback tone.

Considerations

With DCS Call Waiting, a single-line voice terminal user, by knowing a call is waiting, can quickly process calls from locations within the DCS.

Call Waiting—Origination can only be received in System 75 or DEFINITY Generic 1; it cannot be activated.

Interactions

DCS Call Waiting is denied when the following features are activated at the single-line voice terminal:

- Automatic Callback (to or from the voice terminal)

- Data Privacy
- Data Restriction.

Administration

None required.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Distinctive Ringing (V2, V3, or G1)

Description

Activates the alerting, or ringing, device of a called terminal so that the user is aware of the type of incoming call before answering it. Distinctive Alerting functions is a Distributed Communications System (DCS) environment as it does within a System 75.

A detailed description of the Distinctive Ringing feature is given elsewhere in this chapter.

Considerations

DCS Distinctive Ringing allows the user to identify the type of incoming calls. By knowing the type of incoming call, the user can answer each call properly.

When DCS transparency is lost for any reason, terminal-to-terminal calls made between nodes produce 2-burst ringing instead of the usual 1-burst ringing. Loss of transparency may occur when the data link between nodes is down or when data transmission delay exceeds the trunk signaling time.

Interactions

The following features interact with the DCS Distinctive Ringing feature.

- Intercom—Automatic

  This feature and its distinctive ringing are not provided between nodes in a DCS.

- Intercom—Dial

  This feature and its distinctive ringing are not provided between nodes in a DCS.

- Manual Signaling

  This feature and its distinctive ringing are not provided between nodes in a DCS.

- Tie Trunk Access

  In a DCS, tie trunk groups can be administered as either internal or external tie trunk groups. Calls from internal tie trunk groups are treated as terminal-originated calls and receive 1-burst ringing. Calls from external tie trunk groups are treated as externally originated calls and receive 2-burst ringing.

Administration

None required.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Leave Word Calling (V2, V3, or G1)

Description

Enables System 75 terminal users to leave preprogrammed "call me" messages at other terminals within the Distributed Communications System (DCS) network. Messages can be left by calling, called, or covering users.

Leave Word Calling (LWC) transparency in a DCS configuration allows messages from a System 75 or DEFINITY Generic 1 to another node, depending on the storage capability of the remote node.

Considerations

DCS LWC lets users within the DCS leave short, simple messages for other users.

System 75, System 85 Release 2 Version 2 or later, DEFINITY Generic 1, and DEFINITY Generic 2.1 can store LWC messages internally. An Applications Processor (AP) is not required. However, both System 85 Release 2 Version 1 and Enhanced DIMENSION PBX must be connected to an AP in order to store LWC messages.

LWC cannot be successfully activated toward any system that is not capable of storing the messages, either internally or in an associated AP.

Messages from one node, through an intermediate node, to a remote node do not require storage capability at the intermediate node.

The following configurations have LWC transparency in a DCS:

- From System 75 or DEFINITY Generic 1 to System 85 Release 2 Version 2 (or later) or DEFINITY Generic 2.1
- From System 75 or DEFINITY Generic 1 through any intermediate node to another System 75 or a System 85 Release 2 Version 2 (or later) or DEFINITY Generic 2.1
- To System 75 or DEFINITY Generic 1 from any other node.

Retrieval of LWC messages is permitted only from a terminal at the node where the messages are stored.

DCS Leave Word Calling cannot be activated from an attendant console.

Interactions

The following features interact with the DCS Leave Word Calling feature.

- DCS Multi-appearance Conference/Transfer (V2, V3, or G1)

  Activation of LWC is denied after a DCS call has been conferenced or transferred.

- DCS Call Forwarding All Calls (V2, V3, or G1)

  If the forwarding extension and the designated extension are at different nodes, Leave Word Calling cannot be activated at the designated extension for a forwarded call.

Administration

The administration for DCS LWC is the same as that for LWC, which is described fully elsewhere in this chapter.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Multi-Appearance Conference/Transfer (V2, V3, or G1)

Description

Provides transparency of conference calls and the transfer of calls within a Distributed Communications System (DCS) network. A user in the DCS can make conference calls or transfer calls originated from any extension in the DCS network to another extension within the DCS.

In a DCS, if a party in a conference hangs up or completes a transfer leaving only outgoing trunks on the call, an attempt is made to preserve the connection if any of the remaining parties on the call is a DCS tie trunk. This can be accomplished if the DCS tie trunk can signal the remote node when the party hangs up. The remote node sends a reply to the originating node, and disconnect supervision is provided for that trunk.

Conference Calls can be placed and calls can be transferred to users within the DCS by dialing the Uniform Dial Plan extension number.

Detailed descriptions of the Conference—Attendant, Conference—Terminal, and Transfer features are given elsewhere in this chapter.

Considerations

DCS Multi-Appearance Conference/Transfer is useful when it is necessary to talk to more than one party at one time within a DCS. Multi-appearance voice terminals must have an idle appearance in order to transfer a call.

Interactions

The following feature interacts with the DCS Multi-Appearance Conference/Transfer feature.

- Voice Terminal Display

  No display transparency is provided for DCS Multi-Appearance Conference/Transfer.

Administration

None required.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

DCS Trunk Group Busy/Warning Indication (V2, V3, or G1)

Description

Provides attendants with a visual indication that the number of busy trunks in a remote group has reached an administered level. A visual indication is also provided when all trunks in a trunk group are busy.

If an attendant has a Trunk Group Select button assigned to a remote trunk group, the button's Busy lamp lights when all trunks in the trunk group are busy. The lamp goes dark when one of the busy trunks becomes available.

If an attendant has a 3-lamp Trunk Group Select button assigned to a remote trunk group, the button's Warn lamp lights when the number of busy trunks in the trunk group reaches the Busy Warning Threshold. The lamp goes dark when the number of busy trunks in the trunk group falls below the Busy Warning Threshold.

To ensure that the busy, warning, and control status of all Trunk Group Select buttons in the Distributed Communications System (DCS) remain consistent with the status of the corresponding trunk groups, some nodes in the DCS broadcast the status of a different local trunk group, every 50 seconds, to all directly connected nodes. For example, a node with 30 trunk groups would take 1500 (50 x 30) seconds to broadcast the status of all 30 trunk groups. This is called a lamp audit. When a node receives a lamp audit message, its lamps are updated accordingly. As a traffic consideration, a System 75 or DEFINITY Generic 1 node only receives lamp audit messages. It does not broadcast lamp audit messages.

Considerations

Trunk Group Busy and Trunk Group Warning Indication is particularly useful with the Attendant Control of Trunk Group Access feature. The indicators alert the attendant when control of access to local and remote trunk groups is necessary.

This feature is only transparent if the remote switch is directly connected by voice tie trunks.

An enhanced attendant console used on a V1, V2, or V3 system will only operate the Warning lamp on the left six Trunk Group Select buttons (all 12 Busy lamps will light.)

Interactions

If Trunk Group Select buttons are assigned for Loudspeaker Paging Access zones, Trunk Group Busy Indicators will provide a visual indication of the busy or idle status of the zones at the remote location as well as at the local node.

Administration

Administration for DCS Trunk Group Busy/Warning Indication is the same as that for the Trunk Group Busy/Warning Indicator feature which is fully described elsewhere in this chapter.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required for V2, V3, or G1. A TN765 Processor Interface circuit pack and DCS software are required for XEV2 and XEV3.

Dial Access to Attendant

Description

Allows voice terminal users to access an attendant by dialing 0. Attendants can then extend the call to a trunk or to another voice terminal.

Considerations

With Dial Access to Attendant, voice terminal users can dial 0 to access an attendant whenever attendant aid is needed.

A voice terminal user calling the attendant by dial access cannot be added to an existing conference by the attendant.

Interactions

Restriction—Origination (administered to a voice terminal by the Class of Restriction) prohibits placing any calls, including Dial Access to Attendant calls.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Dial Plan

Description

The Dial Plan is the system's guide to digit translation. When a digit is dialed, the system must know what to expect, based on that digit. For example, if a voice terminal user dials a 4, the system must know how many more digits to expect before the call will be processed.

The Dial Plan, or first-digit table, established during administration for each system, provides this information. The table defines the intended use of a code beginning with a specific first digit and relates to the system how many digits to collect before processing the code. The choices of a first digit are 0 through 9, *, and #. Permissible code uses and the allowable number of digits are listed below. Please note that enhancements to this dial plan have been made for V3 and G1 systems. For these enhancements, see the Single-Digit Dialing and Mixed Station Numbering feature description elsewhere in this chapter.

- Extension Numbers

  Flexible numbering allows 2-, 3-, or 4-digit extension numbers with Version 1 systems. Version 2, Version 3, or DEFINITY Generic 1 systems can have 1-, 2-, 3-, 4-, or 5-digit extension numbers. The first digit in the extension number tells the system how many digits to expect the extension number to have. Therefore, all extension numbers beginning with the same digit must be the same length.

  Extension numbers can have a first digit of 1 through 9. For example, if a 3-digit extension number is administered and the first digit is a 4, the extension numbers can range from 400 to 499. Also, if a 4-digit number with a 6 as the first digit is administered, the extension numbers can range from 6000 to 6999.

- Attendant

  Dial access to the attendant group is always by the single digit "0." Version 2, Version 3, or DEFINITY Generic 1 provides for Individual Attendant Access (V2, V3, or G1) by assigning each attendant an individual extension number.

- Trunk Access Codes

  A minimum of one digit and a maximum of three digits can be used. Trunk access codes can have a first digit of 1 through 9. For example, 9 could be used for local trunks, 8 for Wide Area Telecommunications Service (WATS) trunks, and 7 for tie trunks.

- Feature Access Codes

  A minimum of one digit and a maximum of three digits can be used. The * and # buttons can be used as party of a feature access code and, when used, must be the first digit. The * or # counts as one digit. For example, *2 could be used to activate Call Forwarding All Calls and #2 used to deactivate Call Forwarding All Calls.

Feature access codes can also have a first digit of 1 through 9. For example, 32 could be used to activate Call Forwarding All Calls and 33 used to deactivate Call Forwarding All Calls.

With Version 1 or Version 2 systems, the first digit administered for one type of entry in the first-digit table cannot also be administered as the first digit of another entry. For example, when a 9 is used as a trunk access code, 9XX cannot be used as an extension number or as a feature access code.

With Version 2, Version 3, or DEFINITY Generic 1 systems, a Uniform Dial Plan may also be established during administration as part of the Dial Plan. This plan provides a common extension number plan that can be shared among a group of switches. If a Uniform Dial Plan is to be established, all extension numbers must be the same length (4 to 5 digits). A Uniform Dial Plan also requires the following information, so that calls will route to the desired switch.

- A PBX Code, which represents the first 1 to 4 digits of a 4- or 5-digit extension number and can range from 1 to 9999 with a maximum of 240 PBX Codes.
- Whether or not the PBX Code is local to this system—this information is required for each PBX Code.
- An RNX, which is associated with the PBX code and is used to select an Automatic Alternate Routing (AAR) pattern for the call—this information is required for each PBX code.
- A PBX ID (1 to 63), which represents a specific switch—this information is required for each PBX Code when the switch is located within a Distributed Communications System (DCS).

Considerations

The entire Dial Plan is dependent on the first digit dialed. The 12 possible choices of a first digit are 0 through 9, * , and #.

Interactions

All dial access features and services provided by the system require the Dial Plan.

Administration

The Dial Plan is administered on a per-system basis by the System Manager. The following items require administration:

- Area code where the system is located

- Whether or not the serving central office requires the digit 1 to indicate a long-distance call
- Whether or not a Uniform Dial Plan is to be established (V2, V3, or G1)
- The type of code and the number of digits in the code for each first digit.

If a Uniform Dial Plan is to be established, the following items must also be administered:

- Number of digits in plan (4 or 5)
- PBX Codes
- Whether or not each PBX Code is local to the PBX being administered
- RNX (Per PBX Code)
- PBX ID (Per PBX Code).

Hardware and Software Requirements

No additional hardware or software is required.

Digital Multiplexed Interface (V2, V3, or G1)

Description

Supports two signaling techniques: Bit Oriented Signaling and Message Oriented Signaling for direct connection to host computers.

System 75 (V2 or V3) supports only the Bit Oriented Signaling (BOS) version of the Digital Multiplexed Interface (DMI) for multiplexed data communication over Data Services Level 1 (DS1) Tie Trunk Service digital facilities between a host computer and System 75. DEFINITY Generic 1 supports both Bit Oriented Signaling and Message Oriented Signaling. Message Oriented Signaling is used with Integrated Services Digital Network-Primary Rate interface (ISDN-PRI).

The DMI provides twenty-three 64-kbps data channels, plus one 64-kbps channel for Common Channel Signaling. Within the data channel, DMI provides control information exchange and data formats supporting data transport at all standard data rates; each data channel can be used in one of the following transfer modes:

- Mode 0 — 64 kbps Clear Channel
- Mode 1 — 56 kbps DDS Compatible
- Mode 2 — 0-19.2 kbps Synchronous/Asynchronous
- Mode 3 — Multiple Virtual Channels.

With V2 or V3, the signaling mode of the DS1 circuit pack must be optioned, via the system administration, for Common Channel Signaling. The format used is a 1.544-Mbps digital signal that consists of a 1.536-Mbps signal multiplexed with an 8-kbps framing channel. The 1.536-Mbps signal is divided into 24 channels of 64 kbps each.

With G1, ISDN-PRI can also be assigned as the signaling mode. In this case, the TN767 DS1 circuit pack and the TN765 Processor Interface circuit pack must be used. The ISDN-PRI is a 1.544-Mbps digital interface that consists of a 1.536-Mbps signal multiplexed with an 8-kbps framing channel. The 1.536-Mbps signal is divided into 24 channels of 64 kbps each (23 "B" voice or data channels and 1 "D" signaling channel). The D channel multiplexes signaling messages for the 23 B channels.

DMI trunks are accessed the same as tie trunks. The only difference is that DMI trunks are connected to host computers while tie trunks are connected to another switch. Each trunk functions like a Processor Data Module (PDM) since the DMI protocol is identical to the Digital Communications Protocol (DCP) format used by the data modules.

Considerations

System DMI support offers high volume (high speed, high capacity) data transmission, via DS1 digital facilities, between host computers and analog or digital data endpoints.

DMI is widely supported. To date, more than 100 data processing suppliers, communications equipment suppliers, and device manufacturers have licensed DMI specifications and have obtained the rights to implement DMI in their products.

With V2 or V3, DMI trunks can only be connected to host computers. Also, queuing for DMI trunks is not provided. With G1, DMI trunks with ISDN-PRI signaling can be connected to a host computer, another PBX, or a public or private network.

Interactions

The following features interact with the Digital Multiplexed Interface feature.

- Data Restriction

  DMI trunks cannot be data restricted.

- Modem Pooling

  Data calls dialed from a local analog data endpoint to a DMI trunk must contain the Data Origination Access Code to obtain a conversion resource. Data calls on DMI trunks to local analog data endpoints will automatically obtain conversion resources, if available.

Administration

DMI support is assigned on a per-system basis by the System Manager. The following items require administration.

- DS1 Circuit Pack—Assign the circuit pack to the system before administration of the associated trunks.
- Processor Interface Circuit pack—If ISDN-PRI Signaling is used, a TN765 Processor Interface circuit pack must be assigned to work in tandem with the TN767 DS1 circuit pack.
- DMI Trunk Group——Associate the trunks to the groups.

Hardware and Software Requirements

One TN722B (V2, V3, or G1) or TN767 (G1) DS1 circuit pack is required for every 23 DMI trunks required. If ISDN-PRI signaling is used, a TN765 Processor Interface circuit pack is also required (G1).

No additional software is required.

Direct Department Calling and Uniform Call Distribution

Description

Allows direct inward access to an answering group other than the attendant even if the system does not have the Direct Inward Dialing (DID) feature.

A Direct Department Calling (DDC) or Uniform Call Distribution (UCD) answering group can consist of voice terminals and individual attendants (described in the Individual Attendant Access feature elsewhere in this chapter). In addition, a UCD group can consist of data modules, data line circuit ports, or modems.

One extension number is assigned to all voice terminals, individual attendants (V2, V3, or G1), data modules, data line circuit ports, or modems in a group or department, that is, to a set that serves the same function and requires call distribution among the members of the group. Incoming calls to a DDC group or UCD group can be internal or external.

The hunting algorithm used by the system to select an idle terminal or console is the only difference between DDC and UCD.

With DDC, an incoming call rings the first available voice terminal or individual attendant (V2, V3, or G1) in the administered sequence. If the first group member in the sequence is active on a call (busy), or has had his or her calls temporarily redirected (via Send All Calls and Call Forwarding All Calls feature descriptions, or the Hunt Group Busy option described later in the "Considerations" section), the call routes to the next group member, and so on. In other words, incoming calls always try to complete at the first group member in the administered sequence. Therefore, the calls are not evenly distributed among the DDC group members.

With UCD, an incoming call will ring the member of the group that has not received a UCD group call for the longest period of time (the most-idle member). In other words, incoming calls to a UCD group extension number will be distributed evenly among the group members.

When DDC or UCD is not provided, incoming Listed Directory Number (LDN) calls, foreign exchange calls, 800 service calls, and automatic tie trunk calls are normally directed to an attendant who must extend the call. When DDC or UCD is provided on a trunk group, incoming calls are automatically directed to the desired DDC group by the switch. Attendant intervention is not required.

Any voice terminal or individual attendant (V2, V3, or G1) can be a member of one or more DDC and/or UCD groups. Data modules, data line circuit ports, and modems are limited to UCD groups and can be a member of one or more groups. Each member of a group also has its own unique extension number and can be called individually. Multi-appearance voice terminals and attendant consoles (V2, V3, or G1) can have an assigned status lamp that identifies an incoming DDC or UCD call. However, the voice terminal or individual attendant (V2, V3, or G1) must be idle (not active on any call appearance) before a group call will be directed to the terminal or console (V2, V3, or G1). Therefore, a voice terminal can receive only one DDC or UCD call at a time.

A queue can be established for a DDC or UCD group. When all members of the group are active, the queue allows incoming calls to await an idle terminal.

When a call enters the queue, a delay announcement interval is started. This interval (0 to 99 seconds) indicates how long a call will remain in queue before the call is connected to a recorded announcement. If Call Coverage is provided, the Don’t Answer Interval (1 to 6 ringing cycles) may also begin when the call enters the DDC or UCD group queue. After these intervals have begun, one of the following occurs:

- If the Coverage Don’t Answer Interval expires before the delay announcement interval expires, the call is redirected to coverage. If no coverage point is available to handle the call, the call remains in queue and may be connected to delay announcement.
- If the delay announcement interval expires before the Coverage Don’t Answer Interval, the call is connected to a delay recorded announcement, if available. Once a call is connected to a delay announcement, it remains in queue until a group member becomes available. If the announcement is already in use, the delay announcement interval is reset. This process (as described above) continues until the call is answered, goes to coverage, is connected to a delay announcement, or the calling party hangs up.

If the delay announcement interval is administered as "0" seconds, the incoming call will automatically be connected to the announcement, if available. The result is a "forced first announcement," and the call will not attempt to access a hunt group member until after the announcement is heard.

Calls connected to a delay recorded announcement remain in queue while the announcement is heard by the caller. If the call has not been answered by the time the announcement is over, the call is connected to music (if provided) or there will be silence, as long as the call remains in queue. When the call begins ringing a member of the hunt group, the calling party hears audible ringing. Music is not provided after a forced first announcement.

The queue length can be set from 1 to 100 calls if queuing is provided. If queuing is not provided, the queue length must be set to 0. If queuing is not provided, if the queue is full, or if all group members (voice terminals or individual attendants only) have activated the Hunt Group Busy option (described later), calls to a busy group receive busy tone (unless using a Central Office trunk) or redirect via the Call Coverage feature. Lamp indicators may be used to give a warning when the number of calls waiting in the queue reaches a predetermined limit (queue warning limit). The queue warning level can be 0 to 35; however, it cannot exceed the queue length. Although it is possible, the queue warning level should not be set to 0, as this would result in the indicator lamp lighting at all times.

When the queue warning level is reached, the indicator lamp lights and remains lighted until the calls waiting in queue are fewer than the queue warning level. A queue warning level lamp may be provided for each DDC or UCD group queue. The lamp can be installed at any location convenient for the group.

As an example of queue warning level and delay announcement operation, assume that there is an incoming call to a DDC or UCD group with the following parameters:

- Queue length is 10 calls.

- Queue warning level is 5 calls.
- Recorded announcement delay is 20 seconds.

Also assume the following:

- All DDC or UCD group members are busy.
- The call is the fifth call in the queue.

Since all members in the DDC or UCD group are busy, the incoming call enters the queue. The incoming call, being the fifth call in the queue, causes the queue warning level to be reached. This causes the queue warning level lamp to light.

From the indicator lamp, the DDC or UCD group members know the queue warning level has been reached and try to complete their present calls. Meanwhile, the incoming call has been in the queue for 20 seconds and hears the delay recorded announcement. The caller may decide to hang up or may decide to remain in the queue. Assume the caller remains in the queue. When a DDC or UCD group member becomes idle, the longest queued call is directed to that group member. The queue warning level lamp may or may not be lighted at that time, depending on the number of other calls that have been queued. Also, the first four calls in the queue will have heard the delay announcement after being queued for 20 seconds. The queue warning level and delay announcement capabilities are independent of each other.

A Coverage Incoming Call Identification (ICI) button can be assigned to a hunt group member's multi-appearance voice terminal. The Coverage ICI button allows the user who is a member of more than one hunt group to identify a call that is directed to a specific hunt group. When a hunt group member receives a call that is directed to the hunt group assigned to that button, the button's status lamp will light.

Considerations

DDC and UCD are particularly useful when the answering group assigned receives a high volume of incoming calls. Call completion time is minimized and attendant assistance is not required. This feature can also minimize the use of DID trunks.

If DDC and UCD groups are both used in the system, the number of combined groups and the number of voice terminals per group are determined by the size of the system and call traffic requirements. A maximum of 32 (V1, V2, V3) or 99 (G1) groups with up to 32 (V1 or V2), 100 (V3), or 200 (G1) members per group can be provided. The system maximum, however, is 448 (V1, V2, or V3) or 500 (G1) group members.

Each system can contain up to 10 (V1 and V2) or 64 (V3 or G1) different recorded announcements. Each group queue can be assigned one of these announcements as a delay announcement. A delay announcement can be shared among the DDC groups, UCD groups, or a combination of these groups. Delay announcements may be either analog or digital (integrated). Only one caller can be connected to an analog announcement at any one time. With V3 or G1, as many as five callers can be connected to the same integrated announcement. Callers are always connected at the beginning of the announcement. More efficient use of the announcements is realized if the announcements are brief.

Calls incoming on a non-DID trunk group can route to a DDC group instead of to an attendant. Calls incoming on any non-DID trunk group can have only one primary destination; therefore, the trunk group must be dedicated to the DDC group.

If a delay announcement is used, answer supervision is sent to the distant office when the caller is connected to the announcement. Charging for the call, if applicable, begins when answer supervision is returned.

Multi-appearance voice terminals can receive only one DDC or UCD call at a time. A voice terminal is idle for a DDC or UCD call only if all call appearances are idle.

A Hunt Group Busy option can be administered for the system. When a voice terminal user or individual attendant (V2, V3, or G1) dials the Hunt Group Busy activation code followed by the DDC or UCD group extension number (V1 or V2) or the DDC or UCD group number (V3 or G1), or presses the Aux-Work button, the terminal or console (V2, V3, or G1) appears busy to the DDC or UCD group. This effectively removes the member from the group until the user dials the Hunt Group Busy cancellation code or presses the button again. The Aux-Work button can be assigned to multi-appearance voice terminals only. Calls to a busy hunt group receive a busy signal if the caller is internal or incoming on a Direct Inward Dialing (DID), tie, or DS1 tie trunk. A Caller to a busy hunt group hears ringing if the caller is incoming on a Central Office (CO) trunk.

The last available member of a DDC or UCD group cannot activate the Hunt Group Busy option if any calls are remaining in the queue. An attempt by the last available group member to activate the Hunt Group Busy option results in the following:

- New calls to the DDC or UCD group either receive busy tone or redirect to coverage.
- Calls already in the queue continue to route to the last available voice terminal until the queue is empty.
- At the last available voice terminal or console (V2, V3, or G1), the status lamp associated with the Aux-Work button, if provided, flashes until the queue is empty. When no more calls remain in the queue, Hunt Group Busy is activated and the status lamp, if provided, lights steadily. (The same sequence applies when Hunt Group Busy is dial activated instead of button activated, except there is no status lamp.)

Leave Word Calling messages can be stored for a DDC or UCD group and can be retrieved by a member of the DDC or UCD group, a covering user of the group, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization must be assigned to the message retriever. Also, a remote Automatic Message Waiting lamp can be assigned to a group member to provide a visual indication that a message has been stored for the group. One remote Automatic Message Waiting lamp is allowed per DDC or UCD group. The status lamp associated with this button informs the user that at least one message has been left for the group.

Members of a UCD group used for data communications must be of the same type and serve the same function. Either data modules or analog modems can be used in a UCD group, not a mixture of the two, and the group must be dedicated to a specific, intended use.

Since any member of a data UCD group can be used on a given call, option settings must be the same for all group members. This minimizes call setup failures because of incompatible options between the origination data module or modem and the UCD group data module or modem selected for the call.

A Data Extension button can be used to access the associated data module, even if the module is in a UCD group. Individual data modules or modems can originate and receive calls.

Each UCD group and each UCD member is assigned a Class of Restriction (COR). Miscellaneous Restrictions, described in this chapter, can be used to prohibit selected users from accessing certain UCD groups. Either Miscellaneous Restrictions or restrictions assigned through the COR can be used to prohibit the group members from being accessed individually. Unless such restrictions are administered, each group member can be accessed individually as well as through the group.

When a hunt group is changed from Automatic Call Distribution (ACD) to non-ACD, the agent has to enter the Hunt Group Busy deactivation code in order to receive calls in that hunt group. If an Aux-Work button has been administered for that station, then the lamp associated with that button will light, and the button can be pressed to make the agent available for hunt group calls.

Interactions

The following features interact with the Direct Department Calling and Uniform Call Distribution feature.

- Attendant Call Waiting

  An attendant can originate or extend a call to a DDC or UCD group. Attendant Call Waiting cannot be used on such calls. However, such calls can enter the group queue, if provided. Attendant Call Waiting can be used on call to the individual hunt group members.

- Automatic Callback

  Automatic Callback calls cannot be activated toward a DDC or UCD group.

- Call Coverage

  Calls can redirect to or from a DDC or UCD group.

  If a user has an Aux-Work button, and activates or deactivates Send All Calls, the Hunt Group Busy function associated with DDC or UCD is activated or deactivated simultaneously.

If a user has no Aux-Work button, activating or deactivating Send All Calls still makes the user available or unavailable for DDC or UCD calls, but Hunt Group Busy is not activated or deactivated. The Hunt Group Busy activate or deactivate code and the DDC or UCD extension must be dialed to activate the Hunt Group Busy function.

Activating or deactivating the Hunt Group Busy function does not activate or deactivate Send All Calls.

For a call to a DDC or UCD group to be directed to Call Coverage, each voice terminal in the group must be active on at least one call appearance and the queue, if there is one, must be full. If the queue is not full, a call will enter the queue when no voice terminal is available. Queued calls remain in queue for a time interval equal to the Coverage Don't Answer Interval before redirecting to coverage. If any voice terminal in the group is idle, the call directs to that voice terminal.

- Call Coverage

  When a call is redirected via Call Coverage to a hunt group, the calling party will not hear a forced first announcement, if administered. In order for the redirected call to receive an announcement, the announcements must be administered as first or second delay announcements.

- Call Forwarding All Calls

  When activated for a hunt group member, the activating voice terminal appears busy to the DDC or UCD group.

  When activated for the hunt group extension, calls directed to the hunt group are forwarded away from the hunt group. No announcements (other than a forced first announcement, if administered) associated with that hunt group are connected to the call.

- Data Call Setup (to or from a member of a UCD group)

  Voice Terminal Dialing of Data Terminal (Keyboard) Dialing can be used on calls to a UCD group.

- Direct Inward Dialing (DID)

  If DID is provided and the DDC or UCD group extension number is within the range of extension numbers that can be dialed directly, then the group can be called the same as any voice terminal.

- Distributed Communications System (DCS)

  If a call to a hunt group is forwarded to a hunt group at another DCS node, the caller will not hear the forced first announcement of the forwarded-to hunt group.

  If a hunt group is in night service, with a hunt group at another DCS node as the night service destination, a call to the first hunt group will be connected to the first forced announcement of the hunt group serving as the night service destination.

- Individual Attendant Access (V2, V3, or G1)

  Individual Attendant Extensions can be assigned to DDC and UCD groups. Unlike voice terminal users, individual attendants can answer DDC and UCD calls as long as there is an idle call appearance and no other DDC or UCD call is on the console.

- Multi-Appearance Preselection and Preference

  All assigned call appearances must be idle before a DDC or UCD group call is directed to a voice terminal.

- Music-on-Hold Access

  A call placed in a DDC or UCD group queue can receive a delay announcement followed by music.

- Night Service—Hunt Group (V3 or G1)

  When Hunt Group Night Service is activated for a hunt group and the night-service destination is a hunt group, the caller will hear the first forced announcement, if administered. The call is then redirected to the night service destination hunt group. When a member of the night service hunt group becomes available, the call goes to that member.

- Priority Calling

  A priority call directed to a DDC or UCD group is treated the same as a nonpriority call, except that the distinctive 3-burst ringing is heard.

- Station Message Detail Recording (SMDR)

  The system can be administered to record the called number on incoming calls to a particular hunt group or hunt group member.

- Terminating Extension Group

  A Terminating Extension Group cannot be a member of a DDC or UCD group.

- Voice Terminal Display

  On calls dialed directly to a DDC or UCD group extension number, the DDC or UCD group's identity is displayed at the calling extension.

Administration

DDC and UCD are administered by the System Manager. The following items can be administered for each DDC or UCD group:

- Delay announcement

- Delay announcement interval
- Group extension number, name, and type (DDC and UCD)
- Group Path
- Class of Restriction
- 4-digit security code (for use with AP Demand Print feature)
- Whether or not the group is used as the AP Message Center
- Whether or not the group is served by a queue
- Queue length (0 to 35 calls)
- Queue Warning Threshold (0 to 35 calls)
- Port Number assigned to queue warning level lamp
- Group Members (extension numbers).

Also, the system can be administered to record (via SMDR) incoming calls to a particular hunt group or hunt group member.

Hardware and Software Requirements

Each queue warning level lamp requires one port on a TN742, TN746, or TN769 Analog Line circuit pack. A 21C-49 indicator lamp may be used as a queue warning level lamp. This lamp is approximately 2 inches in diameter and has a clear beehive lens. The lamp operates on ringing voltage and can be mounted at a location convenient to the group.

Each delay announcement requires announcement equipment and one port on a TN742 Analog Line circuit pack. If music is to be heard after the delay announcement, a music source and a port on a TN763 Auxiliary Trunk circuit pack is required. Announcement equipment and music sources are not provided by the system.

No additional software is required.

Direct Inward Dialing

Description

Connects calls from the public network directly to the dialed extension number without attendant assistance.

Direct Inward Dialing reduces the attendant's workload and provides the calling party immediate contact with the called party.

Considerations

Direct Inward Dialing (DID) trunk group(s) from the local telephone company central office is required.

Interactions

The Inward Restriction, Manual Terminating Line Restriction, and Termination Restriction features (administered by the Class of Restriction) prevent receiving DID calls at the restricted voice terminal.

When a DID trunk is accessed via a Listed Directory Number (LDN), the call is routed to the attendant. The attendant display indicates that the call is an LDN call. If night service is activated, DID LDN calls route to a designated DID LDN night extension.

If an incoming DID call is forwarded to another extension and answered by the forwarded-to extension, any other calls to the same DID extension within the next 30 seconds will receive busy tone.

Administration

Direct Inward Dialing is administered by the System Manager. The only administration required is the administration of a DID trunk. The DID number assigned in the central office should match the extension numbers of associated system extensions.

Hardware and Software Requirements

Requires one port on a TN753 DID Trunk circuit pack for each DID trunk. No additional software is required.

Direct Outward Dialing

Description

Allows voice terminal users to access the public network without attendant assistance.

The user simply lifts the handset and dials the trunk access code for the desired trunk group. The user is then connected to the public network.

Considerations

Direct Outward Dialing reduces the attendant workload and saves the user time by allowing direct access to the public network. The user does not have to call and make the request to an attendant.

Trunks to the local telephone company central office (CO), a Wide Area Telecommunications Service (WATS) serving office, or a foreign exchange (FX).

Only one CO trunk group is accessible by a single dial access code. An all-busy trunk group cannot redirect calls to another trunk group. Therefore, if more than one trunk group is provided, a separate access code must be established for each.

Interactions

Calling party restrictions (assigned by the Class of Restriction) prevent placing Direct Outward Dialing calls from the restricted voice terminal.

Administration

Direct Outward Dialing is administered on a per-system basis by the System Manager. Administration consists of assigning the various trunk groups and their associated trunk access codes.

Hardware and Software Requirements

Requires one port on a TN747 CO Trunk circuit pack for each assigned trunk. No additional software is required.

Distinctive Ringing

Description

Helps voice terminal users and attendants distinguish between various types of incoming calls.

The ringing cycle, which begins when a voice terminal or attendant console receives an incoming call, is heard by the voice terminal user or attendant. Since the ringing cycle is different for different types of calls, the voice terminal user or attendant can tell what type of call is being received and can handle the call accordingly.

The associated call types, types of users, and ringing cycles are as follows:

Associated Call Type	User	Ringing Cycle (In Seconds)
Internal voice terminal, internal tie trunk, and Remote Access	All voice terminals	1-burst ringing (1.2 on, 4.0 off repetitive)
Intercom	Single-line voice terminals	
Attendant-extended, attendant-originated, and incoming trunk, including external tie trunk	All voice terminals	2-burst ringing (0.2 on, 0.4 off; 0.6 on, 4.0 off repetitive)
Automatic Callback, Priority Calling, and Ringback Queuing callback	Single-line voice terminals	3-burst ringing (0.2 on, 0.1 off; 0.2 on, 0.1 off; 0.6 on, 4.0 off repetitive)
	Multi-appearance voice terminals	3-burst ringing (0.1 on, 0.1 off; 0.1 on, 0.3 off; 0.6 on, 4.0 off repetitive)
Intercom	Multi-appearance voice terminals	Single tone (0.6 on; 4.6 off repetitive)
Manual Signaling	Multi-appearance voice terminals	Single tone (0.2 on)
Redirection Notification	All voice terminals	Single tone (0.2 on)

The following call types and their ringing cycles are received at attendant consoles:

Call Type	Ringing Cycle (In Seconds)
Incoming call	Low-pitched tone (0.4 on, 1.2 off repetitive)
Attendant Recall call and when any call associated with a timed reminder interval returns to the console	High-pitched tone (0.4 on, 1.2 off repetitive)
Calls waiting in queue	Low-pitched tone (0.25 on, 0.8 off repetitive)

Considerations

Ringing allows the user to identify the type of incoming call. By knowing the type of incoming call, the user is able to answer each call in a suitable manner for that type of call.

The 2- and 3-burst ringing is optional only on single-line voice terminals. If Ringing is disabled, the user will hear a 1-burst repetitive tone for all incoming calls. This is useful for equipment interfaced by analog lines, especially if the Off-Premises Station feature is used.

A single distinctive ring is used for each new incoming call when a voice terminal is off-hook or a headset is being used. The CALLMASTER data communications terminal is alerted with a single ring whenever either the headset or handset is plugged into the headset jack.

Interactions

The Distinctive Ringing cycles are altered when the Personalized Ringing (V2, V3, or G1) feature is used.

Administration

Ringing is a standard system feature. No administration is required except for the following items which are set by the System Manager:

- Redirection Notification can be assigned for any voice terminal.
- Distinctive 2- or 3-burst ringing can be disabled for single-line voice terminals.

Hardware and Software Requirements

Requires a 500-type, 2500-type, or 7100-series voice terminal to be installed and connected to a TN742 Analog Line circuit pack. No additional software is required.

Do Not Disturb (V3 or G1)

Description

Allows guests, attendants, and authorized front desk voice terminal users to request that no calls, other than priority calls, terminate at a particular extension number until a specified time. At the specified time, the system automatically deactivates the feature and allows calls to terminate normally at the extension.

Do Not Disturb is a form of Termination Restriction that is associated with an automatic deactivate time. When Do Not Disturb is active at an extension number, the user will receive only those calls associated with the Automatic Callback, Automatic Wakeup, and Priority Calling features, and those calls that are redirected to that extension via the Call Coverage and Call Forwarding All Calls features. All other call attempts will redirect to a recorded announcement, an attendant, or intercept tone, as specified by the System Manager through administration.

This feature may be dial or button accessed from voice terminals equipped with touch-tone dialing or button accessed from attendant consoles and front desk terminals. Users with rotary-dial terminals must call the attendant or front desk user to request Do Not Disturb.

When the Do Not Disturb feature is activated, the system supplies voice prompting to voice terminal users and display prompting to attendants and front desk users.

Feature Activation by Voice Terminal Users

A voice terminal user can activate Do Not Disturb by dial access or by button access if a Do Not Disturb button is assigned to the voice terminal. If dial access is used, the system automatically deactivates the feature at the requested time. If button access is used, deactivation is not automatic.

- Dial Access

  When a user dials the Do Not Disturb feature access code (FAC), the system generates voice messages (through the use of a Speech Synthesis circuit pack) that direct the user to enter a deactivate time. The touch-tone buttons on the voice terminal are used to enter this information. The user may later change or delete the request by dialing the Do Not Disturb FAC again and entering the required information.

  If invalid entries are made (such as 32 for the deactivate time) or if system conditions (such as all voice synthesis ports busy) prevent entry of a Do Not Disturb request, the system informs the user to dial the attendant or front desk assistance.

- Button Access

  If a voice terminal has a Do Not Disturb button, the user can press the assigned button to activate the feature. The handset may be on-hook or off-hook; voice prompting is not required. The user must press the button a second time to deactivate the feature.

The lamp associated with the Do Not Disturb button lights when the feature is activated and remains lighted until the feature is deactivated. An automatic deactivate time is not provided through button access.

Feature Activation by Attendant

The attendant (or authorized front desk user) can activate Do Not Disturb for a user or a group of users. (The assigned Class of Restriction [COR] determines which users are in the group.) The attendant presses the Do Not Disturb—Extension or the Do Not Disturb—Group button. After pressing the Extension button, the attendant enters an extension number; after pressing the Group button, the attendant enters an appropriate COR number.

System prompts appear on the display to direct the attendant on what information to enter, and a displayed message notifies the attendant when the request is confirmed. If a Do Not Disturb request is denied, a displayed message, including a reason for the denial, informs the attendant.

The attendant can cancel a Do Not Disturb request by activating the feature, entering the desired extension number or group COR number, and pressing the Delete button.

Activation of Do Not Disturb Through a Property Management System (PMS)

The system provides an interface to a PMS. This interface can allow activation and deactivation of controlled restrictions. Activation of Do Not Disturb through a PMS is similar to that of Termination Restriction. A scheduled deactivate time is not specified.

Audit Trail Reports

The system keeps an audit trail record of all voice terminals that are in the Do Not Disturb mode and that have Termination Restriction activated. The System Manager or other delegated administration personnel can request a listing of this information to be displayed at a terminal or to be printed at a designated printer.

The following reports can be administered for printing on a daily basis:

- Do Not Disturb Status Report—This report lists all extension numbers with Do Not Disturb active. The specified Do Not Disturb deactivate time for each extension number is also listed.

- Do Not Disturb Plus COR Status Report—This report lists all extension numbers as defined above, plus a list of those extension numbers whose Controlled Restriction level is Termination Restriction. (Termination Restriction is activated by the attendant for a specific extension or COR. A deactivate time is not associated with Termination Restriction.)

Audit trail records do not include Do Not Disturb information for extensions that are both termination and outward restricted.

Considerations

The Do Not Disturb feature lessens the attendant's workload since each voice terminal user can activate the feature. Also, through the voice messages supplied by the system, the user is assured that his or her request is confirmed.

The Do Not Disturb deactivate time may be requested using standard time or military time. If standard time is entered, the system will prompt the user to enter a.m. or p.m.

Up to ten attendants or front desk users can be in the Do Not Disturb display mode at the same time. A front desk user must have console permission Class of Service (COS) in order to activate this feature.

The total number of Do Not Disturb requests combined with the total number of Automatic Wakeup requests cannot exceed 800 (V3) or 1600 (G1).

The number of available speech synthesis ports is the only limit on the number of voice terminal users receiving voice prompting at the same time.

Interactions

The following features interact with the Do Not Disturb feature.

- Automatic Callback

  Do Not Disturb does not block an Automatic Callback call. Return calls will terminate at a voice terminal in the normal way.

- Automatic Wakeup

  An Automatic Wakeup call deactivates Do Not Disturb and alerts the guest at the specified time.

- Call Coverage

  If a point in a coverage path has Do Not Disturb active, calls covering to that coverage point extension will still alert that extension.

- Call Forwarding All Calls

  If Do Not Disturb is active at the forwarding extension, the caller will receive intercept treatment. If Do Not Disturb is active at the forwarded-to extension, the call alerts the forwarded-to extension.

- Property Management System (PMS) Interface

  Check-out from either a PMS or the switch automatically deactivates Do Not Disturb for the specified extension number.

Administration

Do Not Disturb is administered by the System Manager. The following items require administration:

- Feature Access Code (one code can be used to activate and deactivate Do Not Disturb)
- Do Not Disturb button (per voice terminal, optional)
- Do Not Disturb—Extension button (per attendant console or front desk terminal)
- Do Not Disturb—Group button (per attendant console or front desk terminal)
- Intercept treatment for call attempts to a terminal with Do Not Disturb active—Choice is one of the following: intercept tone, recorded announcement, coverage, or attendant.

Hardware and Software Requirements

Requires a TN725 Speech Synthesis circuit pack if voice prompting is used. Each circuit pack has four ports to provide voice prompting.

No additional software is required.

DS1 Trunk Service (V2, V3, or G1)

Description

Provides a digital interface for the following types of trunks:

- Voice-Grade DS1 Tie Trunks
- Alternate Voice/Data (AVD) DS1 Tie Trunks
- Robbed-Bit Alternate Voice/Data (RBAVD) DS1 Tie Trunks (G1)
- Digital Multiplexed Interface (DMI) Tie Trunks
- Integrated Services Digital Network—Primary Rate Interface (ISDN-PRI) Trunks
- Central Office (CO) Trunks
- Foreign Exchange (FX) Trunks
- Remote Access Trunks
- Wide Area Telecommunication Service (WATS) Trunks
- Direct Inward Dialing (DID) Trunks
- Off-Premises Stations.

Voice-Grade DS1 Tie Trunks

The Voice-Grade DS1 tie trunks are an alternative to 4-wire analog E&M tie trunks and may be used to interface with other properly-equipped switching systems.

Voice-Grade DS1 tie trunks can also be used as the following:

- Electronic Tandem Network (ETN) or Tandem Tie Trunk Network (TTTN) tie trunks
- Main/Satellite tie trunks
- Tie trunks used to interface with Enhanced Private Switched Communications Service (EPSCS) and Common Control Switching Arrangement (CCSA) networks
- Release link trunks for Centralized Attendant Service (CAS)
- Access Trunks.

The TN722B or TN767 DS1 circuit pack is used to support Voice-Grade DS1 tie trunks in the Robbed Bit Signaling mode. The Robbed Bit Signaling mode supports 24 trunks for transmission on the circuit pack because the least significant bit ("robbed") in every sixth frame of data transmission is replaced by a signaling bit.

This type of tie trunk uses DS1 tranmission facilities in the "Domestic DS1" format, which is a 1.544-Mbps digital signal that consists of a 1.536-Mbps signal multiplexed with an 8-kbps framing signal.

Alternate Voice/Data (AVD) Tie Trunks

AVD DS1 tie trunks permit alternate voice and data calling between a System 75 or DEFINITY Generic 1 and a System 85 or DEFINITY Generic 2.1.

AVD DS1 tie trunks can be used to connect the system with other digital switches.

The TN722B or TN767 DS1 circuit pack is used to support AVD DS1 tie trunks in the Common Channel Signaling mode. The Common Channel Signaling mode supports 23 trunks for data transmission and 1 trunk for signaling purposes.

This type of tie trunk uses DS1 transmission facilities in the "Domestic DS1" format, which is a 1.544-Mbps digital signal that consists of a 1.536-Mbps signal multiplexed with an 8-kbps framing signal.

Robbed-Bit Alternate Voice/Data (RBAVD) Tie Trunks (G1)

RBAVD DS1 tie trunks permit alternate voice and data calling between a System 75 or DEFINITY Generic 1 and a System 85 or DEFINITY Generic 2.1.

RBAVD DS1 tie trunks can be used to connect the system with other digital switches.

RBAVD DS1 tie trunks can be used to access voice/data Software Defined Network (SDN) services.

For normal AVD facilities, modem pool resources are not automatically inserted. With RBAVD facilities, a data origination access code can be used to force modem pool insertion on the call.

The TN722B or TN767 DS1 circuit pack is used to support RBAVD DS1 tie trunks in the Robbed-Bit Signaling mode.

This type of tie trunk uses DS1 transmission facilities in the "Domestic DS1" format, which is a 1.544-Mbps digital signal that consists of a 1.536-Mbps signal multiplexed with an 8-kbps framing signal.

DMI Tie Trunks

For G1, DMI tie trunks use Bit Oriented Signaling or ISDN-PRI Message Oriented Signaling to interface with a host computer, another PBX, or a public or private network.

The TN722B DS1 Tie Trunk circuit pack supports Voice-Grade DS1, AVD DS1, and DMI tie trunks in the Bit Oriented Signaling modes. In addition, the TN767 DS1 circuit pack supports DMI tie trunks in the ISDN-PRI signaling mode. For details on the DMI signaling modes, see the Digital Multiplexed Interface feature description elsewhere in this chapter.

This type of tie trunk uses DS1 transmission facilities in the "Domestic DS1" format, which is a 1.544-Mbps digital signal that consists of a 1.536-Mbps signal multiplexed with an 8-kbps framing signal.

When the DS1 circuit pack is assigned ISDN-PRI signaling, Robbed Bit Signaling and ISDN-PRI signaling can be used over the same DS1 interface. Trunk groups administered with a communication type of "voice" can use Robbed Bit signaling and trunk groups administered with a communication type of "isdn" can use the ISDN-PRI signaling.

CO, FX, and WATS Trunks (G1)

When the DS1 interface (TN767 DS1 circuit pack) is used to provide CO, FX, or WATS trunk group service with incoming and outgoing types of ground start or loop start, Robbed Bit Signaling must be used.

When the DS1 interface is used to provide CO, FX, or WATS trunk group service with incoming/outgoing dial types administered as auto/auto, auto/delay, auto-immediate, or auto/wink, either Common Channel Signaling or Robbed Bit Signaling can be used.

When the DS1 interface is used to provide CO, FX, or WATS trunk group service with incoming and outgoing types of ground start or loop start, outgoing trunk calls do not receive answer supervision. Instead, the answer supervision is faked by the DS1 circuit pack.

Remote Access Trunks (G1)

Signaling for remote access trunks depends on the trunk group and incoming/outgoing dial types.

DID Trunks (G1)

When the DS1 interface is used to provide DID trunk group service, Robbed Bit Signaling must be used.

Off-Premises Stations (G1)

DS1 off-premises stations do not receive system message waiting indications.

When the DS1 interface is used to provide off-premises stations, Robbed Bit Signaling must be used.

Considerations

DS1 tie trunks offer voice and data transmission, via DS1 digital facilities, at lower cost and faster speed than conventional analog trunks. Data transmission costs are lower than if large analog trunk groups are used. In the future, digital transmission is expected to cost less thus adding to savings over analog facilities.

Each DS1 circuit pack can support up to 24 trunks: 24 trunks for transmision in the Robbed Bit Signaling mode or 23 trunks for clear data transmission, and one trunk to transmit Common Channel Signaling for the other 23 trunks in the Common Channel Signaling mode or ISDN-PRI signaling mode (G1 only).

The system can support a maximum of 20 (V2 or V3) or 30 (G1) DS1 circuit packs.

Interactions

The following features interact with the DS1 Trunk Service feature.

- Centralized Attendant Service (CAS) (V2, V3, or G1)

  AVD tie trunks and RLTs can share the same DS1 interface. RLTs cannot, however, be administered or a communication type of "avd" on the RLT Trunk screen form.

- Distributed Communications System (DCS)

  AVD DS1 tie trunks can only be used in a DCS between two System 75s or DEFINITY Generic 1s or between System 75 or DEFINITY Generic 1 and System 85 or DEFINITY Generic 2.1.

- Electronic Tandem Network (ETN)

  AVD DS1 tie trunks can only be used in an ETN between two System 75s or DEFINITY Generic 1s or between System 75 or DEFINITY Generic 1 and System 85 or DEFINITY Generic 2.1.

- Modem Pooling

  When AVD DS1 tie trunks are used, a conversion resource is not automatically inserted into the connection because the system cannot determine whether the transmission is voice or data. A conversion resource is connected between Voice-Grade tie trunks and digital endpoints.

- Private Network Access

  AVD DS1 tie trunks cannot be used as Enhanced Private Switched Communications Services (EPSCS) or Common Control Switching Arrangement (CCSA) access trunks.

Administration

DS1 trunks are assigned on a per-line basis by the System Manager. The following items require administration:

- DS1 Circuit Pack—Assign the circuit pack to the system before administration of the associated trunks.

- Synchronization Plan—Administer to provide synchronization between the switch's DS1 circuitry and the digital facilities that the switch is connected to.
- Trunk Groups—Associate the trunks to groups, if desired.

Hardware and Software Requirements

One TN722 or TN767 (G1) DS1 circuit pack is required for every 24 trunks using Robbed Bit Signaling or for every 23 trunks using Common Channel Signaling. If ISDN-PRI signaling is used, a TN765 Processor Interface circuit pack is required in addition to the TN767 DS1 circuit pack. A TN741 Tone Generator/Clock circuit pack is required to provide synchronization for the DS1 trunks.

No additional software is required.

EIA Interface (V2, V3, or G1)

Description

Provides an alternative to Digital Terminal Data Modules (DTDMs) and Modular Processor Data Modules (MPDMs), within the system hardware, for interconnection between RS-232 compatible Digital Terminal Equipment (DTE) and the system. The Electronic Industries Association (EIA) Interface consists of a Data Line circuit pack port and an Asynchronous Data Unit (ADU).

The EIA Interface supports speeds of LOW, 300, 1200, 2400, 9600, and 19,200 bps.

A data line port differs from a data module in that the functions (options) are set in the system rather than at the physical hardware. The user does not have physical access to the data line port, but has access to all data module related functions; that is, the user can examine and change such items as speed, parity, etc., via a menu-driven selection mode at the DTE. Also, the System Access Terminal (SAT) (V2 or V3) or the Manager I terminal (G1) can be used to examine and change the functions.

A data line port in conjunction with an ADU can be used to connect the system to the Information Systems Network (ISN). The ISN consists of packet data switches that support data calls between data endpoints. Data line ports provide the most economical access to the ISN. Available ADUs are the Z3A1, Z3A2, Z3A3, and Z3A4.

Digital Communications Equipment (DCE) may also be connected to a data line port by use of a null modem.

Considerations

The system's EIA Interface support offers a convenient and lower cost alternative to data modules. DTEs can connect directly to a Data Line circuit pack that functions as a data module connected to a Digital Line port. Since the user does not have physical access to the data module, all related data module options are settable from the DTE. With a density of eight data line ports per circuit pack, each port provides connections of user's asynchronous EIA RS-232 compatible DTE.

There is no limit to the number of Data Line circuit packs the system can support, subject to slot availability and the system limit of digital data endpoints.

Flow control signaling is not provided.

Interactions

The following features interact with the EIA Interface feature.

- Data Hot Line

  Data Terminal (Keyboard) Dialing permission must be granted before Keyboard Dialing can be accessed.

- Data Terminal (Keyboard) Dialing

  Access to ISN endpoints requires 2-stage dialing: the first stage consisting of dialing a hunt group extension number to access ISN, then the second stage consisting of an ISN address.

Administration

EIA Interface support is assigned on a per-data terminal basis by the System Manager. The following items require administration:

- Data Line circuit pack—Assign a vacant port, port options, and permissions on the circuit pack to the associated DTE.
- Data Extension buttons—Assign Data Extension buttons to multi-appearance voice terminals.

The following permissions can be administered on a data line port to allow DTEs to be used.

- Keyboard Dialing (KYBD)—Must be set to allow data endpoints to receive and send text during data call origination or termination. Text prompts are provided.
- Configuration—Must be set to allow DTEs to change their data module options; that is, examine and change options, such as speed, from the DTE. Keyboard Dialing permission must be granted first.
- Busy Out—Should be set for DTEs that are members of a hunt group, and to allow "busy out" (when DTE turns power off) so that calls will not terminate on that DTE.

The following options can be examined and changed from the DTE if the "configuration" permission has been granted:

- Speed—All speeds (up to 19.2 kbps) at which the DTE can operate are selectable, including Autoadjust. Autoadjust is the capability of the data line port to determine what speed and parity the associated DTE is transmitting at and match it for terminal dialing and/or text feedback purposes.
- Parity—All choices of parity (even, odd, mark, or space) can be selected.
- Permit Mismatch—The EIA Interface may be operated at a higher transmission speed rate than the rate between the Data Line circuit and the far end data module. This allows for calls between digital endpoints with different speeds without changing the speed of the DTE.

- Dial Echoing—Can be set to echo typed characters back to the DTE during dialing.
- Disconnect—Set the signal to indicate "disconnect." Choices are one break greater than 2 seconds or two breaks within 1 second.
- Answer Text—Can be selected when DTE is an intelligent device, to allow text messages to be delivered to the DTE when a call is being answered; also, applies to text generated by the data line circuit and received from the system. The following call progress messages may be answered:
  - INCOMING CALL
  - PLEASE ANSWER
  - TRANSFER
  - FORWARDED
  - ANSWERED
  - ABANDONED
  - DISCONNECTED
  - OTHER END
- Connected Indication—Can be set to allow the text "CONNECTED SPEED= XXXX" to be sent to the DTE when the data call has been established.
- Other Charateristics—The Data Line circuit always operates in automatic answer (provided the DTE is on), asynchronous, and full duplex modes. The "Loss of Carrier Disconnect" is set to off; that is, the Data Line circuit, unlike other data modules, does not disconnect upon loss of EIA updates in the previous 4 seconds.

Hardware and Software Requirements

One TN726 Data Line circuit pack is required for each eight EIA interfaces provided. One ADU is required for each port on the circuit pack.

No additional software is required.

Emergency Access to the Attendant (V3 or G1)

Description

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by the system or can be dialed by system users. Such calls can receive priority handling by the attendant.

Emergency calls to the attendant can be placed in the following ways:

- Automatically by the system

  If a voice terminal has been assigned the Off-Hook Alert option via Class of Service, an emergency call is automatically placed to the attendant when a voice terminal is in the off-hook state and intercept time-out occurs. (The off-hook alert time-out can be changed by the System Manager.)

- Dial access by a system user

  A system voice terminal user can place an emergency call to the attendant by dialing the Emergency Access to the Attendant feature access code.

When an emergency call is placed, one of the available attendant consoles receives visual and audible notification of the call. However, if all attendants are busy, the call enters a unique queue for emergency calls. This queue allows attendants to handle emergency calls separately from other calls. If the queue is full, the call, if administered to do so, can be redirected to another extension.

An emergency call causes the following to occur:

- The system selects the first available console to receive the call, even if the call first entered the emergency queue.
- The Emergency tone alerts the selected attendant and the lamp associated with the Emergency button, if assigned, lights at that attendant console. If the console is an older console (does not have emergency tone capability), normal ringing is heard and the display flashes.
- When the call arrives at an available console, the attendant display shows the following:
  - The call appearance that received the call
  - The calling party identification
  - The calling party extension number
  - The number of emergency calls remaining in queue.

An attendant can place a normal call on hold in order to receive an emergency call.

An audit record is created for each emergency call event. This record includes the following:

- The extension number and name where the call was originated
- The extension number of the attendant or attendant group that answered the call
- The time the call was originated
- One of the following call results:
  - — Complete
  - — Emergency queue full
  - — Abandoned
  - — Emergency night.

The emergency audit records are used to generate an Emergency Access Activity Report. This report summarizes each emergency access call that was attempted during the past 24 hours. This report can be scheduled for printing once a day at a designated printer. Also, if the system has a journal printer, Emergency Access to the Attendant events are printed as they occur.

The System Manager can monitor emergency access call events by displaying them at the administration terminal. The command for listing emergency call events is **list emergency.** Also, a "from" and "to" time option can be used with the command. For example, if the command **list emergency from 8:00 a.m. to 12:00 p.m.** was entered, all emergency call events will be printed at a designated printer.

Considerations

Emergency Access to the Attendant provides a way for users to quickly and easily get in touch with an attendant. This results in more efficient call handling for users.

The Emergency Access to the Attendant queue can contain a maximum of 50 calls. When the emergency queue is full, any overflow should be redirected to an extension number.

The Emergency button on the console has no function other than to provide a visual indication of an incoming emergency call.

The unique Emergency tone cannot be silenced except by answering the emergency call.

The system must have at least one attendant before this feature can be activated.

Interactions

The following features interact with the Emergency Access to the Attendant feature.

- Centralized Attendant Service (CAS)

  If the system is a branch location and if CAS is in effect, an emergency call will be rerouted to the branch attendant group. If the branch does not have an attendant or if the branch is not in CAS Backup Service, the call will be denied.

  If the Branch PBX is in CAS Backup Service, an emergency call will route to the backup position and will be treated the same as any other non-emergency call.

- Class of Restriction

  If the calling voice terminal is assigned Origination Restriction, an emergency call attempt will be denied. However, an Emergency Access to the Attendant call will override any other calling party restrictions, including any controlled restrictions.

- Individual Attendant Access

  An Emergency Call to the Attendant cannot be placed to an individual attendant.

  An Emergency Call to the Attendant does not have priority over a call to an individual attendant.

- Intercept Treatment

  The Intercept With Off-Hook Alert option automatically activates Emergency Access to the Attendant.

- Inter-PBX Attendant Service

  If the system is a branch location and if Inter-PBX Attendant Service is in effect, an emergency call will be rerouted to the local Branch PBX attendant group. If the branch does not have an attendant or if the attendant is not on duty, the call will be denied.

- Night Service

  When Night Service is in effect, Emergency Calls to the Attendant route to the night destination. Such calls are included on the Emergency Audit Record, and the call will be designated as "Emergency Night" in the audit trail.

- Remote Access

  An Emergency Call to the Attendant cannot be placed through Remote Access.

- Restriction—Controlled

  An Emergency Access to the Attendant call will override any Controlled Restriction.

Administration

The Emergency Access to the Attendant feature is optional on a per-system basis. The following items require administration by the System Manager:

- Emergency feature access code
- Emergency button and associated lamp (per attendant console)
- Emergency queue length (up to a maximum of 50)
- Permission to activate Emergency Access to the Attendant via off-hook alert (per Class of Service)
- Extension number where emergency queue overflow will redirect
- Interval the intercept tone is applied before the emergency call is placed.

In addition, the journal printer and the time for the system to print the Scheduled Emergency Access Activity Report are administered through the Feature-Related System Parameters form.

Hardware and Software Requirements

No additional hardware or software is required.

Facility Busy Indication

Description

Provides multi-appearance voice terminal users with a visual indication of the busy or idle status of an extension number, a trunk group, terminating extension group, a hunt group (Direct Department Calling or Uniform Call Distribution group), or any loudspeaker paging zone, including all zones. The Facility Busy Indication button provides the voice terminal user direct access to the extension number, trunk group, or paging zone.

When the lamp associated with the Facility Busy Indication button is lighted, the tracked resource is busy. If the lamp is dark, the resource is idle. If the lamp is flashing, the tracked resource is placing a call to the voice terminal with the button.

Pressing the Facility Busy Indication button automatically selects an idle call appearance and places a call to the resource.

Considerations

With Facility Busy Indication, a user can monitor the busy or idle status of a frequently called extension number. By knowing when the monitored facility is busy or idle, the user can wait until the facility is idle to make a call. This reduces the time spent trying to call busy facilities.

A maximum of 1000 (V1), 1600 (V2 or V3), or 3200 (G1) Facility Busy Indication buttons are allowed in the system, and as many as 100 of these buttons can be administered to track the same resource. A new state of the tracked resource (that is, a change from idle to busy) is updated within 5 seconds after the system detects the change.

Extension numbers, trunk group access codes, and Loudspeaker Paging Access codes can be stored in a Facility Busy Indication button. However, an access code followed by other numbers cannot be stored.

It is possible that an incoming call which causes the lamp to flash may go unanswered. If the lamp represents the status of a trunk group and all trunks in the trunk group become busy before the flashing call is answered, the system software lights the lamp steadily to indicate that all trunks are busy. When a trunk in that trunk group becomes idle, the system software turns off the busy indication and the lamp goes dark. Therefore, the lamp flashes, lights steadily, and goes out while the call has neither been answered nor dropped.

The Facility Busy Indication cannot monitor the status of the attendant console.

Interactions

None.

Administration

Facility Busy Indication is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign the Facility Busy Indication button to a voice terminal or attendant console.

Hardware and Software Requirements

No additional hardware or software is required.

Facility Restriction Levels and Traveling Class Marks (V2, V3, or G1)

Description

Provides up to eight levels of restriction for users of the Automatic Alternate Routing (AAR) and/or Automatic Route Selection (ARS) features.

Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs) provide a method of allowing certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed toll calling only to other corporate locations. Similarly, certain users may be allowed toll calling into more areas than other users. International calling may be denied to all except a few users.

FRLs and TCMs are transparent to the user. Appropriate values are predetermined and programmed into the system. Dialing procedures are unaffected.

Call routing for each call is determined by the dialed Area Code and/or office code (either public or private network). Translation on the first three or six digits of the called number yields one of 254 Routing Patterns, numbered 1 through 254. More than one translation can point to the same pattern. Pattern 0 provides intercept treatment and is used for unassigned private network office codes. Each Routing Pattern contains up to six routing preferences. Each preference includes the following information:

- Trunk Group Number
- Minimum FRL required to access the trunk group.

Routing preferences are listed in ascending FRL order.

Each facility, such as a trunk or voice terminal, capable of originating a call also has an associated FRL. Whether a given call is allowed or not depends on two things: compatibility between FRLs and availability of an idle trunk.

Compatibility is determined by a comparison of the minimum FRL associated with the trunk group and the originating-side FRL. Either can have a value of 0 through 7. Access to the associated trunk group is permitted if the originating-side FRL is greater than or equal to the minimum FRL. Note that lower originating-side FRLs can access fewer routing preferences, whereas lower minimum FRLs permit greater access. Stated another way, a 0 originating-side FRL is the most restricted and a 7 is the least restricted. A 0 minimum FRL is the least restrictive, and a 7 is the most restrictive. Compatibility checking begins with the first-choice route (the first one in the pattern). Assuming access is permitted, availability is checked; that is, is there an idle trunk in the group? If so, the call continues. If not, compatibility is checked on the next routing preference.

If the compatibility check fails on the first-choice route, intercept tone is returned to the user. This call will always fail and need not be retried. If the compatibility check fails on the second or subsequent routing preference, or if all accessible trunk groups are busy, the call may queue on the first choice trunk group. (See Ringback Queuing elsewhere in this chapter for details.)

If the trunk group selected for a call is an intertandem tie trunk group, then a TCM is outpulsed as the last digit. A TCM is equivalent to the originating-side FRL. At the next tandem switch, compatibility and availability checking are done, as before. In this case, the FRL assigned to the incoming intertandem tie trunk group is used as the originating-side FRL. However, if this fails to yield a route and if the TCM is higher than the tie trunk FRL, then the TCM is used in another attempt to complete the call.

Call Originating Facilities

At a switch serving as the call origination point, any of the following can be the originator of an ARS or AAR call:

- Voice terminal
- Remote Access user
- Attendant
- Incoming tie trunk group from a subtending location
- Data terminal capable of Keyboard Dialing.

At a tandem switch, either of the following can be the originator of an ARS or AAR call:

- Incoming Intertandem tie trunk group
- Incoming Access tie trunk group—links a remote main switch to a tandem switch.

Each of these facilities is assigned an FRL via an associated Class of Restriction (COR), either directly or indirectly.

Voice terminals and all incoming tie trunk groups use the FRL contained within the assigned COR. Attendants use the FRL contained within the COR assigned to the attendant group. Data terminals use the FRL contained within the COR assigned to the associated data module.

The Remote Access feature can be accessed via a Direct Inward Dialing (DID) trunk group, tie trunk group, dedicated central office trunk group, 800 Service trunk group, and/or dedicated foreign exchange trunk group. In the absence of a Remote Access Barrier Code, the applicable FRL is contained in the COR assigned to the trunk group. If a Barrier Code is required on Remote Access calls, the applicable FRL is contained in the COR assigned to the Barrier Code.

Call Terminating Facilities

Any of the following trunk types can serve as the termination point for an ARS or AAR call:

- Tie trunk—excluding release link trunks (RLTs), but including Common Control Switching Arrangement (CCSA) and Enhanced Private Switched Communications Service (EPSCS) Access trunks

- Wide Area Telecommunications Service (WATS)
- Local central office (CO)
- Foreign exchange (FX).

Each of these outgoing trunk groups has an assigned COR that contains an FRL. However, this FRL is never used. Terminating-side FRLs are assigned in the Routing Pattern, not to the outgoing trunk group.

Considerations

FRLs provide the means to restrict certain users from placing selected calls while allowing other users to place the same calls.

Originating-side FRLs are assigned via the COR of the originating-side facility, such as an incoming tie trunk group or voice terminal. If an FRL is not assigned, the system assumes an FRL of 0 for all originating facilities except the attendant group. An FRL of 7 is assumed for the attendant group.

A COR is also assigned to each trunk group. If the COR specifies an FRL, the FRL is ignored. The minimum FRL specified in the Routing Pattern is the only FRL used on the terminating side of the call.

On attendant-extended calls, the attendant group FRL is used rather than the FRL of the calling party.

Interactions

FRLs apply only on ARS and AAR calls.

If Station Message Detail Recording (SMDR) 15-digit account codes are used, the FRL field in the SMDR record is overwritten.

Administration

FRLs are assigned by the System Manager as a part of ARS and/or AAR administration. Originating FRLs are assigned on a per-COR basis. Terminating FRLs are assigned on a per-Routing Pattern basis. TCMs do not require assignment.

Assignment Guidelines

The FRL assigned to the facility answering a call is not checked. Terminating-side FRLs apply to trunk groups only. This simplifies assignments. At each switch, the trunk groups available to handle a given call must be listed in the preferred order within the Routing Pattern. The most-preferred choice must be at the top of the list. Up to six choices can be specified. Now the relative value of access to each of the listed trunk groups must be determined. This, of course, is specified via an FRL. On a scale of 0 through 7, the relative value is determined and assigned. Decisions are normally based on the cost of using the facility, although other criteria can be used. The same FRL value can be assigned to more than one trunk group if there is no reason to prefer one trunk group over the other.

If there will be users within the system who are not allowed to make outside calls, use some value other than 0 as the value for the first-choice trunk group. By assigning these users an FRL of 0, none of the trunk groups can be accessed (since all trunk group FRLs are greater than 0). Such calls are denied.

Each Routing Pattern must be individually constructed. The same trunk group can be used in more than one pattern. The associated FRL is assigned within the pattern and is not associated with the trunk group itself. The same trunk group can have a different FRL in a different pattern.

Be consistent in FRL assignments. Do not use a range of 0 through 5 in one pattern and a range of 2 through 7 in another pattern if all users can access the first-choice route. Admittedly, the trunk group with an FRL of 2 may be more expensive than the trunk group with an FRL of 0, but there is no real reason to assign a 2 to a trunk group that everyone can access. For ease of assignments, always use a 0 for such a trunk group.

There should be a COR established for each FRL used in a Routing Pattern. The appropriate COR is then assigned to the users who can access the routes restricted by the FRL value. For example, a middle executive might be able to access all routes with an FRL of 5 or lower, whereas the president can access all routes. In this case, the executive is assigned a COR with an FRL of 5 and the president is assigned a COR with an FRL of 7.

Remote Access users can access the system's features and services the same as an on-premises user. FRL assignment is via Remote Access Barrier Codes. Up to ten Barrier Codes, each with its own COR (and FRL), can be assigned. Although the COR defines other restrictions, ten Barrier Codes are enough to also provide a range of FRL assignments. Assignment of Barrier Code FRLs is the same as if the user were on-premises. The simplest way to assign these FRLs is to duplicate the on-premises FRLs, then merely relate the appropriate Barrier Code to those that will be using Remote Access.

Hardware and Software Requirements

No additional hardware is required. The optional ARS software is required with Version 1 systems. The optional Private Network Access or ARS software is required with Version 2, Version 3, or DEFINITY Generic 1 systems.

Facility Test Calls

Description

Provides a voice terminal user with the capability of making test calls to access specific trunks, touch-tone receivers, time slots, and system tones. The test call is used to make sure the facility is operating properly. A local voice terminal user can make a test call by dialing an access code. An Initialization and Administration System (INADS) terminal user can also make test calls.

Four types of Facility Test Calls can be made:

- Trunk test call

  Accesses specific tie or central office (CO) trunks. Direct Inward Dialing (DID) trunks cannot be accessed.

- Touch-tone receiver test call

  Accesses and tests the four touch-tone receivers located on a Tone Detector circuit pack.

- Time slot test call

  Connects the voice terminal user to a specific time slot located on the Time Division Multiplex (TDM) buses or out-of-service time slots.

- System tone test call

  Connects the voice terminal user to a specific system tone.

Considerations

If a user has a problem with a specific system facility, Facility Test Calls can be used to test that facility for proper operation.

A touch-tone voice terminal must be used to make test calls.

> **Note:** AT&T has designed the Facility Test Calls feature incorporated in this product that, when properly administered by the customer, will enable the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of test call features. In such events, applicable tariffs require that the customer pay all network charges for traffic. AT&T cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

Interactions

None.

Administration

Facility Test Calls is administered on a per-system basis by the System Manager. The only administration required is to assign the Facility Test Calls access code.

Hardware and Software Requirements

No additional hardware or software is required.

Forced Entry of Account Codes (V2, V3, or G1)

Description

Requires users to dial an account code when making certain types of outgoing calls. The conditions under which dialing of account codes is required depends on system administration.

Forced Entry of Account Codes can be assigned for any of the following:

- All Toll Calls

  Toll Calls are defined as those calls which have a 0 or 1 as one of the first two digits of the called number, except service calls (for example, 911 and 411), directory assistance calls, and 800 Service calls.

  This affects all calls made by Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), or trunk access codes (TACs).

- Toll Calls Made By Users With a Specific Class of Restriction (COR)

  If forced entry of account codes is assigned to a specific COR, any voice terminal assigned that COR must dial an account code before making toll calls.

- All Calls Made on a Trunk Group With a Specific COR

  Any trunk group that is assigned a COR with forced entry of account codes cannot be accessed until an account code is dialed. If a call is being routed via AAR or ARS, account code checking is not done on the trunk group's COR.

With Version 3 or DEFINITY Generic 1, any time an account code is required and the user does not enter an account code, intercept tone is heard. An account code is never required for the following:

- Attendant originated call
- Busy verification of a trunk by an attendant or voice terminal user
- Distributed Communications System (unless required by the trunk group's COR)
- Personal Central Office Lines
- Remote Access Without Barrier Codes
- Trunk-to-Trunk Connections.

For details on how account codes are used, see the SMDR Account Code Dialing feature description elsewhere in this chapter.

Considerations

Forced Entry of Account Codes, by requiring account codes to be dialed on specific calls, provides an easy method of allocating the costs of specific calls to the correct project, department, etc. Call information is recorded by the Station Message Detail Recording feature for this purpose.

Account Code length can be up to 15 digits.

The validity of the entered account codes cannot be checked by the system.

Interactions

The following features interact with the Forced Entry of Account Codes feature.

- Automatic Alternate Routing and Automatic Route Selection

  If a trunk group is accessed via AAR or ARS, the trunk group's COR is not used to determine if an account code needs to be entered.

- Busy Verification of Terminals and Trunks

  An attendant or voice terminal user is never required to enter an account code when making a busy verification.

- Call Forwarding All Calls

  If a user is required to enter an account code to call a particular destination, the calls cannot be forwarded to that destination.

- Last Number Dialed

  The SMDR access code and account code dialed are stored as part of the Last Number Dialed. However, some digits may be lost due to the limit on the number of digits stored for this feature.

- Station Message Detail Recording (SMDR)

  SMDR does not record the correct account code if the length of the account code is changed during an active call. For example, if the account code length is 5, a user dials 12345, and the account code length is changed during the call to 2, the SMDR record shows only the first two digits (12) of the account code.

Administration

Forced Entry of Account Codes is administered by the System Manager. The following items require administration:

- Whether or not all toll calls require account code entry (per system)
- Whether or not each COR requires account code entry.

Hardware and Software Requirements

No additional hardware is required. Optional SMDR Account Code Dialing software is required.

Generalized Route Selection (G1)

Description

Generalized Route Selection (GRS) provides the customer voice and data call routing capabilities to select not only least cost routing, but also optimal routing over the appropriate facilities.

The feature maximizes the chance of using the right facility to route the call. Also, if an endpoint incompatibility exists, it provides a conversion resource (such as Modem Pools) to attempt to match the right facility with the right endpoint.

GRS allows customers to use separate routes for voice and data calls. GRS also provides the opportunity to integrate voice and data on the same trunk group, thereby providing certain economies.

GRS is a capability built on the current Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) features. In AAR or ARS, routing is based on the dialed number, the Facility Restriction Level (FRL) of the call originator, the partitioning group number, and the time-of-day. GRS enhances AAR and ARS by providing additional parameters in the routing decision.

The current AAR/ARS networking software is not designed to handle all type calls. For example, voice and voice-grade data calls may be routed over analog or digital facilities, but high speed data calls require exclusively digital facilities. GRS allows the system to use the Integrated Services Digital Network—Primary Rate Interface (ISDN-PRI) call-by-call selection of public network services. It also provides interworking between ISDN-PRI and non-ISDN-PRI entities.

ISDN-PRI interworking is the mixture of ISDN-PRI trunks and non-ISDN-PRI trunks in a call. A mixture of these signaling procedures is required to provide end-to-end connectivity when different type trunking facilities are used.

ISDN-PRI services add four additional routing parameters which are specified on each trunk group preference of the routing pattern. These parameters are:

- BCC—Identifies the type of call, such as voice calls and different type data calls.
- Network Specific Facility—Identifies the services and features to be used to complete a call.
- Band—Identifies the OUTWATS band. Wide Area Telecommunications Service (WATS) is a voice-grade service providing both voice and low speed data transmission calls to defined areas (bands) for a flat rate charge.
- Inter-Exchange Carrier (IXC)—Identifies the specific common carrier, such as AT&T, to be used for a call.

In GRS, there are a number of Bearer Capability Classes (BCCs). Customers may specify routing for each BCC according to their particular transmission needs.

Bearer Capability Classes

The BCCs are the mechanisms by which specialized routing is provided for the various type data calls and voice calls.

Each trunk group preference in the AAR/ARS routing patterns contains a BCC parameter. When a call is originated, a route is selected based on the BCC of the originating facility. BCCs are used to classify the type of traffic permitted on this trunk in the outgoing direction. Details on how a trunk group preference is determined are given in the GRS Operation section of this description.

A set of ISDN-PRI bearer capability and low-layer compatibility parameters are defined by a BCC. These are parameters of a call origination facility.

The system will determine the originating facility's BCC from one of the following:

- From the type of endpoint (the system automatically determines the BCC as shown in Table 3-C on the next page).
- From the administered value of the incoming trunk
- From the ISDN-PRI bearer capability and low-layer compatibility parameters, if the call is an ISDN-PRI trunk-originated call.

The GRS capability will recognize one or more of the following five BCCs for each trunk group preference in the routing pattern (DCP/DMI mode is explained later):

BCC	Type	DCP/DMI Mode
0	Voice-Grade Data and Voice	None
1	56 kbps Data (Mode 1)	1
2	64 kbps Data (Mode 2)	2
3	64 kbps Data (Mode 3)	3
4	64 kbps Data (Mode 0)	0

Table 3-C. BCC Assignment

Endpoint	Voice/ Data Mode	BCC	Comments
Voice Terminal	Voice	0	
Data Line Circuit Pack	2	2	
Voice Data Set	2	2	
Modular Processor Data Module	0,1,2	1,2,4	See Note.
Modular Trunk Data Module-M1	1	1	For ACCUNET Switched 56 kbps Digital Service
Modular Trunk Data Module	2	2	
Digital Terminal Data Module	2	2	
510D Personal Terminal	2	2	
Digital Communications Protocol Interface (v1)	0,2	2,4	See Note.
Digital Communications Protocol Interface (v2)	2	2	
3270T Data Module	3	3	
3270C Data Module	3	3	
3270A Data Module	2,3	2,3	See Note.

Note: For endpoints capable of operating in multiple data modes, the switch automatically determines its current operating mode when the data module originates. Before any call is originated, the default is Mode 2.

Since call origination from a data module determines the mode to be used on the call, it is recommended that the data module user press the Originate/Disconnect button once after changing data options. This way, the right mode is sure to be assigned to the next call.

ISDN-PRI BCC Parameters

ISDN-PRI BCC parameters areas follows:

A. Information Transfer Capability

The information to be transferred (or type of call) requires different transmission facilities. For example, transmission needs for voice calls and data calls are generally different. Voice and voice-grade data calls can be sent over analog trunks, while high speed data calls require digital trunks.

The Information Transfer Capability parameter can have the following four values:

- Voice (speech)
- Voice-grade data (3.1 kHz transmission)
- Unrestricted digital transmission
- Restricted digital transmission.

More than one Information Transfer Capability can be supported by one BCC. (See Table 3-D.)

Table 3-D. Assignment of BCC Based on Information Transfer Capability

DCP/DMI Mode	Information Transfer Capability	BCC	Comments
—	Speech 3.1 kHz	0	Used for Voice/ Voice Grade Data.
M1	Unrestricted/ Restricted Digital	1	Used for Mode 1 Data (56 kbps).
M2	Unrestricted/ Restricted Digital	2	Used for Mode 2 Data (asyn data speed up to 19.2 kbps).
M3	Unrestricted/ Restricted Digital	3	Used for Mode 3 Data (64 kbps).
M0	Unrestricted/ Restricted Digital	4	Used for Mode 0 Data* (64 kbps clear channel).

* Use BCC 4 for an unknown data mode that requires 64 kbps channel.

B. Low-Layer Compatibility

The low-layer compatibility information element provides remote compatibility checking. This element is used with the bearer capability element and determines the mode of the originating caller. The low-layer compatibility information element is sent only in case of data calls.

C. DCP/DMI Mode

The Digital Communications Protocol (DCP) and the Digital Multiplexed Interface (DMI) modes are data parameters of the originating data facility. These modes are not applicable to voice.

The mode values (0, 1, 2, and 3) are administered for data and Alternate Voice/Data (AVD) non-ISDN-PRI trunk groups. These mode values determine the BCC of the trunk groups.

Determination of BCC at Tandeming or Terminating System

Determination of the BCC for an incoming call from an ISDN-PRI trunk to a tandem or terminating switch is based on the BCC parameters received on the signaling channel (D-channel) of the trunk.

Determination of the BCC for an incoming call from a non-PRI trunk will be as follows:

- If the incoming trunk is a voice trunk, then the BCC is defaulted to 0.
- If the incoming trunk is a data or AVD trunk, then the BCC is administrable.

GRS Operation

The AAR/ARS routing pattern will contain an indication for each trunk group preference showing which BCC or BCCs can use that trunk group. A trunk group preference may have more than one BCC.

GRS uses a "look-ahead" algorithm when determining which preference in a routing pattern to choose. GRS first attempts to find an exact match between the originator's BCC and the corresponding allowed BCC for any of the preferences in the routing pattern. Therefore, if preference 1 does not have an exact match (even though there are available members in preference 1), it will be skipped over if a subsequent preference in the same pattern has an allowed BCC that exactly matches the originator's BCC.

As an example of how GRS chooses a trunk group preference, assume preference 1 in a pattern has BCC 0 and BCC 2 set to yes, while preference 2 has BCC 1, BCC 3, and BCC 4 set to yes. A voice or Mode 2 data call accessing this pattern will use the first preference, while a Mode 1, Mode 3, or Mode 0 data call will use the second, independent of the availability of trunks in the first preference.

When an exact match is not found in any of the routing pattern preferences, calls are treated as follows:

- Calls With an Originating BCC of 0:

  A BCC 0 originated call (such as voice or analog modem) will not be denied routing by GRS, even if the routing pattern lacks a preference with BCC 0 set to "yes". This allows the user to use voice transfer to data when making a data call, without the need for data preindication.

  If a BCC 0 originated call accesses a routing pattern for which no preference has BCC 0 set to "yes", then GRS will choose a preference with BCC 2 set to "yes", if one exists. If none exists, the next preferred order would be a preference with BCC 1 set to "yes", followed by BCC 3, and finally, BCC 4. Since each preference must allow at least one BCC to be passed, a BCC 0 (voice) originated call will never be blocked by GRS. The call is of course still subject to other restrictions, such as FRL restrictions.

- Calls With an Originating BCC of 2:

  If a BCC 2 originated call accesses a routing, pattern for which no preference has BCC 2 set to "yes", then GRS will choose a preference with BCC 0 set to "yes", if one exists. If none exists, the call will be blocked with intercept treatment.

- Calls With an Originating BCC of 1, 3, or 4:

  A DCP/DMI Mode 0 (BCC 4), Mode 1 (BCC 1), or Mode 3 (BCC 3) originated call requires an exact match on at least one preference in a routing pattern in order for GRS to allow the call to complete. For example, a Mode 1 originated call will complete only if the accessed routing pattern has a preference with BCC 1 set to "yes".

When an ISDN-PRI trunk group preference is accessed, the BCC information encoded and sent in the outgoing ISDN SETUP message to the distant-end is determined as shown below. The BCC information sent to the far end is important, because the BCC information that the far end receives in the SETUP message will become the originating BCC for the far-end's incoming trunk call.

- If an exact match of the originator's BCC has been found, then that BCC is encoded and sent in the ISDN SETUP message to the far end.

- If an exact match is not found, but the call is allowed to proceed, then the BCC encoded in the SETUP message sent to the far end is that of the routing pattern. For example, if a BCC 2 (for example, DTDM) endpoint originates a call that accesses a pattern that has one preference with only BCC 0 set to "yes", then the switch automatically inserts a modem pool for this call. In effect, the modem pool is converting BCC 2 to BCC 0. The far end cannot distinguish this call from a BCC 0 originated call that has no modem pool inserted. Therefore, BCC 0 is sent in the SETUP message. This may in turn determine routing decisions by the far end. Additional routing decisions are made as shown in the following tables.

1. BCC determination on calls from endpoints to ISDN-PRI trunks:

Calls from Endpoints to ISDN-PRI Trunks					
Originating BCC	Chosen BCC from the Routing Pattern				
	BCC 0	BCC 1	BCC 2	BCC 3	BCC 4
BCC 0	P	PT	PM	PT	PT
BCC 1	B	P	B	B	B
BCC 2	PM	B	P	B	B
BCC 3	B	B	B	P	B
BCC 4	B	B	B	B	P

B Block the call with intercept treatment.

P Allow the call and send the originating endpoint’s BCC in the SETUP message.

PT Allow the call and send the BCC chosen from the routing pattern in the SETUP message.

PM Insert a pooled modem for the call and send the BCC chosen from the routing pattern in the SETUP message.

2. BCC determination on calls from trunks to ISDN-PRI trunks:

Calls from Trunks to ISDN-PRI Trunks					
Originating BCC	Chosen BCC from the Routing Pattern				
	BCC 0	BCC 1	BCC 2	BCC 3	BCC 4
BCC 0	P	PT	PT	PT	PT
BCC 1	B	P	B	B	B
BCC 2	PT	B	P	B	B
BCC 3	B	B	B	P	B
BCC 4	B	B	B	B	P

B Block the call with intercept treatment.

P Allow the call and send the incoming trunk's BCC in the SETUP message.

PT Allow the call and send the BCC chosen from the routing pattern in the SETUP message.

The system does not insert pooled modem for any interworking trunk-to-ISDN-PRI trunk calls. The BCC of an incoming trunk is determined as follows:

ISDN-PRI Trunk — BCC is in the received SETUP message.

AVD Trunk — BCC is the BCC value administered on the trunk group form.

Data Trunk — BCC is the BCC value administered on the trunk group form.

Voice Trunk — BCC is 0.

3. BCC determination on calls from ISDN-PRI trunks to endpoints (GRS not involved):

Calls from ISDN-PRI Trunks to Endpoints (GRS is not involved)					
Originating BCC	Terminating Endpoint BCC				
	BCC 0	BCC 1	BCC 2	BCC 3	BCC 4
BCC 0	P	P	PM	P	P
BCC 1	P	P	P	P	P
BCC 2	P	P	P	P	P
BCC 3	P	P	P	P	P
BCC 4	P	P	P	P	P

P — Allow the call, and (1) if it is a voice originated call, let the calling user decide whether the terminating endpoint is the correct endpoint or not based on audible feedback (for example, data tone), or (2) if it is a data call, the data handshake procedure will establish or drop the call based on the compatibility of the endpoints.

PM — Insert a pooled modem and terminate the call to the endpoint.

Considerations

ACCUNET Digital Service

The system will be able to use ARS tables to route calls to this network. BCC 1 is a 56 kbps service. If an ACCUNET 56 kbps Digital Service trunk group is in a routing pattern that uses GRS, BCC 1 should be set to "yes".

Integrated Access on Non-ISDN-PRI Trunks

The T1 carrier access to the AT&T serving office will allow sharing of the same trunk group for voice and data calls. For example, the same trunk group may carry both voice calls (requires a BCC of 0) and Mode 1 data calls (requires a BCC of 1). This situation requires the trunk group preferences in the ARS routing pattern to be administered with a "yes" for both BCC 0 and BCC 1.

Voice and Voice-grade Data Calls

Voice and voice-grade data calls cannot be routed separately, because they are assigned the same BCC.

Signaling

The Signaling considerations are as follows:

A. Mixed Message Oriented Signaling (MOS)/Robbed Bit Signaling

The system will support a mixed environment of trunk signaling on a single DS1 interface. (DS1 is a transmission facility and is also commonly known as a T1 carrier.)

The mixture of A/B Robbed Bit Signaling and ISDN-PRI Signaling can be combined on a single DS1 interface.

(Robbed-Bit Signaling utilizes or "robs" the least significant bit in every sixth frame of a particular channel and replaces it with a signaling bit.)

Trunk group administration will be used to determine the signaling type of each channel. The signaling channel (D-channel) of a DS1 ISDN-PRI interface will be the 24th channel, and this is not available for Robbed Bit Signaling. In addition, a DS1 may be administered exclusively to provide Bit Oriented Signaling (BOS) or Robbed Bit Signaling.

(MOS, also called ISDN-PRI signaling, is a signaling technique where the signaling information is in the form of messages and BOS is a signaling technique where the signaling information is organized as A and B bits.)

B. Over ISDN-PRI Trunks

If ISDN-PRI trunks are chosen to tandem the call, the parameters that are received from an originating ISDN-PRI trunk will be sent to the next PBX via ISDN-PRI trunks and are used to determine the BCC.

If the originating facility is non-ISDN-PRI, the BCC is determined as follows:

- Automatically by the switch for voice or analog trunks
- By administration for data and AVD trunks.

C. Over Non-ISDN-PRI Trunks

If non-PRI trunks (such as analog or digital Robbed-Bit Signaling or DMI-BOS) are selected, the BCC cannot be sent. In this case, the next PBX will determine the BCC.

D. Inter-Exchange Carrier

The Inter-Exchange Carrier (IXC) access will allow access to other common carriers. Each common carrier or network provider has a separate unique network identifier (IXC). For ISDN-PRI calls, the IXC selected for routing the call is administered in the IXC field of routing preference for each of the routing patterns. If the IXC code is not administered, then Generalized Route Selection will route the call on the presubscribed IXC. The IXC field is passed to the far-end switch in the SETUP message. (The SETUP message is a call establishment message of the originating system. The SETUP message is used to specify services and features on a call-by-call basis.)

For non-ISDN-PRI calls routed by an Inter-Exchange Carrier, the IXC code that is passed to the Station Message Detail Recording (SMDR) device for its record may not be the actual code used for routing the call, unless it is an equal access IXC. For other carriers for which there is no IXC code of the proper form, the customer selects an unused IXC code for the SMDR record and puts it in the IXC field of the routing preference. It is the customer's responsibility to check that the IXCs match.

E. Electronic Tandem Network (ETN) Services

The ISDN-PRI trunks can be used to interconnect DEFINITY 2.1 and DEFINITY Generic 1 to provide ETN services. (An ETN is a network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations.)

The ETN Traveling Class Mark (TCM) will be passed between the tandem nodes by the TCM information element of the SETUP message on the ISDN-PRI facilities. (TCMs represent the Facility Restriction Levels and are used by the distant tandem switch to determine the best available facility consistent with the user's calling privileges.) The Satellite Hop Control (Conditional Routing) Count and End-to-End Connectivity message, which is used in System 85 (G2.1), will be tandemed in the SETUP message without being analyzed. For non-ISDN-PRI tandem trunks, the TCM is outpulsed after the destination address.

Interactions

The following features interact with the Generalized Route Selection feature.

- Automatic Route Selection (ARS) and Automatic Alternate Routing (AAR)

  In ARS/AAR, routing is based on the dialed number, the Facility Restriction Level of the call originator, the partitioning group number, and the time-of-day. In GRS, routing of calls is additionally based on the BCC to distinguish voice from data calls. For all trunks, ISDN-PRI as well as non-ISDN-PRI, the BCC is checked to see if the route selected is compatible.

- AAR/ARS Partitioning

  It is possible to perform GRS administration for each partition separately by using different routing patterns.

- Interworking

  GRS will support interworking; that is, the routing patterns may contain a combination of ISDN-PRI and non-ISDN-PRI trunking facilities. For non-ISDN-PRI trunking facilities, the BCC is determined by (default) administration. For ISDN-PRI trunking facilities, the BCC is determined by the information received on the signaling channel (D-channel) of the trunk.

- Call by Call Service Selection

  For each preference in a routing pattern, the customer may optionally administer an IXC code and a Network Specific Facility parameter to be used when an outgoing call is made using an ISDN-PRI facility. Call by Call (CBC) Service Selection allows the dynamic identification of a specific service type request on a per call basis. For CBC Service Selection feature, the trunk group is administered as CBC. This allows the customer to pool several types of services together and assigns the service type to them on a call basis.

- Distributed Communications System (DCS)

  The DCS can make use of the ISDN-PRI trunks as trunking facilities. However, a separate signaling link is still required for passing the call and feature information to provide feature transparency.

Administration

The following additional items are administered by the System Manager for GRS.

- Routing pattern BCCs—For each trunk group in the Routing Pattern, there will be an indication of what BCCs can use that preference. The values are 0, 1, 2, 3, and 4. More than one BCC may be supported by one trunk group preference. The BCCs assigned to a trunk group in a Routing Pattern may or may not be the same value assigned to the same trunk group in another Routing Pattern.

- Trunk Group BCCs—Each non-ISDN-PRI, data, or AVD trunk group is administered a BCC to identify the type of traffic on the trunks in that group.

For endpoints (data modules, voice terminals, etc.) a read-only BCC field appears on the screen form for that endpoint. This field reflects the endpoint's current BCC which is determined automatically by switch software.

Hardware and Software Requirements

No additional hardware is required.

Optional AAR, ARS, and ISDN-PRI services software is required.

Go to Cover

Description

Allows users, when making a call to another internal extension, to send the call directly to coverage.

Go to Cover is activated by pressing a Go to Cover button. This button can be used any time during a call attempt. If activated prior to ringing, the call does not attempt to direct to the called extension, but goes directly to coverage. Go to Cover can also be used later in the call.

Details of how Go to Cover is used in conjunction with Call Coverage are given in the Call Coverage feature description elsewhere in this chapter.

Consideration

Go to Cover gives the calling party the option to send calls directly to coverage.

Interactions

The following features do not redirect to coverage unless the caller presses the Go to Cover button:

- Intercom—Automatic
- Intercom—Dial
- Priority Calling.

Go to Cover can only be used if the called party is assigned a call coverage path; that is, the called party must have alternate answering positions assigned.

Administration

Go to Cover is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Go to Cover button.

Hardware and Software Requirements

No additional hardware or software is required.

Hold

Description

Allows voice terminal users to disconnect from a call temporarily, use the voice terminal for other call purposes, and then return to the original call.

Multi-appearance Voice Terminal Hold

Multi-appearance voice terminals have a Hold button for activating the Hold feature. Multi-appearance voice terminal users can hold a call on each call appearance. To hold a call, a user, while active on a call, simply presses the Hold button and the call is held at the call appearance being used for the call.

Single-line Voice Terminal Hold (V1)

A single-line voice terminal user can place a call on hold by pressing the Recall button or by flashing the switchhook. The user can then place another call or activate a feature, and then return to the original call by pressing Recall or flashing the switchhook again.

Single-line Voice Terminal Hold (V2, V3, or G1)

Two types of Hold (soft hold and hard hold) are provided for single-line voice terminal users. With soft hold, the user can hold the current call, consult with another party or activate/deactivate a feature, and return to the soft held call. This type of hold is used to conference or transfer a call that **includes the held call.** Hard hold can be used to hold the current call and then perform operations that **do not include the held call.** These operations could include calling another party, answering a waiting call and transferring or conferencing the waiting call with another party, activating or deactivating features, etc.

To activate soft hold, a user, while active on a call, presses the Recall button or flashes the switchhook. The user can then conference or transfer the call that is on hold. If the user dials another party and presses the Recall button or flashes the switchhook a second time, the held call is conferenced with the user and the other party. The system ignores any subsequent presses of the Recall button or flashing of the switchhook. The user can transfer the call by hanging up after conferencing the call.

To activate hard hold, a user, while active on a call, presses the Recall button or flashes the switchhook. The user then dials the "Answer Hold-Unhold" access code. This puts the call on hard hold. The user can then perform any operation that does not involve the held call. When the user wants to return to the hard held call, the user should go on-hook. The held call will ring the voice terminal and can then be answered.

If a user has a call waiting and activates hard hold, the current call is placed on hard hold and the waiting call is answered automatically.

Considerations

With the Hold feature, voice terminal users can temporarily disconnect from one call and handle another call. For example, a busy voice terminal user who receives another call can place the first call on hold and answer the second call. This results in fewer missed calls. The Hold feature can also be used when a user receives a call and needs to make another call to obtain information for the calling party.

A call involving an attendant cannot be held.

One party on hold can hear music if the Music-on-Hold Access feature is provided. The music is removed when the voice terminal user reenters the call.

Interactions

The following features interact with the Hold feature.

- Automatic Callback

  A single-line voice terminal user cannot receive an Automatic Callback call while it has a call on hold.

- Bridged Call Appearance

  Any user, active on a bridged call, can place the call on hold. If a call on a bridged call appearance is placed on hold and no other users with a bridged call appearance of the same extension number are connected to the call, the status lamp at the Bridged Appearance button indicates that the call is on hold. If the primary extension or another bridged appearance is connected to the call, the status lamp at all bridged appearances indicates an active status for the call.

- Leave Word Calling

  A held multi-appearance voice terminal user can activate Leave Word Calling toward the holding user.

  A single-line voice terminal user cannot activate Leave Word Calling toward another user while a call is on soft hold.

- Personal Central Office Line (PCOL)

  When a user, active on a PCOL call, puts the call on hold, the status lamp associated with the PCOL button does not track the busy/idle status of the PCOL.

Administration

The Hold feature is administered by the System Manager. The only administration required is to assign the Answer Hold-Unhold feature access code.

Hardware and Software Requirements

No additional hardware or software is required.

Hot Line Service

Description

Allows single-line voice terminal users, by simply lifting the handset, to automatically place a call to a preassigned extension number, public or private network telephone number, or feature access code.

The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the Hot Line Service user lifts the handset, the system automatically routes the call to the stored number and the call completes as though it had been manually dialed. If the appropriate feature access code is prefixed to the stored number, Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), Data Privacy, or Priority Calling can be used on the call. Also, if the Public or Private Network Access code is the stored number, the voice terminal user will be connected to an outgoing trunk and can dial the outside number.

A Hot Line Service voice terminal receives calls allowed by its Class of Restriction. Call reception is not affected by Hot Line Service. Likewise, the Hot Line Service destination is not affected by Hot Line Service.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension number, or any individual extension number within any of the groups can be a Hot Line Service destination. Also, any extension number within a DDC group, UDC group, or TEG can have the Hot Line Service feature assigned.

Considerations

The Hot Line Service feature is useful in any application where very fast service is required. Also, if a voice terminal is used only for accessing a certain facility, it can be assigned to Hot Line Service. The Hot Line Service voice terminal user simply lifts the handset and is connected to that facility.

The number of voice terminals that can be assigned Hot Line Service is not limited, and the number of voice terminals that can be assigned the same destination is not limited. The limit, if any, would be on the number of entries that can be stored in the Abbreviated Dialing lists.

A Hot Line Service user cannot activate any feature unless the access code is, or is part of, the destination number.

Interactions

The following features interact with the Hot Line Service feature:

- Bridged Call Appearance—Single-Line Voice Terminal (G1)

  If a single-line voice terminal is administered for Hot Line Service, bridged appearances of that voice terminal's extension will also place a hot line call automatically when a user goes off-hook on that bridged appearance.

- Loudspeaker Paging Access

  Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

- Ringback Queuing

  If a Hot Line Service call accesses a trunk group with Ringback Queuing assigned, the call can queue unless the voice terminal is termination restricted by its Class of Restriction. Queuing, when applicable, is automatic on single-line voice terminals; dialing is not required.

Administration

Hot Line Service is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Abbreviated Dialing Lists
- Hot Line Destination Number.

Hardware and Software Requirements

No additional hardware or software is required.

Hunting

Description

Checks for the active or idle status of extension numbers in one or more ordered groups. If all members of a group are active, the call can route to another group through Call Coverage or can wait in a queue for an available group member, if a queue is provided.

Hunting is accomplished through the Automatic Call Distribution (V3 or G1), Call Coverage, Direct Department Calling, and Uniform Call Distribution features. The order of hunting is defined under each feature.

Considerations

Hunting is useful whenever a group of voice terminal users receives a high volume of calls. It minimizes call completion time and attendant assistance is not required.

Interactions

With V2, V3, or G1, individual attendant extensions can be in hunt groups. However, attendant return call features will not work for these types of calls.

Administration

Hunting is administered through the Automatic Call Distribution (V3 or G1), Call Coverage, Direct Department Calling, and Uniform Call Distribution features. Administration of each of these features is described under that feature description elsewhere in this chapter.

Hardware and Software Requirements

No additional hardware or software is required for Call Coverage, Direct Department Calling, and Uniform Call Distribution. Automatic Call Distribution requires Automatic Call Distribution software.

Individual Attendant Access (V2, V3, or G1)

Description

Allows users to access a specific attendant console. Each attendant console can be assigned an individual extension number.

A user can access an individual attendant by simply lifting the handset and dialing the extension number assigned to the desired attendant. An individual attendant extension number can also be assigned to users' abbreviated dialing button for fast access to the specific attendant.

Individual attendants can be accessed by voice terminal users, incoming trunks, remote access, and other attendants. A specific attendant, when called, can extend the call to another trunk or extension.

Each attendant has a queue that allows two incoming calls to wait. This individual attendant queue has priority over all other attendant seeking calls.

Whenever a call is in an individual attendant's queue, the top lamp of the Forced Release button (basic console) or the Personal lamp (enhanced console) lights to indicate this condition. Call Waiting tones are provided only on calls to the attendant group and are not provided for waiting individual attendant calls.

An individual attendant can be a part of a hunt group. The hunt group can be a Direct Department Calling (DDC) group or a Uniform Call Distribution (UCD) group. Calls to individual attendants and calls to the attendant group have priority over hunt group calls to an individual attendant.

Any call made from an attendant console which is assigned an individual extension is considered to be made from the individual attendant, not the attendant group.

Considerations

With Individual Attendant Access, attendant consoles can become more flexible by assigning each one an individual extension number. An individual attendant extension allows an attendant to use features that an attendant group cannot use; for example, individual attendant extensions can be a member of a DDC or UCD group. An individual attendant can also be accessed when the Centralized Attendant Service feature is in effect. Another advantage is that each attendant extension can have its own Class of Restriction (COR) and Class of Service (COS).

The Position Available lamp on the attendant console only indicates whether or not attendant group calls can be accepted. It does not indicate whether or not individual attendant calls can be accepted.

Each attendant console has one position busy button. When the lamp associated with this button is lighted, the attendant will not receive attendant group calls but can still receive individual attendant calls.

Since hunt groups have better queuing and make-busy features than individual extensions, it may be desirable to assign an individual attendant as the only member of a hunt group. This way the individual attendant can receive calls as a hunt group member for more efficient handling of calls.

Interactions

The following features interact with the Individual Attendant Access feature.

- Abbreviated Dialing

  Individual attendant extensions can be in Abbreviated Dialing lists. Individual attendants, however, cannot have their own Abbreviated Dialing lists.

- Attendant Display

  For calls to or from individual attendants, individual attendant names (when specified) will be displayed instead of the individual attendant extensions.

- Bridged Extension

  Individual attendant extensions cannot be assigned to a bridged call appearance.

- Busy Verification of Terminals and Trunks

  An individual attendant extension cannot be busy verified.

- Call Coverage

  Individual attendant extensions can be points in a coverage path but cannot be a member of a coverage answer group.

- Call Park

  Individual attendants can park calls on their own extension or another individual attendant extension.

- Call Pickup

  Individual attendant extensions cannot be in Call Pickup groups.

- Centralized Attendant Service (CAS)

  Individual attendants can be accessed when CAS is in effect.

- Class of Restriction (COR) and Class of Service (COS)

  Each individual attendant extension has its own COR and COS. However, it is recommended that an individual attendant and the group with which he or she is associated be assigned the same COR.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  Individual attendant extensions can be assigned to DDC and UCD groups. Unlike voice terminal users, individual attendants can answer DDC and UCD calls as long as there is an idle call appearance and no other DDC or UCD call is on the console.

- Facility Busy Indication

  An individual attendant extension can be stored in a Facility Busy Indication button.

- Integrated Directory

  The names and extensions of the individual attendants are stored in the directory associated with this feature.

- Leave Word Calling

  A message from an attendant will indicate whether it is from the attendant group or whether it is from an attendant which has an individual extension.

- Night Service—Night Console Service

  Activation and deactivation of this feature affects only calls to the attendant group. Calls to individual attendant extensions are allowed when night service is active. A night-only attendant console with an individual extension can receive individual attendant calls when night service is not active.

- Privacy—Attendant Lockout

  This feature applies only to attendant group calls. Individual attendant calls are not affected.

- Voice Terminal Display

  For calls from individual attendants, individual attendant names (when specified) will be displayed instead of the individual attendant extensions.

Administration

Individual Attendant Access is administered on a per-attendant console basis by the System Manager. The following items require administration for each attendant console:

- Extension number
- Name
- Class of Restriction
- Class of Service.

Hardware and Software Requirements

No additional hardware or software is required.

Information System Network (ISN) Interface (V2, V3, or G1)

Description

The AT&T ISN is a packet switched local area network that links mainframe computers, minicomputers, word processors, storage devices, personal computers, printers, terminals, and communications processors into a single system. The interface to System 75 is via an Asynchronous Data Unit (ADU). A Modular Processor Data Module (MPDM) may be used but the ADU is more economical. Also, future versions of the ISN will have integrated ADUs.

The ISN is a packet-switched local area network. This local area network is made up of one or more modular data-only digital communications switches.

The interface between the system and the ISN is via a Data Line Port in conjunction with an ADU. A (Modular) Processor Data Module [(M)PDM] may be used instead of the ADU, but the ADU is more economical. Also, future versions of the ISN will have integrated ADUs. The (M)PDM or ADU connects to an Asynchronous Interface Module (AIM) on the Packet Controller or Terminal Concentrator (see Figure 3-21). This interface allows the system and the ISN to share data capabilities.

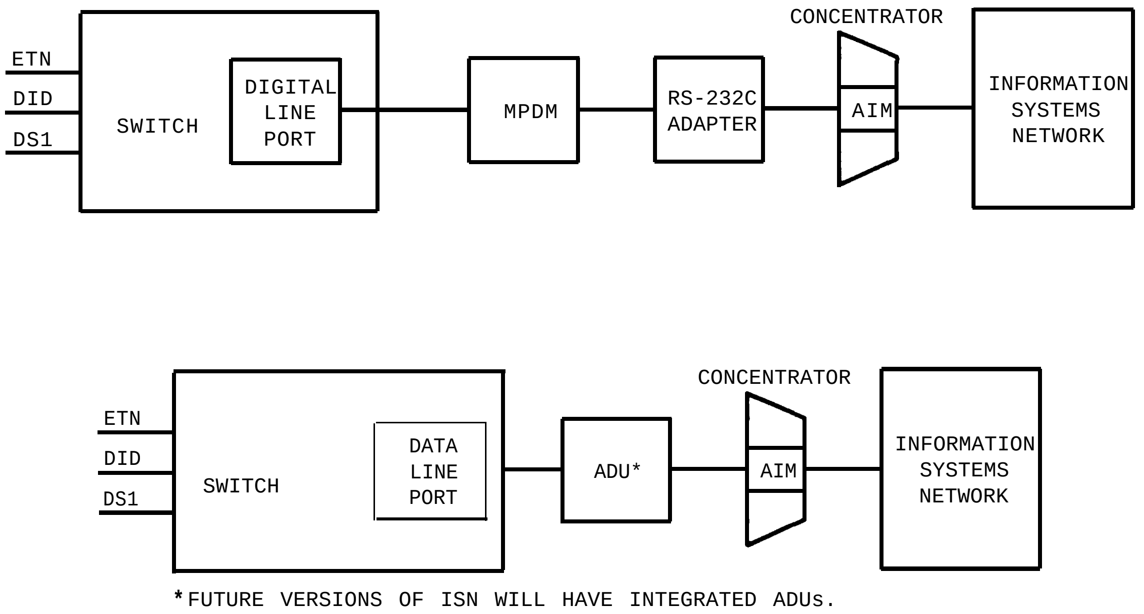


Figure 3-21. System to ISN Connectivity

Data is transferred between the system and the ISN on one-way trunks (either incoming or outgoing). Each data line port is administered for a specific data rate, which can be any of the common asynchronous data rates ranging from LOW to 19,200 bps.

Considerations

Connectivity between ISN and the system provides the following major benefits:

- Users on ISN may (in addition to having access to other endpoints directly connected to ISN) have access to any endpoint connected to the system or addressable from the system.

- Users who either connect to or have access to the system may also access endpoints connected to ISN.

Since the ISN switches are modular, the local area data communications network can be designed so that it is both versatile and cost-effective. A single packet controller can be configured to support from 40 to 1920 data ports.

Interactions

The following features interact with the Information System Network (ISN) Interface feature.

- Abbreviated Dialing

  Outgoing lines cannot use Abbreviated Dialing.

- Automatic Circuit Assurance

  Automatic Circuit Assurance is not provided for data line port links to or from the ISN.

- Data Call Setup

  Data Terminal (Keyboard) Dialing is used to access ISN endpoints. A data call to an ISN data endpoint from a system digital data endpoint requires 2-stage dialing. A user must first dial the extension assigned to the outgoing ISN group, and then interact with ISN and enter the second address (data endpoint).

- Data Hotline

  Outgoing lines cannot use hot line calling.

- Modem Pooling

  If an analog data endpoint is used in an ISN connection, and a conversion resource is needed, the system will obtain a conversion resource from the appropriate pool.

- System Measurements

  No traffic measurements are made on data line port links to or from the ISN.

- Uniform Call Distribution

  Outgoing lines should be members of a Uniform Call Distribution (UCD) group. This way, the system automatically selects an idle port when a user tries to access the ISN.

Administration

Data module extensions used to access ISN must be administered by the System Manager as data lines connected to the ISN. The System Manager can then administer the other options required for each data line. These options include:

- Keyboard Dialing—If the line is incoming (to the system), Keyboard Dialing should be enabled so that the system can be accessed by the ISN. If the line is outgoing (to the ISN), Keyboard Dialing should be disabled.

- Configuration—This option should be disabled on both incoming and outgoing lines to prevent the ISN from changing the data line configuration.

- Busy Out—This option should be enabled for outgoing lines so that a member of the outgoing ISN group can be "busied out" and let calls go through another member of the group.

- Speeds—Data speeds should be selected according to individual needs, and should be the same as those at connecting ISN ports. Only one speed should be assigned to each data line port.

- Autoadjust—This option is not needed with the ISN, and should be disabled on incoming lines. This option can only be set if Keyboard Dialing is enabled.

- Permit Mismatch—This option should be disabled on both incoming and outgoing data lines.

- Disconnect—The disconnect sequence should be administered according to the characteristics of the device. This option can only be set if Keyboard Dialing is enabled.

- Parity—This option should be administered as even. This option can only be set if Keyboard Dialing is enabled.

- Dial Echoing—This option should be disabled so that characters are not echoed back to the ISN. This option can only be set if Keyboard Dialing is enabled.

- Answer Text—This option should be disabled, and can only be set if Keyboard Dialing is enabled.

- Connected Indication—This option should be disabled, and can only be set if Keyboard Dialing is enabled.
- Class of Restriction (COR)—Outgoing lines should be origination restricted. Incoming lines should be termination restricted.

Hardware and Software Requirements

One TN726 Data Line circuit pack is required for each eight ISN interfaces. No additional software is required.

Integrated Directory

Description

Allows internal system users with display-equipped terminals to access the system database, use the touch-tone buttons to key in a name, and retrieve an extension number from the system directory. The directory contains an alphanumeric listing of the names and extension numbers assigned to all voice terminals administered in the system.

The Integrated Directory feature can be accessed by display-equipped voice terminal users or attendants with an assigned Integrated Directory button.

The names in the directory will be those administered by the System Manager on the individual voice terminal forms. Names cannot exceed 15 characters (including spaces and commas) and can be entered in one of the following three formats.

- Last name, comma, first name, space, then middle name or initial, if desired. For example, the following entries are acceptable:

  Jones,Betty Ann
  Smith,A E
  Thomas,John J
  Abbott,Lynn

- First name, space, second name or initial, and then last name. For example, the following entries are acceptable:

  Betty Ann Jones
  A E Smith
  John J Thomas
  Lynn Abbott

- A single entry is also acceptable:

  Cafeteria
  1J409
  2F816
  Purchasing

The following is an example of a typical Integrated Directory database:

1J409
Abbott,Lynn A
Brown,Kent J
Cafeteria
Carr,Danny
Carter, Ann
2F816
.
.
.

Purchasing
Barbara Quincey
Roberson,Don T
William Ruoff
Smith,A E
Streck,R T

The touch-tone buttons are used to key in the numbers and letters labeled on them. The following exceptions apply:

- Button 7 (PRS) is also used for a Q.
- Button 9 (WXY) is also used for a Z
- Button * is used for a space or comma.
- Button # is not used.

To activate the Integrated Directory feature, the user presses the Integrated Directory button. This puts the voice terminal in the Integrated Directory mode and turns off the tones normally generated when a touch-tone button is pressed. The touch-tone buttons are now used exclusively for keying in names and not for dialing.

After the Integrated Directory button is pressed, the alphanumeric display will show DIRECTORY—PLEASE ENTER NAME. Names are always keyed in the following order: last name, comma, and then first name or initial. When searching for a single entry, the letters or numbers would be keyed in order. Several letters might be needed to get the correct entry.

When a button is pressed, the display will show the first name that matches the first letter on the button. For example, if a user is searching for the name Ann Carter and presses button 2 to key in the letter C, the display might show Abbott,Lynn A and an extension number. (Button 2 matches A before it matches C.) If the user presses button 2 again to key in the letter A, the display will stay the same. (Again, AB is matched before CA.) If the user now presses button 7 to key in an R, the display might show Carr,Danny and an extension number.

At this point, the user can press button 8 to key in the letter T or can press the Next Message button on the alphanumeric display. Pressing Next Message displays the next name in the directory and, in this case, might be Ann Carter.

When the desired name and extension number are displayed, the user can automatically place a call to that person by pressing the Return Call button.

If a name is entered but not found in the directory, the display will show NO MATCH—TRY AGAIN. To search for another name, the user presses the Integrated Directory button again, and the feature is reactivated.

To exit the Integrated Directory mode, the user presses one of the other mode buttons assigned to the alphanumeric display module, for example, the Normal mode button.

Considerations

With Integrated Directory, users spend less time looking up names and extension numbers. Instead of searching through lists or directories, a user simply keys in the desired name and the display shows the name and extension number. Less dialing time is also required if a Return Call button is provided. When the desired extension is displayed, the user just presses the Return Call button to automatically place the call.

The maximum size of the directory is 400 entries (V1), 800 entries (V2 or V3), or 1600 entries (G1). The maximum length of the name is 15 characters (including spaces and commas). The extension number cannot exceed four digits (V1) or five digits (V2, V3, or G1).

A maximum of ten users can activate Integrated Directory at the same time. If more than ten users try to activate the feature at the same time, the Integrated Directory button lights and the display shows "Directory unavailable — Try Later."

The entire directory cannot be searched by pressing button 2. Pressing button 2 and then continually pressing Next Message will display, one by one, all entries beginning with A, B, C, and 2. If all entries have been displayed and Next Message is pressed again, the display will repeat from the first entry in the listing associated with button 2.

When the voice terminal is in the Integrated Directory mode, it cannot be used to make calls or to access features by dial code. It can, however, still be used to activate other features of to place calls if dialing is not required. Also, a user can enter the Integrated Directory mode while active on a call, and calls can be received when the Integrated Directory mode is active.

Interactions

The following features interact with the Integrated Directory feature.

- Attendant Display and Voice Terminal Display

  If prefixed extensions are used in the system's dial plan, the prefix is not displayed when the extension is displayed. With V3, the Return Call button cannot be used to dial prefixed extensions, because this button causes the system to dial the displayed number, which does not contain the entire extension. With G1, the Return Call button can be used to dial prefixed extensions, because the G1 system will dial the prefix, even though it is not displayed.

- Touch-Tone Dialing

  Call origination and feature access by dial code is not allowed when the Integrated Directory feature is active.

Administration

Integrated Directory is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Display Module
- Integrated Directory Button
- Return Call Button
- Messaging Cartridge (for 7404D)
- Next button.

Hardware and Software Requirements

No additional hardware or software is required.

Integrated Services Digital Network—Primary Rate Interface (G1)

Description

Allows connection of the system to an Integrated Services Digital Network (ISDN) by using a standard ISDN frame format called the Primary Rate Interface (PRI). The ISDN gives the system users access to a variety of public and private network services and facilities.

The ISDN-PRI is a 1.544 Mbps digital interface that consists of a 1.536 Mbps signal multiplexed with an 8 kbps framing channel. The 1.536 Mbps signal is divided into 24 channels of 64 kbps each (23 "B" voice or data channels and 1 "D" signaling channel). The D channel multiplexes signaling messages for the 23 B channels. The ISDN-PRI is consistent with the International Consultative Committee for Telegraph and Telephone (CCITT) Recommendation Q.931 and Q.921 for ISDN signaling.

ISDN-PRI signaling in the system is supported by the TN767 DS1 Trunk circuit pack coupled with the TN765 Processor Interface circuit pack. The D channel (signaling channel) is switched through the TN765 circuit pack.

With the ISDN-PRI, the system can interface with a wide range of other products including network switches, PBXs, and host computers. These products include the following:

- 4 ESS Switch
- DEFINITY Communications System Generic 2.1
- DEFINITY Communications System Generic 1
- Any other products that adhere to the ISDN-PRI signaling protocol.

As an example of how the ISDN-PRI is used in private and public network configurations, see Figures 3-22 and 3-23, respectively. As seen in these figures, the ISDN-PRI can be used to interface a Private Branch Exchange (PBX) to a Public Switched Network, a PBX to a Host Computer, or a PBX to another PBX.

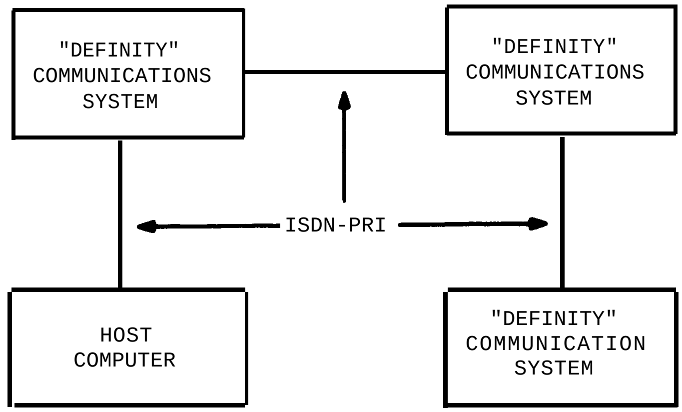


Figure 3-22. ISDN-PRI Private Network Configuration

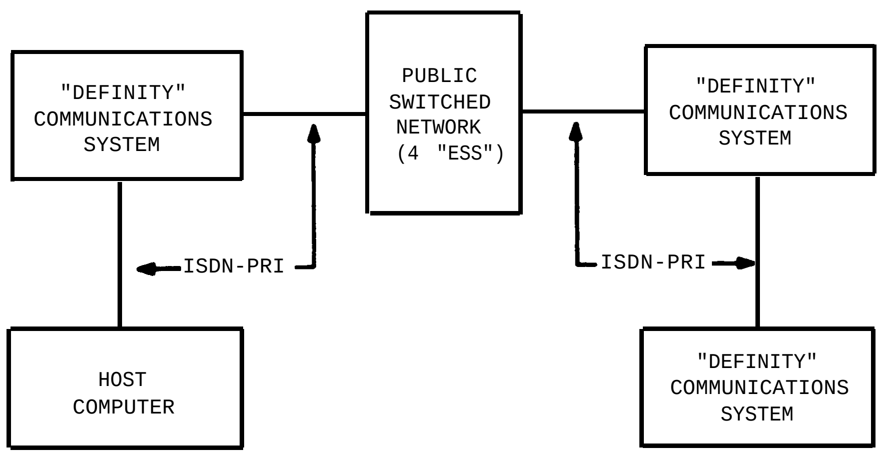


Figure 3-23. ISDN-PRI Public Network Configuration

ISDN-PRI Services

The ISDN-PRI provides system users with following services:

- Call by Call Service Selection
- Call Identification Display
  - — Station Identification Number (SID)
  - — Automatic Number Identification (ANI)
  - — Calling and Connected Number Display
  - — Calling and Connected Party Name Display
- SID/ANI to Host Call Identification (G1)
- Private Network Services.

Call by Call Service Selection

Call by Call Service Selection allows the same ISDN-PRI trunk group to carry calls to a variety of services or facilities (such as a Software Defined Network [SDN], MEGACOM telecommunications service, MEGACOM 800 service, etc.) and/or carry calls using different inter-exchange carriers. This feature is described in detail under the Call by Call Service Selection feature description elsewhere in this chapter.

ISDN-PRI Call Identification Display

ISDN-PRI Call Identification Display provides a transparent name/number display for all display-equipped voice terminals within an ISDN-PRI network. The feature is transparent in that the same information is provided at all ISDN-PRI facilities. Voice terminals using this feature should be digital voice terminals with a 40-character alphanumeric display.

ISDN-PRI Display Information is provided in addition to the normal Voice Terminal Display and Attendant Display features, when the network supports end-to-end ISDN-PRI connectivity. When both ISDN-PRI and DCS display information, or DCS display information only, are received, the switch will display the DCS display information in the DCS format. If ISDN display information is received, and no DCS display information is received, then the ISDN display information is displayed in the ISDN formats.

Two types of identification numbers are provided with the ISDN-PRI. These identification numbers may be used in the various types of displays used with the ISDN-PRI. The two types of identification numbers are as follows:

- Station Identification Number (SID)

  A 10-digit Direct Distance Dialing (DDD) number associated with a specific station. When a system user makes a call that uses the ISDN, that user's SID is provided by the system for the ISDN.

- Automatic Number Identification (ANI)

  The calling party's billing number that is provided to an inter-exchange network via Equal Access or Centralized Automatic Message Accounting (CAMA). This number is stored at either a local or network switch. If a customer is connected directly to the AT&T network, the ANI is the customer's billing number stored in that network. If the SID is not provided on an incoming ISDN-PRI call, the system uses the ANI for the station identification number.

The following types of display information are provided with the ISDN-PRI.

- Calling Party's Number

  The calling party's number is shown on the called party's display. On calls generated from the system, the calling party's number is a 10-digit DDD number. This number is provided only if the outgoing ISDN-PRI trunk group is administered to send the SID to the network. On calls incoming to a system, the network may provide either the SID or ANI as the calling party's number. Dashes are inserted in the displayed number between the area code (if shown), the office code, and the local number. Extension numbers and 12-digit international numbers are shown without dashes.

- Calling Party's Name

  The calling party's name is shown on the called party's display. On calls generated from a system, the calling party's name is provided if the ISDN-PRI trunk group is administered to send the name to the network. Other public or private networks may also provide the calling party's name. If the called party's name is not available, the called party's display will show "CALL FROM" instead, followed by the calling party's number.

- Connected Party's Number

  The called party's number is shown on the calling party's display as the calling party dials the number. When a call is made over private or public networks via ISDN-PRI facilities, the calling party receives a 10-digit identification number of the party who answers the call. The calling party's display always shows the number associated with the first party to answer the call (this may or may not be the party that was actually called). The format of the called party's number is the same as that of the calling party's number, described previously.

- Connected Party's Name

  The called party's name is shown on the calling party's display. If the trunk group is administered to send the name, the system provides the called party's name to the calling party on incoming ISDN-PRI calls. The calling party's display shows the name associated with the first party to answer the call (this may or may not be the party that was actually called).

The display fields that may be used for the ISDN-PRI are as follows:

- Name—Maximum of 15 characters
- Number—Maximum of 12 characters
- Miscellaneous Call Identification—Maximum of 8 characters
- Reason for Call Redirection—Maximum of 2 characters.

The display information will vary, depending on the type of call, how the call is handled (for example, whether it is redirected or not), and what information is available on the call. The display information for basic calls (those with just a calling and called party) and for redirected calls is given in the following paragraphs.

Basic ISDN-PRI Call

A basic ISDN-PRI call has both a calling and a called party, and the called party answers the call. When the calling party goes off-hook, "a=" is shown on the display. The digits are then displayed as they are dialed. These digits may be overwritten by the trunk group name if the Outgoing Display field of the Trunk Group Administration form is administered as "yes." Once the call is connected, the displays for the calling and called parties are as follows.

If **both the name and number information are available,** the displays are as follows. The MISCID (Miscellaneous Identification) field may be blank if that information is not provided.

- Calling Party Display

a= CALLED NAME	CALLED NUMBER	MISCID

- Called Party Display

a= CALLING NAME	CALLING NUMBER	MISCID

If **only the name information is available,** the displays are as follows:

- Calling Party Display

```
a= CALLED NAME          MISCID
```

- Called Party Display

```
a= CALLING NAME         MISCID
```

If **only the number information is available,** the displays are as follows:

- Calling Party Display

```
a= ANSWERED BY   CALLED NUMBER   MISCID
```

- Called Party Display

```
a= CALL FROM   CALLING NUMBER   MISCID
```

If **neither the name nor number information is available,** the displays are as follows:

- Calling Party Display (shows one of the following, depending on administration)

a= DIALED NUMBER	MISCID

a= TRUNK NAME	MISCID

- Called Party Display

a= TRUNK NAME	MISCID

Redirected ISDN-PRI Call

Redirected ISDN-PRI calls are those calls which have been redirected from the called party's extension by features such as Call Coverage, Call Forwarding All Calls, Bridged Call Appearance, and Call Pickup. Once the call is connected, the displays for the calling, called, and connected parties are as follows.

- Calling Party Display

```
a= CONNECTED NAME CONNECTED NUM     MISCID
```

- Called Party Display

  The following information is displayed if the called party bridges onto the redirected call after it has been answered. In this situation, the connected party's display (given later) will show the same information. The calling party's display will also be updated if the calling and called parties are on the same switch.

```
a= CONFERENCE 2
```

- Connected Party Display

  The connected party is the party who answers the redirected call. The "R" indicates the reason for redirection.

```
a= CALLING ID to CALLED ID      R
```

SID/ANI to Host Call Identification (G1)

The SID/ANI to Host feature enables SID/ANI information to be passed from the switch to the ISDN Gateway, so that the ISDN Gateway can forward the information on to a host for data screen delivery to agents in a split.

By delivering call identification information such as SID/ANI and switch information such as the answering agent's extension to an adjunct network (ISDN Gateway), the adjunct can automatically deliver data screens to agents for new call arrivals and call transfers.

Figure 3-24 shows a simplified diagram of a SID/ANI to host arrangement. The ISDN Gateway is a 3B2 or 6386 computer connected to the switch on one side and to a host computer on the other side. The connection to the switch is over a synchronous interface with BX.25 protocol.

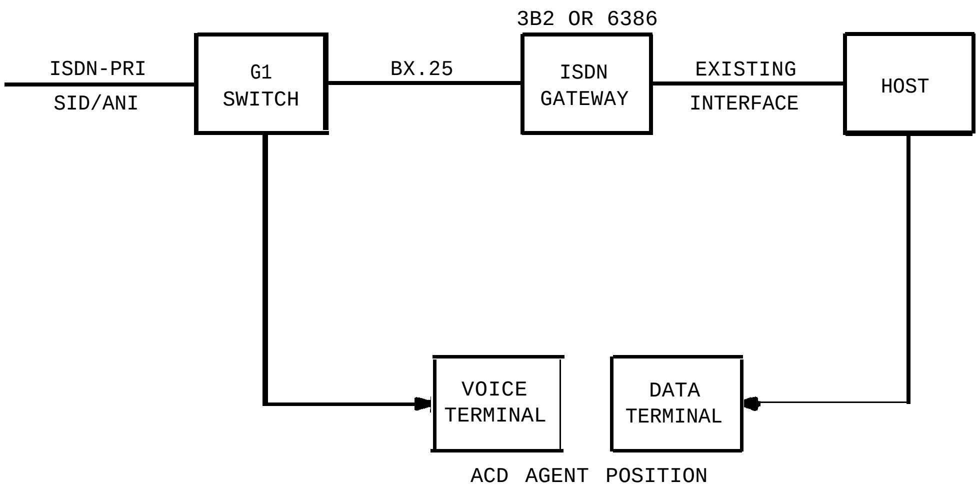


Figure 3-24. SID/ANI to Host Configuration (G1)

The ISDN Gateway is administered as an Outgoing Call Management (OCM) link.

The maximum number of OCM-SID/ANI links that can be supported in G1 is one. OCM and SID/ANI applications cannot co-reside on the same adjunct software.

SID/ANI information can be used by the host computer to identify the caller, so that the caller information can be retrieved from a database. Two possible applications for SID/ANI information are described below:

- Data Screen Delivery

  In this application, the switch makes the routing decisions. As an incoming ISDN-PRI call comes into an ACD split, the switch passes the calling party's SID/ANI information to the ISDN Gateway. The ISDN Gateway forwards this information on to the host computer which performs a database search on the caller information (SID/ANI) and retrieves caller data to fill a selected data screen based on the service dialed. When an available agent is alerted on the switch for the call, the switch passes the agent identification to the ISDN Gateway so the data screen can be sent to the agent's data terminal.

- Voice Response Integration

  In this application a Voice Response System, connected to the switch, makes the routing decisions.

  Incoming calls are connected to the Voice Response System. The Voice Response System plays announcements and collects information from the caller before deciding where to route the call.

  When ready to route the call, the Voice Response System transfers the call to an ACD split and the ISDN Gateway transfers the selected data screen (based on SID/ANI information) to the ACD agent that answers the call.

Private Network Services

In addition to providing access to switched public networks, the ISDN-PRI can provide private network services by connecting DEFINITY Generic 1 and DEFINITY Generic 2.1 systems in an Electronic Tandem Network (ETN) or Distributed Communications System (DCS) configuration. This gives customers more efficient private networks that support new integrated voice and data services. ETN and DCS services are provided as follows:

- ETN Services

  DEFINITY Communications Systems that function as tandem nodes in an ETN can be interconnected using DS1 trunking facilities and an ISDN-PRI. All signaling between the tandem switches is done with the ISDN-PRI D channel and normal ISDN procedures. The ISDN-PRI can also be used to connect ETN tandem and main switches. In this case, the main switch collects all of the address digits from local users as well as users at other satellite and tributary switches, and sends a SETUP message over the ISDN-PRI to the tandem switch.

  Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) are used with the ISDN-PRI and DS1 trunking facilities to access ETN facilities. The AAR and ARS features are used to collect the dialing information for the SETUP message that is sent from the main switch.

- DCS Services

  ISDN-PRI facilities can be used in a DCS arrangement whenever tie trunks are used to connect the DCS nodes.

  Most DCS features are not affected by the ISDN-PRI. However, the ISDN-PRI does have a minor impact on a few of the DCS features, as far as the functions that the local and remote switches must perform. Even though a DCS feature may be slightly affected in this manner, the use of the feature is still the same. If there is a conflict between a DCS message and an ISDN-PRI message on a call (for example, the calling extension number in the DCS message and the calling party's number in the ISDN-PRI message), the DCS message is used.

ISDN-PRI Interworking

ISDN-PRI Interworking is the combination of both ISDN-PRI trunking facilities and non-ISDN-PRI trunking facilities on a call. A non-ISDN-PRI trunking facility is any trunk facility supported by the system that does not use the CCITT recommended Q.931 message set for signaling. Non-ISDN-PRI trunking facilities include facilities such as Analog trunks, Alternate Voice/Data (AVD) DS1 trunks, and DS1 trunks with bit-oriented signaling or robbed-bit signaling.

The system supports the conversion of ISDN-PRI signaling to non-ISDN-PRI in-band signaling and the conversion of non-ISDN-PRI in-band signaling to ISDN-PRI signaling for Interworking purposes.

A mixture of ISDN-PRI and non-ISDN-PRI signaling is required to provide end-to-end signaling when different types of trunk facilities are used on a call. See Figure 3-25 for an example of Interworking. In this example, a call for someone at Switch B comes into Switch A. Interworking allows the ISDN-PRI signaling of the call to be converted at Switch A to non-ISDN-PRI in-band signaling before the call forwards to Switch B. Even though the call comes into Switch A on an ISDN-PRI trunk, Switch A can send the call to Switch B over a non-ISDN-PRI trunk by converting the signaling information.

The system provides accurate Station Message Detail Recording (SMDR) billing information on calls that are not Interworked. Accuracy of SMDR billing information on Interworked calls is equivalent to the accuracy provided by the public network.

The system does not support the conversion of DCS feature transparency messaging into ISDN-PRI messaging. Therefore, DCS-provided feature transparency will be lost when the call leaves the DCS network. The basic call, however, will still go through.

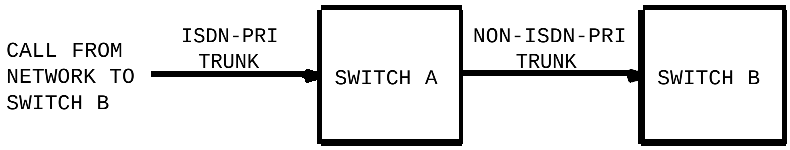


Figure 3-25. Interworking Example

Considerations

With the ISDN-PRI, system users have access to a variety of services that are only available through the ISDN.

ISDN-PRI Call Identification Display is provided on a call only if that call is routed through all ISDN-PRI facilities. Non-ISDN-PRI facilities will not carry the necessary display information. If the called party is at a non-ISDN-PRI facility, the system will display either the dialed digits or the trunk group name (depending on administration) on the calling party's display.

ISDN-PRI facilities support equal access to all inter-exchange carriers.

Interactions

The following features interact with the Integrated Services Digital Network—Primary Rate interface feature.

- Attendant Display

  The information provided by ISDN-PRI Call Identification Display is in addition to the display features already provided.

  When an ISDN-PRI call is redirected to the attendant, and both name and number display information are available, the name will be displayed on the console for the calling and called party identification.

- Bridged Call Appearance

  ISDN-PRI Call Identification Display information is provided at both the primary extension number and the extension number with the bridged call appearance. Both displays show the same called party information, whether the call is made from the primary extension number or the bridged call appearance. On a call to a primary extension number, the calling party's display shows the identification of the called primary extension number, even if the call is answered by the bridged call appearance.

- Call Forwarding All Calls

  When an ISDN-PRI call is forwarded, no ISDN-PRI Call Identification Display information is shown on the display of the forwarding extension.

  The forwarded-to extension's display shows information on the calling party, called party (if the forwarded-to station is on the same switch), and the reason for redirection.

- Call Pickup

  When an ISDN-PRI call is answered via Call Pickup, the calling party's display identifies the answering party, the called party's display identifies the calling party, and the answering party's display identifies both the calling and called parties.

- Conference—Attendant

  A conference call is identified as a conference call with "n" number of conferees. This display information is generated locally and will not change the display of a user on another switch.

- Conference—Terminal

  A conference call is identified as a conference call with "n" number of conferees. This display information is generated locally and will not change the display of a user on another switch.

- Distributed Communications System

  If both Distributed Communications System (DCS) and ISDN-PRI features are provided over the same facility with a DEFINITY Generic 1 system, the ISDN-PRI display information is displayed in DCS format.

- Facility Restriction Levels and Traveling Class Marks

  The Traveling Class Mark (TCM) used to pass on the originating facility's Facility Restriction Level (FRL) is sent by ISDN facilities in the SETUP message.

- Hold

  When an ISDN-PRI call is placed on hold, the display of the party who activates hold goes blank and then identifies the newly connected party if there is one. The held party's display remains unchanged. When the held party is reconnected to the holding party, the holding party's display is updated to indicate the current status of the call.

- Hunting

  On ISDN-PRI calls to a hunt group extension, the calling party's display will identify either the group or the group member who answers the call, depending on administration.

- Terminating Extension Group

  On ISDN-PRI calls to a Terminating Extension Group (TEG), the calling party's display will identify either the group or the group member who answers the call, depending on administration.

- Transfer

  When an ISDN call is transferred, the display of the party who transfers the call goes blank. The transferred party's display does not change. The display of the transferred-to party identifies the party who transferred the call.

  When an ISDN call is transferred to a party on the same switch as the party who transfers the call, the information on the display of the party who transfers the call is shown on the transferred-to party's display.

Administration

ISDN-PRI is administered on a per-system basis by the System Manager. The following items require administration.

- DS1 Circuit Pack
- DS1 Synchronization Plan

- Processor Interface Circuit Pack
- ISDN-PRI Signaling Link (D-channel)
- ISDN-PRI Trunk Group
- GRS Routing Patterns.

The following administration items are required for ISDN-PRI Call Identification Display:

- A Direct Distance Dialing SID Prefix Table that includes the following items:
  - Extension Length—From 1 through 5
  - Extension Code—Defines a set of extensions with the same leading digits as the extension code
  - SID Prefix—Used to create a 10-digit DDD number for an ISDN-PRI Station Identification (SID) Number. The SID prefix can be from six through ten digits in length. The sum of the number of DDD prefix digits and extension digits must equal ten (except when 5-digit extension numbers or 10-digit DDD prefix numbers are used). The system will not send call identification information on calls from extensions that have an extension code with the DDD prefix left blank.
- Whether the group name or member name is displayed on the calling party's display (per Hunt Group and Terminating Extension Group).

The following administration is required for SID/ANI to Host Call Identification (G1):

- OCM-SID/ANI and Automatic Call Distribution (ACD) must be administered as "yes" on the System Parameters—Customer Options form.
- Trunk groups can be assigned the type of calling party information to be passed using SID/ANI to Host on a call-by-call basis. The choices for each trunk group are: SID only, ANI only, prefer SID but accept ANI, prefer ANI but accept SID, and blank.

Hardware and Software Requirements

A TN767 DS1 circuit pack is required for assignment of a signaling link and up to 23 ISDN-PRI Trunk Group members. The DS1 provides 24 ports. A TN768 Tone Clock circuit pack is required to provide synchronization for the DS1 circuit pack. A TN765 Processor Interface circuit pack is required for use with the TN767 DS1 circuit pack.

Display-equipped voice terminals are required for the display of ISDN-PRI Call Identification Display information.

One processor interface link is required per ISDN-PRI.

If SID/ANI to Host Call Identification (G1) is desired, a computer, such as a 3B2, is required for use as the ISDN Gateway.

ISDN-PRI software is required.

Intercept Treatment

Description

Provides an intercept tone or a recorded announcement or routes the call to an attendant for assistance when calls cannot be completed or when use of a feature is denied.

- Intercept Treatment—Tone

  Provides a siren-type tone to internal calls that cannot be completed as dialed.

  Intercept Tone is provided to voice terminals when users lift the handset and do not dial within 10 seconds, pause longer than 10 seconds between digits during the dialing process, or remain connected to Loudspeaker Paging for longer than an administered interval.

  When a single-line voice terminal user receives Intercept Tone for 30 seconds and does not hang up or does not hang up within 10 seconds after other parties have disconnected, the user receives dial tone for a new call origination.

  When multi-appearance voice terminal users receive Intercept Tone for 30 seconds and do not hang up, the call appearance returns to idle. If the multi-appearance user is the last party left on a call, the call appearance immediately returns to idle.

  With Version 3 or DEFINITY Generic 1 systems, if a voice terminal extension is assigned a Class of Service (COS) with Off-hook Alert, and the user of that voice terminal receives Intercept Tone for a specified period of time and does not hang up, an emergency call is placed to the attendant.

- Intercept Treatment—Recorded Announcement

  Provides a recorded announcement to Direct Inward Dialing and incoming Parivate Network Access calls that cannot be completed as dialed. The System Manager selects and records the message.

  Toll charges do not apply to Direct Inward Dialing and Private Network Access calls routed to Recorded Announcement.

- Intercept Treatment—Attendant

  Allows attendants to provide information and assistance to callers on all Direct Inward Dialing or incoming Private Network Access calls that cannot be completed as dialed. Normal toll charges apply to these calls.

- Intercept Treatment—Station (V3 or G1)

  Allows a specific voice terminal to receive certain calls that cannot be completed because of a controlled restriction (see Controlled Restrictions feature) or because the called party has activated Do Not Disturb. The controlled restrictions which can be administered to send calls to station intercept are Outward, Termination, and Station-to-Station.

The calling party hears audible ringing while the call is being routed to the voice terminal assigned for Intercept Treatment. The calling party receives no indication that the call is receiving Intercept Treatment.

Considerations

The Intercept Tone lets a user know when a call cannot be completed as dialed. The user can then hang up or try the call again. When Direct Inward Dialing and Private Network Access calls cannot be completed as dialed, a recorded announcement can be provided or, for more personal service, the calls can be routed to an attendant or voice terminal user (V3 or G1).

Ten (V1 and V2) or 64 (V3 or G1) recorded announcements can be used with the system. None, some, or all of these announcements can be used for Intercept Treatment.

Only one person can be connected to an announcement at any given time. The caller is always connected to the beginning of the announcement.

Interactions

Attendant Intercept and Recorded Announcement Intercept (both optional) cannot be used together. Direct Inward Dialing calls cannot be assigned Intercept Treatment—Tone.

Administration

The Intercept Tone is standard and requires no administration. However, administration is required to determine whether Direct Inward Dialing and Private Network Access calls are routed to the attendant or to an announcement. With Version 3 or DEFINITY Generic 1 systems, administration is required to determine whether calls sent to intercept because of controlled restrictions are routed to intercept tone, a voice terminal, an attendant, or an announcement. If an announcement is to be used, the announcement must be administered.

Hardware and Software Requirements

Requires announcement equipment and one port on a TN742 Analog Line circuit pack for each announcement. With Version 3 or DEFINITY Generic 1 systems, a TN750 Announcement circuit pack can be used to provide 16 different announcements. The announcements can be recorded directly onto the TN750 circuit pack. No announcement equipment is required when this circuit pack is used. No additional software is required.

Intercom—Automatic

Description

Provides a talking path between two voice terminal users. Calling users press the Automatic Intercom button and lift the handset, or vice versa. The called user receives a unique intercom alerting signal, and the status lamp associated with the Dial or Automatic Intercom button, if provided, flashes.

Considerations

With Automatic Intercom, users who frequently call each other can do so by pressing one button instead of dialing an extension number.

Single-line voice terminal users can receive Automatic Intercom calls, but cannot originate them.

A combination of Automatic and Dial Intercom can be used between terminals so that Automatic Intercom applies in one direction and Dial Intercom applies in the other.

Two terminals with Automatic Intercom to and from each other, or terminals with combined Automatic and Dial Intercom to each other, must be in the same Intercom group.

Interactions

The following features interact with the Intercom—Automatic feature.

- Call Coverage

  Intercom calls are redirected only if the caller activates Go to Cover.

- Data Privacy and Data Restriction

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. Intercept tone is received when the ICOM button is pressed under this condition.

- Dial Intercom

  This feature must be provided. Users assigned an Automatic Intercom button must be a member of the same Dial Intercom group as the destination extension number.

- Single-Digit Dialing and Mixed Station Numbering

  Prefixed extensions greater than five digits (including the prefix) in length cannot be assigned to intercom lists.

Administration

Automatic Intercom is assigned on a per-voice terminal basis by the System Manager. Before Automatic Intercom can be assigned, the associated Intercom group must be established. Each Intercom group requires the following administration:

- Intercom group number
- Length of dial code
- Extension number within the group
- Dial codes to access Intercom group members.

Once the Intercom group is established, Automatic Intercom buttons can be assigned to members of the group. The following items must be administered for each button.

- Intercom group number to be accessed
- Dial code assigned to group member to be accessed.

Hardware and Software Requirements

No additional hardware or software is required.

Intercom—Dial

Description

Allows multi-appearance voice terminal users to gain rapid access to as many as 32 other voice terminal users within an administered group. Calling voice terminal users lift the handset, press the Dial Intercom button, and dial the 1- or 2-digit code assigned to the desired party. The called user receives alerting tone, and the status lamp associated with the Intercom button, if provided, flashes.

Considerations

With Dial Intercom, a group of users who frequently call each other can do so by pressing a Dial Intercom button and dialing a 1- or 2-digit code instead of dialing an extension number.

Up to 32 Intercom groups can be established. Each group can have up to 32 members, with a maximum of 128 members per system.

Single-line voice terminals can receive Dial Intercom calls, but cannot originate them.

A combination of Dial and Automatic Intercom can be used between terminals so that Dial Intercom applies in one direction and Automatic Intercom applies in the other.

A Dial Intercom user can place an intercom call to all members in the group, including Automatic Intercom members.

Two terminals with Dial Intercom to and from each other, or two terminals with combined Dial and Automatic Intercom to and from each other, must be in the same Intercom group.

Interactions

The following features interact with the Intercom—Dial feature.

- Automatic Intercom

  Users assigned this feature must be a member of a Dial Intercom group.

- Call Coverage

  Intercom calls are redirected to Call Coverage only if the caller activates Go To Cover.

- Single-Digit Dialing and Mixed Station Numbering

  Prefixed extensions greater than five digits (including the prefix) in length cannot be assigned to intercom lists.

Administration

Dial Intercom is administered by the System Manager. The following items require administration:

- Intercom groups
  - Group number
  - Length of dial code
  - Extension numbers within the group
  - Dial codes to access Intercom group members.
- Dial Intercom buttons.

Hardware and Software Requirements

No additional hardware or software is required.

Inter-PBX Attendant Calls (V2, V3, or G1)

Description

Allows attendant positions for more than one branch location to be concentrated at one central, or main, location. Incoming trunk calls to the branch location, as well as attendant-seeking voice terminal calls, are routed over tie trunks to the attendants at the main location.

Inter-PBX Attendant Calls are incoming tie trunk calls from the branch location to the main location with the attendant group as the destination. If no attendant in the attendant group is available, these calls are queued. When an attendant becomes available, the call is routed to that attendant for handling. Calls can be extended the same as if the call is an incoming call to the main location. When the attendant releases the call, the tie trunk associated with the call is tied up with the call until the call is dropped.

A System 75 or DEFINITY Generic 1 can be a branch or a main location for this feature. A branch location can have local attendants. These local attendants can be accessed by the Individual Attendant Access (V2, V3, or G1) feature. The attendants at the main location are the local attendants for the main location and also the attendant for the Inter-PBX Attendant Calls.

Considerations

With Inter-PBX Attendant Calls, the number of attendants required at each branch location is reduced. Also, users at each branch location can use the main location to access each of the other branch locations.

The Inter-PBX Attendant Calls feature can also be used within an Electronic Tandem Network (ETN), where, for example, the attendant group for the network could be located at the main switch and serve other tandem switches connected by tie trunks.

Interactions

The following features interact with the Inter-PBX Attendant Calls feature.

- Attendant Control of Trunk Group Access

  If Inter-PBX Attendant Calls is enabled, and a call at a branch location attempts to access a controlled trunk group, the call is routed to the local attendant at the branch location, if there is one. If there is no local attendant, the call is routed to the attendant group at the main location.

- Attendant Display and Distributed Communications System (DCS) Attendant Display

  In a DCS environment, an incoming Inter-PBX Attendant Call from a branch location is displayed at the attendant console the same as a local call.

In a non-DCS environment, an incoming Inter-PBX Attendant Call is displayed at the attendant console as an incoming tie trunk call.

- Attendant Recall

  If an attendant at the main location holds an Inter-PBX Attendant Call, the calling parties at the branch location cannot recall the attendant.

- Call Coverage

  At a branch location with Inter-PBX Attendant Calls enabled, a call redirected to a coverage path with the attendant group as a coverage point will skip that coverage point. It will go to the next coverage point at the local switch, if administered, or will continue to ring at the extension that is the previous coverage point. If the attendant group "0" is the only coverage point, it will continue to ring at the principal's extension.

- Centralized Attendant Service (CAS) (V2, V3, or G1)

  CAS and Inter-PBX Attendant Calling cannot be enabled at the same time.

- Dial Access to Attendant

  When a user at a branch location dials a single digit "0" and the Inter-PBX Attendant Calls feature is enabled, the call is routed to an attendant at the main location.

- Night Service

  The Inter-PBX Attendant Calls feature is deactivated when the branch location is put into night service, and reactivated when the branch location is taken out of night service.

Administration

Inter-PBX Attendant Calls is administered by the System Manager. The following items require administration:

- Branch location access to Inter-PBX Attendant Calls
- Inter-PBX Attendant Calls Trunk Group
- Inter-PBX Attendant Access Code.

Hardware and Software Requirements

Requires a tie trunk group between the branch and main locations. No additional software is required.

Intraflow and Interflow (V3 or G1)

Description

Allows Automatic Call Distribution (ACD) calls to be redirected from one split to another split under busy or unanswered conditions. Intraflow provides redirection of ACD calls to other splits within the system. Interflow uses the Call Forwarding All Calls feature to redirect ACD calls to an external location.

Intraflow allows splits to be assigned coverage paths. Also, a split can be a point in a coverage path. Thus, Intraflow uses the Call Coverage feature to redirect ACD calls from one split to another split according to the coverage path's redirection criteria. For instance, a split's coverage path can be administered so that incoming ACD calls are automatically redirected to another split during busy or unanswered conditions.

An ACD call is intraflowed to another split whenever the assigned Call Coverage redirection criteria is met. For a detailed description of Call Coverage redirection criteria, see the Call Coverage feature description elsewhere in this chapter.

If an ACD call is intraflowed to another split, the system attempts to terminate the call to an available agent. If an agent is not available, the system tries to place the call in queue at the covering split. The call enters the covering split queue, unless the split's "inflow threshold" is met or the queue is full. The inflow threshold is a parameter that is assigned to each split. If the oldest call in the split queue has remained in that queue for a length of time greater than the inflow threshold (0 to 999 seconds), then ACD calls cannot be intraflowed into that split. If an ACD call meets the Call Coverage redirection criteria, but cannot be intraflowed to another split or point in the coverage path, it will remain in queue at the original split even though coverage tone may be heard.

A split can be administered such that ACD calls intraflowed from that split to a covering split have priority over other calls in queue at that split. If an ACD call intraflows from a split with "priority on Intraflow" to a covering split, and enters the queue at the covering split, that call is positioned in the queue ahead of any nonpriority calls but behind other priority calls already in the queue. In other words, all priority calls are answered before any nonpriority calls.

If the covering split is assigned a second delay announcement, an ACD call intraflowed to that split will receive the announcement, if the call remains in queue for a length of time equal to the second delay announcement interval. After the announcement is heard, the caller hears either music-on-hold or silence until the call goes to an agent. An ACD call intraflowed to a covering split is never connected to the first delay announcement assigned to the covering split.

As an illustration of how Intraflow works, assume the following:

- A call is intraflowed from split 1 to split 2.
- Split 1 is assigned priority on intraflow.

- Split 2 has a queue with three priority calls and four nonpriority calls.
- Split 2 has an inflow threshold of 90 seconds and the oldest call in queue at split 2 has been in queue for 60 seconds.
- Split 2 has been assigned a second delay announcement and has a second delay announcement interval of 45 seconds.
- Music-on-Hold is provided.

When the call is intraflowed from split 1 to split 2, the call enters the split 2 queue and is positioned in the queue ahead of the four nonpriority calls. The intraflowed call is then the fourth call in queue. Assume the call stays in the queue for 45 seconds and is still not answered. The call, at the end of 45 seconds, is connected to the second delay announcement for split 2. When the announcement is complete, the caller hears music-on-hold until the call is connected to an available agent. If the second delay announcement is administered to repeat, the system will attempt to connect the call to the second delay announcement again after the delay interval has expired (45 seconds.)

The Interflow feature allows ACD calls to be redirected from one split to a split on another switch or to another external location. This is accomplished by forwarding calls that are directed to the split extension to an off-premises location via the Call Forwarding All Calls feature. For details on how calls are forwarded to an off-premises extension, see the Call Forwarding All Calls (V2, V3, or G1) feature description elsewhere in this chapter.

A Coverage Incoming Call Identification (ICI) button can be assigned to an agent's multi-appearance voice terminal. The Coverage ICI button allows the agent to identify a call that is intraflowed from another split. When an agent receives a call that has intraflowed from the split assigned to that button, the button's status lamp will light.

Considerations

Intraflow and Interflow provide the means to redirect ACD calls to alternate splits. Intraflow provides for redirection of ACD calls when certain conditions are met (such as Busy or Don't Answer). Therefore, calls can be directed to less busy splits, resulting in more efficient call handling. Interflow provides for all ACD calls to a specific split to be redirected to a split at another location.

The inflow threshold associated with Intraflow can be from 0 to 999 seconds.

Interactions

The following features interact with the Intraflow and Interflow features.

- Attendant Display and Voice Terminal Display

  These features provide call and queue identification for the covering split agents.

- Automatic Call Distribution

  When Intraflow is provided, the Coverage Don't Answer Interval (1 to 99 maximum ringing cycles) associated with Call Coverage may begin when the call enters the split queue. If the Coverage Don't Answer Interval expires before either of the two delay announcement intervals expires, the call is redirected to coverage. If no coverage point is available to handle the call, the call remains in queue and may then be connected to a delay announcement. If either of the delay announcement intervals expires before the Coverage Don't Answer Interval, the call is connected to a delay recorded announcement, if available.

- Call Pickup

  Any ACD call redirected to a covering split agent who is a member of a Call Pickup group cannot be answered by other members of the Call Pickup group.

- Temporary Bridged Appearance

  If an ACD call terminates to a split agent, but is intraflowed to another split before being answered, the Temporary Bridged Appearance at the split agent's terminal or console is no longer maintained.

Administration

Intraflow and Interflow are administered by the System Manager. The following items require administration:

- Coverage Paths

  The same coverage path can be used for as many splits as desired. For efficient operation of the Intraflow feature, it is recommended that the redirection criteria for a split's coverage path be administered so that calls are redirected under busy or don‘t answer conditions.

- Don't Answer Interval and Don't Answer Interval for Subsequent Redirection

  The Don't Answer Interval specifies the number of ringing cycles heard at the agent's terminal before the call is redirected to the first coverage point. This interval is recommended to be two rings but can be administered from 1 to 99 rings. All splits with the same coverage path are assigned the same Don't Answer Interval.

  The Don't Answer Interval for Subsequent Redirection specifies the number of rings at a covering split before the call attempts to redirect to the next coverage point. This interval is recommended to be two rings but can be administered from 1 to 99 rings. This interval is administered as a system parameter.

- Whether or not each split has priority on Intraflow

- Inflow threshold

- Coverage ICI buttons as required
- Another items listed under administration of the ACD feature.

Hardware and Software Requirements

No additional hardware is required. ACD software is required.

Last Number Dialed

Description

Automatically redials the last number dialed when users press the Last Number Dialed button or dial the Last Number Dialed feature access code.

The system saves the first 16 digits (V1) or 24 digits (V2, V3, or G1) of the last number dialed whether the call attempt was manually dialed or an Abbreviated Dialing button was pressed.

Considerations

Last Number Dialed prevents the user from having to redial a busy number. If a user has dialed a busy number and that was the last number dialed, the user simply activates Last Number Dialed by button or dial access code. The system automatically dials the same number again.

Special characters (Pause, Wait, Mark, or Suppress) stored in an Abbreviated Dialing button are recognized by the system and will be outpulsed when such a number is automatically redialed by the Last Number Dialed feature.

When a manually dialed number is redialed automatically, a delay in dialing is not recorded. The system will outpulse the numbers as one continuous digit string. Thus, to accomplish automatic redialing, the distant end must accept the outpulsed digits without delay.

Last Number Dialed information is not saved on tape and can be used only for the next call origination. End-to-end signaling digits manually dialed are never saved.

Interactions

The following features interact with the Last Number Dialed feature.

- Abbreviated Dialing

  With Version 2, Version 3, or DEFINITY Generic 1, if the previously called number was in an Abbreviated Dialing privileged list, and if the user is not normally allowed to dial the number because of his or her Class of Restriction, Intercept Treatment is given when using Last Number Dialed. To redial the number, the user must again use the Abbreviated Dialing privileged list.

- Automatic Callback

  Automatic Callback can be used after the Last Number Dialed feature is used on a call to an internal voice terminal.

- Bridged Call Appearance

  Activation of the Last Number Dialed feature causes the last number dialed from the voice terminal to be redialed, regardless of which extension number is used (primary or bridged call appearance).

Administration

Last Number Dialed is administered by the System Manager. The following items require administration:

- Feature Access Code for Last Number Dialed
- Last Number Dialed button.

Hardware and Software Requirements

No additional hardware or software is required.

Leave Word Calling

Description

Allows internal system users to leave a short preprogrammed message for other internal users. Users can activate Leave Word Calling (LWC) at any time during a call attempt.

The LWC feature electronically stores a standard message, for example, CARTER, ANN 2/7 10:45a 2 CALL 3124. This message means that Ann Carter called two times, the last time on the morning of February 7, and wants a return call to extension 3124.

When a message is stored, the Message lamp on the called voice terminal automatically lights. This lamp is referred to as an Automatic Message Waiting lamp since the status of the lamp is controlled automatically by the system.

Another voice terminal may also receive an indication that an LWC message has been left for the called party. This is accomplished via a remote Automatic Message Waiting lamp at another voice terminal. The remote Automatic Message Waiting lamp is a status lamp associated with a button assigned for this purpose. The remote Automatic Message Waiting lamp lights at the same time that the Message lamp lights at the called voice terminal. A common use of a remote Automatic Message Waiting lamp is to provide an indication of an executive's message on a secretary's voice terminal. If the executive calls from outside to receive any messages, the secretary knows at a glance if any messages have been left. Remote Automatic Message Waiting lamps also allow an indication of LWC messages left for a Direct Department Calling (DDC) group, a Uniform Call Distribution (UCD) group, an Automatic Call Distribution (ACD) split, a Terminating Extension Group (TEG), and a Personal Central Office Line (PCOL) group.

When identical messages are entered in the system, the date, time, and number of messages are updated. When nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

Messages can be stored by calling, called, and covering users. A covering user can be through the Call Coverage, Call Pickup, or Call Forwarding All Calls features. Messages are stored as follows:

- Storage by Calling User
  - Before dialing the desired extension number, the user presses the LWC button or dials the LWC access code and then dials the desired number.
  - After dialing the desired number but before the call is answered, a multi-appearance voice terminal user presses the LWC button or a single-line voice terminal user presses the Recall button and dials the access code.
  - After the call has been answered by any user, the calling user presses the LWC button or the Recall button and dials the access code.

- Storage by Called User
  - After answering the call, the called user presses the LWC button. This leaves a message for the calling user to call back. (A called user can store an LWC message by dialing the LWC access code only if the called user has an analog voice terminal.)
- Storage by Covering User
  - After answering the call, the covering user presses the Coverage Callback button. This stores a message for the called user to call the calling user.
  - After answering the call, the covering user presses the LWC button. This leaves a "call me" message for the originally called user.

In addition, a user placed on hold can activate LWC and leave a message for the holding user to place a return call.

Messages are retrieved by users who have the Voice Terminal Display or Attendant Display feature. Users without the Voice Terminal Display feature have their messages retrieved by systemwide message retrievers or by covering users in their Call Coverage path.

If an Applications Processor (AP) is provided with the system, LWC messages can be retrieved by a Message Center agent or by authorized users through the AP Demand Print feature.

If the following conditions are met, messages for users can be retrieved by selected voice terminal users or any attendant:

- The retriever must be in the called user's Call Coverage path or must be administered as a systemwide retriever.
- Permission to retrieve messages must be administered to the called voice terminal.

A calling user who left an LWC message can cancel that message if it has not already been accessed. The calling user lifts the handset, presses the LWC Cancel button or dials the access code, and dials the extension number of the called party. This deletes the message (even if the count was more than one) and causes all Message lamps associated with the called voice terminal to go dark (if the called user has no other messages).

Messages are protected by restricting unauthorized users from displaying, canceling, or deleting messages. A Lock function restricts a voice terminal, and an Unlock function releases the restriction. The Lock function is activated by dialing a systemwide access code. The Lock function is canceled by dialing a systemwide access code and then an Unlock security code unique to the voice terminal. These functions apply only to the voice terminal where the operation is performed. A status lamp can be assigned to show the locked or unlocked status of the voice terminal.

Considerations

Leave Word Calling lets users automatically leave short, simple messages for other users. When a voice terminal's message lamp is lighted, the user simply has the message retrieved by an authorized user. This reduces the time spent making handwritten notes.

Ten terminals, or nine terminals and the attendant console group, can be administered as systemwide message retrievers.

A system maximum of 1000 (V1) or 2000 (V2, V3, or G1) messages can be stored by the system (without an AP), and a systemwide maximum number of messages not to exceed 125 per user can be administered.

If the system does not have an AP and if the stored message level reaches 95 percent of capacity, the status lamp associated with all Coverage Message Retrieval buttons in the system will flash. These lamps will continue to flash until the stored message level drops below 85 percent capacity. Authorized retrievers can selectively delete messages to gain storage space. Old messages are not automatically purged by the system.

Interactions

The following features interact with the Leave Word Calling feature.

- Audio Information Exchange (AUDIX) Interface

  Leave Word Calling Cancel cannot be used to cancel an AUDIX message.

- Bridged Call Appearance

  A Leave Word Calling message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension number assigned to the bridged call appearance. When a user calls a primary extension, and activates Leave Word Calling, the message is left for the primary extension, even if the call was answered at a bridged call appearance.

- Call Coverage

  The LWC feature can be used with or without Call Coverage. However, the two features complement each other. The Coverage Callback option of the Call Coverage feature is provided by the LWC feature. Also, a caller can activate LWC for the called party even if the call has been answered by a covering user.

- Conference/Transfer

  A member of a conference call cannot activate LWC because that user cannot be uniquely identified.

  With G1, after Leave Word Calling has been activated for a party on a conference or transfer, the conference/transfer originator cannot press the Conference/Transfer button a second time to return to the original call. The conference/transfer originator must select the call appearance button to return to the previously held call.

Administration

Leave Word Calling is administered by the System Manager. The following items require administration:

- AP Demand Print button (per voice terminal)
- Identities of authorized systemwide LWC retrievers
- Locking and unlocking message retrieval and cancellation (per voice terminal)
- Lock dial access code (systemwide)
- Lock status lamp (per voice terminal)
- LWC activation (per voice terminal and the attendant group)
- LWC activation dial access code (systemwide)
- LWC button (per voice terminal)
- LWC Cancel button (per voice terminal)
- LWC cancellation dial access code (systemwide)
- LWC reception (per voice terminal and per Hunt group, that is, DDC group, UCD group, TEG, and PCOL group)
- Maximum number of messages not to exceed 125 per user (systemwide)
- Remote Automatic Message Waiting lamp on another voice terminal (one allowed per extension number [V1 or V2] or 80 allowed per extension [V3, or G1], including an extension number for a DDC group, UCD group, TEG, and PCOL group; 50 [V1] or 80 [V2, V3, or G1] allowed per system)
- Retrieval permission for covering users (per voice terminal)
- Unlock dial access code (systemwide)
- Unlock security code (per voice terminal).

All buttons associated with the display modes are administered through the Attendant Display and Voice Terminal Display features.

Hardware and Software Requirements

No additional hardware or software is required.

Line Lockout

Description

Removes single-line voice terminal extension numbers from service when users fail to hang up after receiving dial tone for 10 seconds and then intercept tone for 30 seconds.

Line Lockout occurs as follows:

- A user does not hang up after the other party on a call is disconnected.

  In this case, the user will receive dial tone for 10 seconds and then will receive intercept tone for 30 seconds. The voice terminal is then taken out of service, if the handset is still lifted.

- A user pauses for 10 seconds between digits while dialing.

  In this case, the user will receive intercept tone for 30 seconds. The voice terminal is then taken out of service, if the handset is still lifted.

The out-of-service condition remains in effect until the voice terminal user hangs up.

Considerations

The out-of-service condition provided by Line Lockout does not tie up switching facilities or call processing time. The facilities are then available for other users.

This feature does not apply to multi-appearance voice terminals.

Interactions

Call intercept is provided by the Intercept Treatment feature.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Loudspeaker Paging Access

Description

Provides attendants and voice terminal users dial access to voice paging equipment.

As many as nine individual paging zones can be provided by the system. (A zone is the location of the loudspeakers, for example, conference rooms, warehouses, or storerooms.) In addition, one zone can be provided by the system to activate all zones simultaneously.

Each of the ten zones provided by the system is assigned an individual trunk access code (TAC). The TACs are used to activate Loudspeaker Paging Access. A user can activate Loudspeaker Paging Access by dialing the TAC of the desired paging zone. In addition, the TACs can be stored in Abbreviated Dialing lists. This allows multi-appearance voice terminals to activate the feature via Abbreviated Dialing buttons. Attendants can use a Direct Trunk Group select button to activate Loudspeaker Paging Access, if the desired paging zone's TAC is assigned to one of the buttons.

Once a user has activated Loudspeaker Paging Access for the desired zone, the user can speak into the handset and make the announcement.

In addition to, or instead of the system loudspeaker paging equipment, a PagePac paging system can be used. A PagePac paging system has a distinct advantage over the switch paging system in that a PagePac system requires only one port on one circuit pack to provide as many as 39 paging zones. The switch paging system requires a separate port for each paging zone with a maximum of nine zones. Three different PagePac paging systems are available for use with System 75 or DEFINITY Generic 1:

- PagePac 20

  This is the smallest PagePac system. The basic system provides a single paging zone with an input source for music over the paging system. The music can also serve as the music for the Music-on-Hold Access feature. Additional add-on hardware is available to provide multi-zone paging for 3, 9, or 39 paging zones.

- PagePac VS

  This system provides one to three paging zones. It also permits the paging of all zones simultaneously. Additional hardware is available to provide music and/or talkback over the paging system.

- PagePac 50/100/200

  This system provides up to 24 paging zones. Additional hardware is available to provide music and/or talkback over the paging system.

PagePac equipment is also easy to use. A user simply dials the extension number (PagePac 50/100/200 only) or TAC assigned to the PagePac system. This connects the user to the PagePac equipment. If there is only one paging zone, the user then uses the handset of the voice terminal to page someone. If there are multiple zones, the user, after hearing a steady tone, dials a 1- or 2-digit code to access the desired zone(s) before paging.

Considerations

With Loudspeaker Paging Access, a user can be paged at any location with loudspeaker paging equipment. This feature is particularly useful when used in conjunction with the Call Park feature. When a user is away from his or her location and receives a call, an incoming call can be answered and parked by another user. The called party can then be paged and told what extension number the call is parked on. The called party can then answer the parked call from a nearby voice terminal.

The system can have up to nine individual zones plus one zone to activate all zones simultaneously. A PagePac paging system can be used to provide up to 39 paging zones.

A Listed Directory Number or Direct Inward Dialing call cannot be connected to the paging facility. However, the attendant can make the page and park the incoming call using the Call Park feature.

Interactions

The following features cannot be used with Loudspeaker Paging:

- Attendant Conference
- Terminal Conference
- Data Call Setup
- Hold
- Ringback Queuing
- Transfer.

Normally, a call to a busy single-line voice terminal results in a call waiting tone being heard by the called voice terminal user. If that user is in the process of paging, the call waiting tone is not heard.

It is not possible to use a PagePac paging system for Code Calling Access when multi-zone paging is desired. The PagePac systems expect a 2-digit code to access a particular zone. The system, however, immediately plays the chime code once a connection is established.

Administration

Loudspeaker Paging Access is administered by the System Manager. The following items require administration:

- The Deluxe Paging and Call Park Timeout to Originator field on the Feature Related System Parameters form must be administered as "no".
- Up to ten (one per zone) Loudspeaker Paging Access buttons (per multi-appearance voice terminal and attendant console). Buttons are assigned through the Attendant Direct Trunk Group Selection. Abbreviated Dialing, and Facility Busy Indication features.
- Trunk access codes and Class of Restriction (per zone provided).
- Paging expiration interval (from 10 seconds to 10 minutes).
- Station Message Detail Recording activation.

If a PagePac paging system is to be used, it must be assigned a TAC or extension number (PagePac 50/100/200 only).

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack for each zone. Paging interface equipment, consisting of a 278A adapter and a 24 volt power supply, is also required for each individual zone. (This hardware can be shared with the Code Calling Access feature. Each feature is activated by the assigned trunk access code.)

If PagePac equipment is used, one port on a TN747 CO Trunk circuit pack, TN742 Analog Line circuit pack, or TN763 Auxiliary Trunk circuit pack is required (depending on which PagePac system is used).

No additional software is required.

Loudspeaker Paging Access—Deluxe (G1)

Description

Provides attendants and voice terminal users with integrated dial access to voice paging equipment and Call Park capabilities.

When Loudspecker Paging Access—Deluxe (also called Deluxe Paging in the rest of this description) is activated, the call is automatically parked. The Call Park feature does not have to be activated separately. This is consistent with the Code Calling Access feature activation. In addition to the automatic Call Park capability, Deluxe Paging also lets parked calls return to the parking user with special distinctive alerting upon expiration of the Call Park Timeout interval.

Deluxe Paging also provides the Meet-Me Paging and Meet-Me Conferencing functions. With Meet-Me Paging, a user can simply activate Deluxe Paging, make the announcement for someone else to call him or her back, and hang up. When the paged party answers, he or she is connected to the paging party. With Meet-Me Conferencing, another party can easily be paged and added onto a conference call.

The customer has the option of having either normal Loudspeaker Paging Access (described elsewhere in this chapter) or Loudspeaker Paging Access—Deluxe (Deluxe Paging). This description describes only Deluxe Paging.

Paging Zones

As many as nine individual paging zones can be provided by the system. (A zone is the location of the loudspeakers, for example, conference rooms, warehouses, or storerooms.) In addition, one zone can be provided by the system to activate all zones simultaneously.

Each of the ten zones provided by the system is assigned an individual trunk access code (TAC). The TACs are used to activate Deluxe Paging. A user can activate Deluxe Paging by dialing the TAC of the desired paging zone. In addition, the TACs can be stored in Abbreviated Dialing lists. This allows multi-appearance voice terminals to activate the feature via Abbreviated Dialing buttons. Attendants can use a Direct Trunk Group select button to activate Deluxe Paging, if the desired paging zone's TAC is assigned to one of the buttons.

PagePac Paging

In addition to, or instead of, the system loudspeaker paging equipment, a PagePac paging system can be used. A PagePac paging system has a distinct advantage over the switch paging system in that a PagePac system requires only one port on one circuit pack to provide as many as 39 paging zones. The switch paging system requires a separate port for each paging zone with a maximum of nine zones. Three different PagePac paging systems are available for use with DEFINITY Generic 1:

- PagePac 20

  This is the smallest PagePac system. The basic system provides a single paging zone with an input source for music over the paging system. The music can also serve as the music for the Music-on-Hold Access feature. Additional add-on hardware is available to provide multi-zone paging for 3, 9, or 39 paging zones.

- PagePac VS

  This system provides one to three paging zones. It also permits the paging of all zones simultaneously. Additional hardware is available to provide music and/or talkback over the paging system.

- PagePac 50/100/200

  This system provides up to 24 paging zones. Additional hardware is available to provide music and/or talkback over the paging system.

PagePac equipment is also easy to use. A user simply dials the extension number (PagePac 50/100/200 only) or TAC assigned to the PagePac system followed by the extension number where the call is to be parked. This connects the user to the PagePac equipment. If there is only one paging zone, the user then uses the handset of the voice terminal to page someone. If there are multiple zones, the user, after hearing a steady tone, dials a 1- or 2-digit code to access the desired zone(s) before paging.

Operations

User operations vary depending on the type of voice terminal the user has and whether or not the user is an attendant. Therefore, the various user operations are described separately for single-line voice terminals, multi-appearance voice terminals, and attendants.

Activation of Deluxe Paging by Single-Line Voice Terminal Users

1. Go off-hook to get dial tone.

   If already on a call with another party, press the Recall button or flash the switchhook. The other party is placed on hold and recall dial tone is heard.

2. Dial the trunk access code for the desired paging zone. (Dial tone is heard.)

3. Dial the extension number where the call is to be parked. (Confirmation tone is heard and the call is temporarily parked.)

   To park the call on your own extension, dial a "#"instead of the extension number.

4. Make the announcement. (The loudspeaker paging timer starts.)

5. Press the Recall button before the administered Loudspeaker Paging Timeout Interval expires and go on-hook. (The paging equipment is released, the parked call is now waiting to be answered, and the timer for the Call Park Timeout Interval starts.)

If another party was on the call and was placed on hold in Step 1, that party and the paging party are in conference, are both parked on the call, and will both be connected to the paged party when he or she answers the call. (This is known as Meet-Me Conferencing.)

If the Loudspeaker Paging Timeout Interval expires before the Recall button is pressed, the paging user receives confirmation tone, the paging equipment is released, the call is automatically parked on your extension, and the calling party hears music (if provided). When the paged party answers the call, he or she is connected to the paging party. The paging party can then transfer the call to the calling party.

If the Call Park Timeout Interval expires, the call returns to the paging user with the proper distinctive alerting (one-burst for internal calls and conference calls with both internal and external parties; two-burst for external calls).

If no answer-back is required on the call, hang up instead of pressing the Recall button. The parked call is dropped and the paging equipment is released.

Activation of Deluxe Paging by Multi-Appearance Voice Terminal Users

1. Go off-hook to get dial tone.

   If already on a call with another party, press the Transfer button. The other party is placed on hold and dial tone is heard.

2. Dial the trunk access code for the desired paging zone. (Dial tone is heard.)

3. Dial the extension number where the call is to be parked. (Confirmation tone is heard and the call is temporarily parked.)

   To park the call on your own extension, dial a "#" instead of the extension number.

4. Make the announcement. (The loudspeaker paging timer starts.)

5. Press the Transfer button before the administered Loudspeaker Paging Timeout Interval expires and go on-hook. (The paging equipment is released, the call is parked and is now waiting to be answered, and the timer for the Call Park Timeout Interval starts.)

   If another party was on the call and was placed on hold in Step 1, that party is parked on the call and will be connected to the paged party when he or she answers the call (answer-back). The Conference button can be pressed instead of the Transfer button to allow both the paging and held parties to be connected to the paged party on answer-back. (This is known as Meet-Me Conferencing.)

   If the Loudspeaker Paging Timeout Interval expires before the Transfer button is pressed, the paging user receives confirmation tone, the paging equipment is released, the call is automatically parked on your extension, and the calling party hears music (if provided). When the paged party answers the call, he or she is connected to the paging party. The paging party can then transfer the call to the calling party.

If the Call Park Timeout Interval expires, the call returns to the paging user with the proper distinctive alerting (one-burst for internal calls and conference calls with both internal and external parties; two-burst for external calls).

If no answer-back is required on the call, hang up instead of pressing the Transfer or Conference button. The parked call is dropped and the paging equipment is released.

Activation of Deluxe Paging by an Attendant for Another Party

1. Press the Start button. The other party is placed on hold and the attendant gets dial tone.

2. Dial the trunk access code for the desired paging zone. (The attendant gets dial tone.)

3. Dial the extension number where the call is to be parked. (Confirmation tone is heard and the call is temporarily parked.) The attendant can also dial a "#" instead of the extension number to park the call on his or her individual attendant extension (if assigned).

4. Make the announcement. (The loudspeaker paging timer starts.)

5. Press the Release button before the administered Loudspeaker Paging Timeout Interval expires. (The paging equipment is released, the parked call is now waiting to be answered, and the timer for the Call Park Timeout Interval starts. If the Split button is pressed, the timer for the Call Park Timeout Interval starts, and both the held party and the attendant are parked and will be connected to the call upon answer-back.)

   If the Loudspeaker Paging Timeout Interval expires before the Release button is pressed, the attendant receives confirmation tone, the paging equipment is released, and the call is automatically parked on the console.

   If the Loudspeaker Paging Timeout Interval expires before the Release button is pressed, the attendant receives confirmation tone, the paging equipment is released, the call is automatically parked on the console, and the calling party hears music (if provided). When the paged party answers the call, he or she is connected to the paging party. The paging party can then transfer the call to the calling party.

Activation of Deluxe Paging Answer-Back by the Paged Party

1. Go off-hook to get dial tone.

2. Dial the answer-back feature access code. (Dial tone is heard.)

3. Dial the extension number where the call is parked, or dial "#" if the call is parked on the extension you are using. (Music-on-Hold, if provided, is removed from the parked call, all parties associated with the parked call receive confirmation tone, and the answer-back and parked parties are connected.)

Unparking a Loudspeaker Paging Call

If a user wishes to unpark a loudspeaker paging call that is parked on his or her extension, this can be accomplished by pressing the lighted Call Park button.

Considerations

With Loudspeaker Paging Access—Deluxe, a user can be paged at any location with loudspeaker paging equipment. Integrated Call Park capabilities allow the paging party to park the call without dialing a separate Call Park feature access code. When a user is away from his or her location and receives a call, an incoming call can be answered by another user. The called party can then be paged and told what extension number the call is parked on. The called party can then answer the parked call from a nearby voice terminal.

The system can have up to nine individual zones plus one zone to activate all zones simultaneously. A PagePac paging system can be used to provide up to 39 paging zones.

A Listed Directory Number or Direct Inward Dialing call cannot be connected to the paging facility. However, the attendant can make the page and park the incoming call.

For non-local access of Deluxe Paging (such as Remote Access users, tie trunk users, etc.) the "#" cannot be used to park the call on your own extension.

Interactions

The following features interact with the Loudspeaker Paging Access—Deluxe feature.

- Bridged Call Appearance

  If the parked call includes a shared Terminating Extension Group (TEG), a shared Personal Central Office Line, and/or a redirected call with a Temporary Bridged Appearance, the maximum number of off-hook parties on the call is five, instead of six. The sixth position is reserved for the answer-back call.

- Call Coverage

  If a coverage call is parked by Deluxe Paging, the Temporary Bridged Appearance at the principal extension is maintained as long as the covering user remains off-hook or places the call on hold.

- Call Park

  A call cannot be parked on more than one extension at the same time.

  More than one call cannot be parked at the same extension at any given time. If a user tries to park a Deluxe Paging call on an extension that already has a parked call, that user receives intercept treatment.

  The Call Park feature provides up to ten common shared extensions for use by attendants or by voice terminal extensions with console permissions. These extension numbers are not assigned to a voice terminal, but are stored in system translations and used to park a call. These extension numbers are particularly useful when one party is paged at the request of another party. The calling party is parked by Deluxe Paging and the extension number is announced. Common shared extensions should be assigned to the optional selector console in the 00 through 09 block (bottom row) in any hundreds group that the attendant can easily identify. The lamp associated with the extension number will identify call parked or no call parked (instead of active or idle status).

  If the Call Park Timeout Interval expires during Deluxe Paging, the call normally returns to the originator of the Deluxe Paging call. However, with Remote Access and Tie Trunk Access, the call goes to the attendant.

- Call Pickup

  If a Call Pickup call is parked by Deluxe Paging, the Temporary Bridged Appearance at the principal extension is maintained as long as the answering pickup group member remains off-hook or places the call on hold.

- Call Waiting Termination

  Normally, a call to a busy single-line voice terminal results in a call waiting tone being heard by the called voice terminal user. If that user is in the process of paging, the call waiting tone is not heard.

- Code Calling Access

  It is not possible to use a PagePac paging system for Code Calling Access when multi-zone paging is desired. The PagePac systems expect a 2-digit code to access a particular zone. The system, however, immediately plays the chime code once a connection is established.

- Conference—Attendant

  The maximum number of conferees on a parked Deluxe Paging call is five. The sixth conferee position is reserved for the answer-back call.

  A Deluxe paging call cannot be conferenced unless a party was placed on hold and parked with the call. The reason for this is that paging equipment cannot be placed on hold.

- Conference—Terminal

  For multi-appearance voice terminals, the maximum number of conferees on a parked Deluxe Paging call is five. The sixth conference position is reserved for the answer-back call.

  Single-line voice terminals can have a maximum of two conferees on a parked Deluxe Paging call.

  A Deluxe paging call cannot be conferenced unless a party was placed on hold and parked with the call. The reason for this is that paging equipment cannot be placed on hold.

- Data Call Setup

  If the Data button has been pressed as a pre-indication for modem pooling, access to Deluxe paging is denied.

- Data Privacy

  If a call, with Data Privacy activated, is parked by Deluxe Paging, Data Privacy for that call is automatically deactivated.

- Direct Inward Dialing (DID)

  A DID call cannot be connected to a Deluxe Paging facility.

- Hold

  Deluxe Paging facilities cannot be placed on hold.

- Hunt Groups

  If a hunt group member parks a call using Deluxe Paging, the call is parked on the member's own individual extension, not the hunt group extension.

- Leave Word Calling

  If a user parks a call for his or her extension with the Conference button, any parking or parked parties cannot activate Leave Word Calling because that party cannot be uniquely identified.

- Manual Originating Line Service

  Users that are assigned Manual Originating Line Service cannot access Deluxe Paging.

- Music-on-Hold Access

  Music-on-Hold, if provided, is connected to the parked party when there is only one conferee left on the parked call. Music-on-Hold is not connected to a parked call with more than one conferee.

- Multiple Listed Directory Numbers

  A Listed Directory Number call cannot be connected to a Deluxe Paging facility.

- Night Service

  If a night station user parks a Night Service call with Deluxe Paging, the call is parked on the night station's primary extension.

- Personal Central Office Line (PCOL)

  If a PCOL call is parked by Deluxe Paging, the Temporary Bridged Appearance of the call is maintained at the PCOL extension until the call is disconnected.

- Remote Access

  Remote Access users can access Deluxe Paging unless they are restricted by Class of Restriction from doing so.

- Ringback Queuing

  Ringback Queuing is not provided for Deluxe Paging.

- Terminating Extension Group (TEG)

  If a TEG member parks a call using Deluxe Paging, the call is parked on the member's own individual extension, not the TEG extension.

- Transfer

  A Deluxe paging call cannot be transferred unless a party was placed on hold and parked with the call. The reason for this is that paging equipment cannot be placed on hold.

Administration

Deluxe Paging is administered by the System Manager. The following items require administration:

- The Deluxe Paging and Call Park Timeout to Originator field on the Feature Related System Parameters form must be administered as "yes".

- Up to ten (one per zone) Deluxe Paging buttons (per multi-appearance voice terminal and attendant console). Buttons are assigned through the Attendant Direct Trunk Group Selection, Abbreviated Dialing, and Facility Busy Indication features.
- Trunk access codes and Class of Restriction (per zone provided).
- Answer-back access code.
- Paging expiration interval (from 10 seconds to 10 minutes).
- Call Park expiration interval (from 1 to 90 minutes in intervals of 1 minute).
- Station Message Detail Recording activation.
- Console permissions to allow voice terminal extensions to park calls on common shared extensions (assigned via Class of Service).
- Up to 10 common shared extension numbers.

If a PagePac paging system is to be used, it must be assigned a trunk access code or extension number (PagePac 50/100/200 only).

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack for each zone. Paging interface equipment, consisting of a 278A adapter and a 24-volt power supply, is also required for each zone. (This hardware can be shared with the Code Calling Access feature. Each feature is activated by the assigned trunk access code.)

If PagePac equipment is used, one port on a TN747 CO Trunk circuit pack, TN742 Analog Line circuit pack, or TN763 Auxiliary Trunk circuit pack is required (depending on which PagePac system is used).

No additional software is required.

Manual Message Waiting

Description

Enables multi-appearance voice terminal users, by pressing a designated button on their own terminals, to light the status lamp associated with the Manual Message Waiting button at another multi-appearance voice terminal. Activating the feature causes the lamp to light on both the originating and receiving voice terminals. Either terminal user can cause the lamp to go dark by pressing the button.

Considerations

This feature can be administered only to pairs of voice terminals, such as a secretary and an executive. The secretary might press the designated button to signal the executive that a call needs answering. The executive might press the button to indicate "Do Not Disturb" or "Not Available" to the secretary. (The button can be marked to reflect the intended use.)

Interactions

None.

Administration

Manual Message Waiting is administered on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Manual Message Waiting buttons to the voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Manual Originating Line Service

Description

Connects single-line voice terminal users to the attendant automatically when the user lifts the handset.

The attendant code is stored in an Abbreviated Dialing list. When the Manual Originating Line Service voice terminal user lifts the handset, the system automatically routes the call to the attendant using the Hot Line Service feature.

A Manual Originating Line Service user can receive calls allowed by the assigned Class of Restriction. Call reception is not affected by Manual Originating Line Service.

Considerations

Manual Originating Line Service is useful in any application where all call originations are screened by the attendant. The user simply lifts the handset and is connected to the attendant.

The number of single-line voice terminals that can be assigned Manual Originating Line Service is not limited.

Interactions

A Manual Originating Line Service call is a Hot Line Service call to the attendant.

A Manual Originating Line Service voice terminal user cannot activate features that require dialing.

When a Night Service feature is activated, the Manual Originating Line Service call redirects.

Administration

Manual Originating Line Service is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Abbreviated Dialing Lists (the attendant code must be a list entry)
- Hot Line Destination.

Hardware and Software Requirements

No additional hardware or software is required.

Manual Signaling

Description

Allows a voice terminal user to signal another voice terminal user. The receiving voice terminal user hears a 2-second burst of tone.

The signal is sent each time the button is pressed. If the receiving voice terminal is already being rung with an incoming call, Manual Signaling is denied. The status lamp associated with the Manual Signaling button at the originating voice terminal will flutter briefly to indicate the denial.

Considerations

With Manual Signaling, one voice terminal user can signal another voice terminal user. The meaning of the signal is prearranged between the sender and the receiver.

When a voice terminal user presses the Manual Signaling button, the associated status lamp lights for 2 seconds.

Interactions

None.

Administration

Manual Signaling is assigned on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Manual Signaling button to the originating voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

Modem Pooling

Description

Allows switched connections between digital data endpoints (data modules) and analog data endpoints, and acoustic coupled modems. The analog data endpoint can be either a trunk or line circuit.

Data transmission between a digital data endpoint and an analog endpoint requires a conversion resource since the Digital Communications Protocol (DCP) format used by the data module is not compatible with the modulated signals of an analog modem. The conversion resource translates the DCP format into modulated signals and vice versa.

The Modem Pooling feature provides pools of conversion resources.

Integrated conversion resources (Versions 1, 2, and 3 and DEFINITY Generic 1) and combined conversion resources (Version 2, Version 3, and DEFINITY Generic 1) are available with the system. The integrated type has functionality integrated on the TN758 Pooled Modem circuit pack, which provides two conversion resources and each one emulates a Trunk Data Module (TDM) cabled to a 212 Modem. The combined type is a TDM cabled to any TDM-compatible modem to provide a conversion resource.

When a conversion resource is required, the system queries the digital data module associated with the call to determine if its options are compatible with those supported by the pools. If the data module options are not compatible, the originating user receives intercept treatment. If the options are compatible, the system obtains a conversion resource from the appropriate pool. If a conversion resource is not available, the user receives reorder treatment. If all data calls, including analog, are not successfully established, the call will be disconnected within 15 seconds (handshake time-out).

In almost all cases, the system can detect the need for a conversion resource. Data calls originated from an analog data endpoint to a digital data endpoint require that the user indicate the need for a conversion resource, since the system considers an analog call origination as a voice call. This need is indicated by dialing the Data Origination Access Code before dialing the digital data endpoint. Use of Data Call Preindication before One-Button Transfer To Data is recommended when establishing data calls that use toll network facilities. Needed conversion resources are reserved before any toll charges are incurred.

Version 2, Version 3, or DEFINITY Generic 1 provides a "HOLD Time" parameter to specify the maximum time any conversion resource may be held but not used (while a data call is in queue).

Version 1 Modem Pooling supports asynchronous transmissions at 0 to 300 (LOW), 300 and 1200 bps, and synchronous transmission at 1200 bps.

Version 2, Version 3, or DEFINITY Generic 1 with combined conversion resources additionally supports the following configurations:

- IBM bisynchronous protocols typically used in 3270 and 2780/3780 applications. Both require 2400 bps or 4800 bps, half duplex, synchronous transmission.
- Interactive IBM-TSO applications using 1200 bps, half duplex, asynchronous transmissions.
- DATAPHONE II data communications service switched network modems supporting asynchronous and synchronous communications, and autobaud at 300, 1200, or 2400 bps.
- Version 2, Version 3, or DEFINITY Generic 1 can operate up to 19.2 kbps.
- Different pools (V2, V3, or G1) can have different data transmission characteristics.

The following modem options are supported by the integrated (only) pool:

- Receiver Responds to Remote Loop
- Loss of Carrier Disconnect
- Send Space Disconnect
- Receive Space Disconnect
- CF-CB Common
- Options TDM and Modem, for Combined Conversion resources (on devices)
- Speed, Duplex, and Synch (administered).

Considerations

Modem Pooling offers a pool of conversion resources which increase data call flexibility. Conversion resources allow analog data endpoints, using modems, to communicate with digital data endpoints (using data modules). Also, pooling of conversion resources allows maximum use of such facilities.

Data Call Preindication is recommended for off-premises data calls involving toll charges.

On data calls between a data module and an analog data endpoint, Return-to-Voice releases the conversion resource and returns it to the pool. The voice terminal user is then connected to the analog data endpoint.

For traffic purposes, Version 2, Version 3, or DEFINITY Generic 1 accumulates data on modem pooling calls separate from voice calls. Measurements on the pools are also accumulated.

Version 1 can support one pool of 32 integrated conversion resources. Version 2, Version 3, or DEFINITY Generic 1 can support up to five pools; all five combined, integrated, or any mix. Each pool has a capacity of up to 32 conversion resources. Version 2, Version 3, or DEFINITY Generic 1 has a limit of 160 conversion resources for all 5 pools.

Use of Modem Pooling cannot be restricted. Also, queuing for conversion resources is not provided, although calls queued on a hunt group retain reserved conversion resources while queued.

Mixing of modems from different vendors within a combined pool should be avoided since a difference in transmission characteristics may exist. Mixing is possible, but satisfactory results are not guaranteed.

Data transmission characteristics (speed, duplex, and synchronization mode), as administered, must be identical to the TDM and modem optioning by the customer.

Each data call that uses Modem Pooling uses four time slots (not just two). As a result, heavy usage of Modem Pooling could affect the TDM bus blocking characteristics.

Interactions

The following features interact with the Modem Pooling feature.

- Data Call Setup

  Data calls to or from a TDM cannot use Modem Pooling.

- Data-Only Off-Premises Extensions

  Modem Pooling is not possible on calls to or from a Data-Only Off-Premises Extension, when this type of digital data endpoint uses a TOM.

- Data Privacy and Data Restriction

  With V1, V2, and V3, when a call using a modem pool is made, Data Privacy and/or Data Restriction is turned off. With G1, the insertion of a modem pool does not turn off Data Privacy and/or Data Restriction.

- Digital Multiplexed Interface (DMI)

  Data calls originated from a local analog data endpoint to a DMI trunk must dial the Data Origination Access Code to obtain a conversion resource. Data calls on DMI trunks to local analog data endpoints automatically obtain conversion resources.

- DS1 Tie Trunk Service (V2, V3, or G1)

  Conversion resources used for Modem Pooling can only be connected to Alternate Voice/Data (AVD) DS1 tie trunks via Data Terminal Dialing or by dialing the feature access code for data origination.

Administration

Modem Pooling is assigned on a per-pool basis by the System Manager. The following items require administration.

- Conversion Resources—For integrated conversion resources, assign Pooled Modem circuit packs. For combined conversion resources, assign TDM and associated modems ports, speed (up to three speeds), and duplex and synchronization characteristics.
- HOLD Time (per pool basis)—Specify the maximum time (1 to 99 minutes) any conversion resource may be held and not used (while a data call waits in a queue). Default value is 5 minutes.
- Data Origination Access Code—Allow users to indicate a need for a conversion resource on an analog data call origination.

Hardware and Software Requirements

One TN758 Pooled Modem circuit pack is required for each two integrated conversion resources provided. With Version 2, Version 3, or DEFINITY Generic 1, combined conversion resource requires one port on the Digital Line circuit pack and one port on an Analog Line circuit pack.

No additional software is required.

Move Agents From CMS (V3 or G1)

Description

Allows a Call Management System (CMS) user to move agents from one split to another via the screen on the CMS terminal. This feature gives the user of the CMS screen some of the same capabilities that the System Manager has with the System Access Terminal (SAT) or DEFINITY Manager I terminal. The user of the CMS screen can, with a single request, move one agent or multiple agents from the same split to another split.

When the CMS screen is used to move agents from one split to another, all split-associated buttons assigned to an agent are automatically assigned to the new split. Split-associated buttons are those buttons assigned to an agent's voice terminal or console that are associated with a specific split. For example, the After Call Work button is a split-associated button. If an agent is assigned an After Call Work button for one split, and is moved to another split via the CMS screen, the After Call Work button, instead of being associated with the first split, is then associated with the second split. However, if a user already has a specific split-associated button for both the old split (split from which the agent is removed) and the new split (split to which the agent is added), the button assignments remain unchanged. This keeps duplicate split-associated buttons from being assigned to the same split. The following buttons are split-associated buttons:

- Manual-In
- Auto-In
- Auxiliary Work
- After Call Work
- Assist
- Oldest Queued Time (OQT)
- Number of Queued Calls (NQC)
- Incoming Call Identification (ICI).

To move an agent(s) from one split to another split, the CMS user must select the CMS screen that is used to move agents, change the split assignments of the desired agent(s), and press the CHANGE button on the CMS terminal. When the CHANGE button is pressed, the request for the move(s) is sent from the CMS to the switch. The switch then attempts to make the requested move(s) and lets the CMS know whether or not the move(s) was successful or not. The CMS screen then displays a message to indicate whether the move(s) was successful or unsuccessful. If any of the following conditions exist, the move will be unsuccessful:

- The agent or either of the splits (old or new) is not assigned to the system.

- Either of the splits (old or new) is administered as not being measured by CMS.
- ADD, CHANGE, or REMOVE administration (local or remote) is being done on the system.
- The agent is not a membrr of the old split.
- The agent is logged into the old split.
- The agent is already a member of the new split.

Even though the CMS screen can be used to move multiple agents with one request, the system, upon receiving such a request, attempts to make each move individually. Therefore, each agent move is made independently. If, for some reason, one of the requested agents cannot be moved, this does not affect the other move agent requests. The CMS is notified of and displays the success or failure of each agent's move.

Considerations

The Move Agents From CMS feature makes the moving of agents from one split to another fast and simple. Unlike using the SAT (V3) or Manager I terminal (G1), the CMS screen can be used to move multiple agents with a single request.

Up to 32 agents can be moved to another split with a single "move agent" request. Multiple requests must be used to move more than 32 agents.

Only agents that are administered as being "measured by CMS" can be moved using the Move Agents From CMS feature.

Only agents with voice terminal extensions can be moved using the Move Agents From CMS feature. Individual attendants serving as agents cannot be moved using the Move Agents From CMS feature.

Agents cannot be added to or removed from splits via the Move Agents From CMS feature. They can only be moved from one split to another. Additions and removals must be done via the SAT (V3) or Manager I terminal (G1).

System administration (local or remote) that requires use of the ADD, CHANGE, or REMOVE commands cannot be done while the system is making agent moves as a result of a request from the CMS. After the moves are complete, this type of administration can be done.

Interactions

The following features interact with the Move Agents From CMS feature.

- Agent Call Handling

  If an agent is moved from one split to another via the Move Agent From CMS feature, the agent's split-associated buttons are reassigned to the new split. The agent should be provided with a set of button labels so the buttons can be updated accordingly.

Administration

None required.

Hardware and Software Requirements

A CMS adjunct is required. Automatic Call Distribution (ACD) and CMS software is required.

Multi-Appearance Preselection and Preference

Description

Provides multi-appearance voice terminal users with options for placing or answering calls on selected appearances.

- Ringing Appearance Preference

  When a user lifts the handset to answer an incoming call, the system automatically connects the user to the ringing call appearance. If more than one call is incoming, the user is automatically connected to the eldest (first-in) ringing call appearance. The in-use (red) lamp tracks the ringing appearance and the answered appearance.

- Idle Appearance Preference

  When a user lifts the handset to place a call, the system automatically connects the user to an idle appearance even if an incoming call is ringing at another appearance. The in-use (red) lamp tracks an idle appearance when the handset is lifted.

- Preselection

  Before lifting the handset to place or answer a call, the user can manually select an appearance (press a call appearance button or a feature button) where the in-use lamp is dark. Preselection is used, for example, when the user wants to reenter a held call or activate a feature. Preselection also activates the speakerphone if the voice terminal is so equipped.

  The Preselection option overrides both Preference options. If the user does not lift the handset within 5 seconds after using Preselection, the selected appearance returns to idle.

  Preselection can be used with a feature button. For example, if an Abbreviated Dialing button is pressed, a call appearance is automatically selected and, if the user lifts the handset within 5 seconds, the call is automatically placed. Preference only applies if there is a ringing call and if the user lifts the handset. Preference dictates whether the user is connected to the ringing call appearance or to an idle call appearance. If there is no incoming call, the user is automatically connected to an idle call appearance upon lifting the handset. This is true, regardless of the Preference option assigned.

Considerations

Multi-Appearance Preselection and Preference is used to select the call appearances to which users will be connected when they lift the handset.

Multi-appearance voice terminals can have from two to ten call appearances. One of these call appearances is reserved for placing calls or for receiving a Priority Calling call. If a voice terminal has two call appearances and one of them is active, a nonpriority call cannot access the other call appearance, even if the call appearance is idle. Also, the reserved call appearance is not a fixed-position button. It is simply the last idle call appearance. For example, assume a voice terminal has ten call appearances. Any nine can be in use, but the tenth (last) one is reserved. This aspect of system operation should be considered when determining the number of call appearances for a voice terminal. The default value and recommended number of call appearances is 3.

All incoming and outgoing calls require a call appearance. There are no hidden or free call appearances. For example, consider a member of a Call Pickup group with a Call Pickup button. When a call rings some other group member, it can normally be answered by pressing the Call Pickup button. However, pressing the button selects a call appearance for the call, if available. If a call appearance is not available, the call cannot be picked up. Similarly, calls originated using the Facility Busy Indication feature calls also require a call appearance. In this case, the call cannot be completed unless an idle call appearance is available. A Facility Busy Indication button on a called voice terminal provides a visual indication of the busy or idle status of another facility. It does not provide a talking path. These facts should be considered when determining the number of call appearances for a voice terminal.

Interactions

The following features interact with the Multi-Appearance Preselection and Preference feature.

- Call Coverage

  If Cover All Calls (part of the Call Coverage feature) is the redirection criteria to be used for a voice terminal, Idle Appearance Preference should also be assigned to the voice terminal. This allows the principal (called party) to lift the handset without being accidentally connected to a call which should be screened.

- Automatic Incoming Call Display

  Incoming calls are not displayed if Idle Appearance Preference is activated.

Administration

The Idle Appearance Preference option is administered on a per-terminal basis by the System Manager. If Idle Appearance Preference is not administered, the voice terminal will have Ringing Appearance Preference. Both preference options cannot be used on the same voice terminal, and no preference is not an option. Administratively, Idle Appearance Preference (yes or no) is the only choice. No, which is the system default, selects Ringing Appearance Preference. No administration is required for preselection.

Hardware and Software Requirements

No additional hardware or software is required.

Multiple Listed Directory Numbers

Description

Allows a publicly published number for each incoming and two-way (incoming side) foreign exchange (FX) and local central office (CO) trunk group assigned to the system. Also allows up to eight Direct Inward Dialing (DID) numbers to be treated as Listed Directory Numbers (LDNs).

When a CO or FX LDN is called, a trunk group is accessed. The trunk group then routes the call to the incoming destination designated for that trunk group. The incoming destination for an FX or CO trunk group can be one of the following:

- Attendant group
- Automatic Call Distribution (ACD) split
- Direct Department Calling (DDC) group
- Uniform Call Distribution (UCD) group
- Remote Access.

All DID LDN calls route directly to the attendant group.

Considerations

The Multiple Listed Directory Numbers feature provides publicly published numbers for a business. These numbers allow public access to an attendant. LDNs are also useful when it is necessary that the public be able to contact a particular DDC or UCD group. The feature can also be used for Remote Access.

A unique display for incoming call identification can be provided for each LDN, including the DID numbers.

A maximum of 50 Multiple Listed Directory Numbers is allowed per system.

Interactions

If Night Service has been activated and a night console is not assigned or is not operational, incoming LDN calls route as follows:

- DID LDN calls route to a designated DID LDN night extension. If no DID LDN night extension is designated, DID LDN calls route to the attendant.
- Other incoming calls on trunk groups route to the night destination specified for the trunk group. If the night destination is the attendant, calls route to the DID LDN night extension, if specified. If no DID LDN night extension is specified, calls route to the attendant. If no night destination is specified for the trunk group, the calls route to

the normal incoming destination for that trunk group. If that destination is an attendant, calls route to the DID LDN night extension.

- Internal calls and coverage calls to the attendant route to the DID LDN night extension.

Administration

Multiple Listed Directory Numbers is administered by the System Manager. The following items require administration:

- Incoming destination for each CO trunk group and each FX trunk group used for LDNs
- Up to eight DID LDNs
- DID LDN night extension
- A unique name for each LDN (optional, for display purposes).

Hardware and Software Requirements

No additional hardware or software is required.

Music-on-Hold Access

Description

Provides music to a party that is on hold, waiting in a queue, parked, or on a trunk call that is being transferred (V2, V3, or G1). The music lets the waiting party know that the connection is still in effect.

The system provides automatic access to the music source.

Considerations

The music provided by Music-on-Hold Access lets the waiting party know that he or she is still connected. Waiting parties are less likely to hang up. This results in a greater number of completed calls.

If a multiple-party connection is on hold, waiting in queue, or parked, music is not provided.

The number of parties that can be connected to Music-on-Hold Access simultaneously is not limited.

With V2, V3, or G1, transferred trunk calls can be administered to receive either music or silence.

Interactions

When any one of the following features is activated, music is provided when one party is waiting or held:

- Hold
- Conference—Terminal
- Transfer (can be music or silence on trunk calls with V2, V3, or G1).

In addition to these three features, a single party in Call Park can receive music. Also, a call placed in queue for a Direct Department Calling group, Uniform Call Distribution group, or Automatic Call Distribution split can receive a delay announcement followed by music.

If a call with either Data Privacy or Data Restriction activated is placed on hold, Music-on-Hold Access is withheld to prevent the transmission of some musical tone which a connected data service might falsely interpret as a data transmission.

Administration

Music-on-Hold Access is administered on a per-system basis by the System Manager. The only administration required is the assignment of the port number used to provide the feature, and (with V2, V3, or G1) the assignment of whether music or silence is heard on transferred trunk calls.

Hardware and Software Requirements

Requires the music source and one port on a TN763 Auxiliary Trunk circuit pack. Also, if the music source is not FCC registered, a 36A voice coupler is required to provide an interface and system protection for the music source. See *AT&T System 75—Wiring,* 555-200-111, or *DEFINITY Communications System Generic 1—Wiring,* 555-204-111, for a detailed description of Music-on-Hold hardware requirements. No additional software is required.

Names Registration (G1)

Description

Automatically sends a guest's name and room extension from the Property Management System (PMS) to the switch at Check-In, and automatically removes this information at Check-Out. In addition, the guest's call coverage path (for example, a coverage path that terminates at a voice mail adjunct or a hotel operator) will also be sent from the PMS to the Switch during Check-In, and set to the "Default Coverage Path for Client Rooms" at Check-Out.

The information provided by the Names Registration feature may be displayed on any attendant console or display-equipped voice terminal located at various hotel personnel locations (for example, Room Service, Security, etc.). This allows personnel at these locations to provide personalized greetings to calling guests. For example, if John Smith called room service, the restaurant personnel with a display-equipped voice terminal will see John's name and room extension and can answer with a personalized greeting.

Since the updates are sent automatically from the PMS to the switch, the System Manager does not have to manually add guest names into the switch via the Manager I terminal. Normally, in a hotel environment where the daily turnover of guests is large, manual administration of the updates using a Manager I terminal would be a full time administrative task, and would be duplicating the information already resident in the PMS. By linking the automatic updates to the check-in and check-out sequences, the hotel can provide personalized displays more efficiently.

Check-In and Check-Out

During the check-in procedure, information about the guest is obtained and stored in the hotel's Property Management System. At this time, the PMS sends a check-in message to the switch. When the check-in message is sent, the switch removes the outward restriction on the telephone in the guest room, changes the status of the room to occupied, clears any previous wake up calls and message waiting lamp indications, and deactivates Do Not Disturb. Guest Name Registration during check in would add two more operations to those already being performed. These operations would be to update the PBX names internal table and the call coverage path for the guest station. Names Registration enhances the above list of operations by automatically sending a guest's name, extension (room) number, and preferred call coverage path upon check-in. Also, at check-out, Names Registration automatically changes the call coverage path to the administered "Default Coverage Path for Client Rooms".

The Check-in and Check-Out functions are described in the Property Management System feature description elsewhere in this chapter.

Guest Information Input/Change

Guest Information Input/Change allows guest information (name or coverage path) to be entered or altered subsequent to the check-in message. Hotel personnel can change this information at the PMS and it is automatically sent to the switch.

The Guest Information Input/Change function is used when the guest's name associated with an extension must be changed, input of a guest's name must be made after the check-in sequence has taken place, or a change in call coverage arrangement must be made. For example, a hotel may check in airline personnel prior to their arrival at the hotel in order to guarantee their reservation. However, hotel personnel may be unaware of the guests' names until their actual arrival. The names of the airline personnel can be updated using the Guest Information Input/Change function upon actual arrival.

Names Registration Information Format

For both Names Registration and Guest Information Input/Change formats, the guest's name may consist of as many as 15 characters.

The format used by the PMS (last name first, plus first initial and title, etc.) will be sent to the switch and is displayed as it is stored within the switch. All spaces and commas within the name display must also be encoded within the 15 characters. In addition to the 15-character guest name, an extension number (normally up to five digits, but may be up to six digits with prefixed extensions), which corresponds to the guest's room number, will appear on display-equipped voice terminals at hotel service desks.

The guest's name may be in all uppercase letters, all lowercase letters or a mixture of upper and lower case. If a hotel would like to be able to use the Integrated Directory feature (described elsewhere in this chapter), the guest's name must be entered using one of the following methods:

- Last Name, comma, First Name (that is, Jones, Fred)
- Last Name, comma, First Name, space, Title/Middle Initial/Name (that is, Jones,Fred Mr)
- Last Name Only (that is, Jones)
- First Name, space, Middle Name, space, Last Name (will appear as Jones,Fred A).

Only alphanumeric characters, commas, and spaces may be used in the Name field when Integrated Directory is desired. When the feature is not in use, the guest name may be sent to the switch using the above methods. Periods may be used; however, the periods will not be displayed.

Call Coverage

Both Names Registration and Guest Information Input/Change messages contain call coverage path numbers. These numbers are not displayed but are used to configure the appropriate call coverage arrangements for guest phones. Path arrangements for voice mail, text message centers, any available coverage point, or no coverage at all is sent by the PMS for automatic call coverage reconfiguration.

The Call Coverage paths are established at the switch and are then used by the PMS to alter the call coverage arrangement for a guest. If a customized arrangement is desired, the PMS must send a coverage path number (1-600), and manual administration of the specific path can be performed through the Manager I terminal.

Considerations

The guest information provided by the Names Registration feature allows hotel personnel to provide personalized greetings to calling guests. Since guest information updates are sent automatically from the PMS to the switch, the System Manager does not have to manually add guest names into the switch via the Manager I terminal. By linking the automatic updates to the check-in and check-out sequences, the hotel can provide personalized displays more efficiently.

A maximum of 15 characters can be entered as a guest's name on the PMS.

The call coverage path numbers sent by the PMS to the switch for automatic reconfiguration should be limited to those administered in the switch and stored in the PMS.

The guest's room extension number can have a maximum of five digits.

The PMS controls the format of the name displayed on display-equipped voice terminals.

Interactions

The following features interact with the Names Registration feature.

- Call Coverage

  Establishing call coverage arrangements is not limited to the automatic update during check-in messages sent from the PMS. Hotel personnel require alternate coverage points other then those designated for guests. The switch can still be used to manually administer call coverage paths through the Manager I terminal, while automatic updates can still be sent from the PMS for guests' extensions.

- Class of Service (COS)

  If an extension has Client Room COS, the save translation operation clears the station name and sets the coverage path to the Default Coverage Path for Client Room when stored on tape. The existing information is not affected. However, if the translations are read in, existing extensions will be affected until a database swap synchronizes the switch and PMS.

- Property Management System Interface

  During a Room Change/Room Swap, the name originally associated with the first station number is changed/swapped to the second room station along with call coverage path, automatic wake-up entries, message waiting status, and controlled restrictions.

Administration

PMS administration, as described in the Property Management System Interface description elsewhere in this chapter, is required. In addition to this, the items in the following paragraphs should be considered.

To maintain necessary guest security, hotels do not divulge guests' room numbers to other guests or callers. For this reason, display-equipped voice terminals should not be assigned to guests' rooms. A guest with this capability is able to dial another extension and view the guest's name at that extension.

Call Coverage paths must be administered on the switch, and the associated path numbers must be used by the PMS to establish coverage arrangements. If only one coverage arrangement is used by a hotel, this number must be used. For suite rooms, pre-arranged paths can be administered on the switch and the numbers stored within the PMS that will allow one room in the suite to be the coverage point for the other. Special customized arrangements at time of check in (coverage from one guest room to another) are performed by sending the coverage path number from the PMS and then manually administering the attributes of the path at the switch.

Both the PMS and the switch are able to alter guests' names stored in the switch. The last change that is made (by either system) is the change that is used.

The communication protocol used between the switch and the PMS must be administered as "transparent."

The Default Coverage Path for Client Rooms must be administered.

Hardware and Software Requirements

A PMS, if used, can be connected through a Modular Processor Data Module (MPDM) and port on a Digital Line circuit pack or through an Asynchronous Data Unit (ADU) and a port on a Data Line circuit pack. A journal printer can be used and also requires an MPDM and a port on a Digital Line circuit pack or an ADU and a port on a Data Line circuit pack.

Optional Hospitality Services software is required to provide the Property Management System Interface feature.

Network Access—Private

Description

Allows calls to be connected to the following types of networks:

- Common Control Switching Arrangement (CCSA)
- Electronic Tandem Network (ETN)
- Enhanced Private Switched Communications Service (EPSCS)
- Tandem Tie Trunk Network (TTTN).

A private network provides call routing over facilities dedicated to the customer.

Considerations

With Network Access—Private, calls can be made to other switching systems without having to use the public network.

A total of 50 (V1), 60 (V2), or 99 (V3 or G1) trunk groups can be assigned to the system, including private network trunk groups.

Unless prohibited by the Class of Restriction (COR), all incoming Private Network trunks except CCSA can access outgoing trunks without attendant or terminal user assistance. All incoming CCSA calls must route to an attendant or a terminal user.

When off-network calling is specified as part of the CCSA and EPSCS service, long-distance calls route as far as possible over these networks before terminating on the public network. Thus, charges for toll calls are reduced. The COR administered to individual system users determines whether access to this capability is allowed or denied.

Interactions

None.

Administration

Network Access—Private is administered by the System Manager. The following items require administration:

- Tie trunk groups used with private networks.
- Whether or not access to CCSA and/or EPSCS off-network calling is provided. (This assignment is made on a per-COR basis.)

Hardware and Software Requirements

Requires one port on an analog or DS1 Tie Trunk circuit pack for each trunk assigned. No additional software is required.

Network Access—Public

Description

Provides voice terminal users and attendants with access to and from the public network.

Outgoing access is provided to the following:

- Local central offices (COs)
- Foreign exchange (FX) offices
- Wide Area Telecommunications Service (WATS) offices.

Incoming access is provided from the following:

- Local COs
- FX offices
- 800 Service offices.

Considerations

The Automatic Route Selection feature can be used to select the most-preferred route, where possible, for outgoing calls to the public network. Alternatively, trunk access codes can be dialed for manual route selection. Long-distance carrier access codes can be dialed to select particular carriers.

Interactions

None.

Administration

Network Access—Public is administered by the System Manager. All trunk groups used for Network Access—Public must be administered.

Hardware and Software Requirements

Requires one port on a TN747B CO Trunk circuit pack or TN767 DS1 (G1) circuit pack for each trunk assigned. No additional software is required.

Night Service—Hunt Group (V3 or G1)

Description

Hunt Group Night Service allows an attendant or a split supervisor to individually assign a hunt group or split to the night service mode. All calls terminating on the hunt group or split in the night service mode will be redirected to the hunt group/split's designated Night Service Extension (NSE).

Considerations

The Hunt Group Night Service feature gives added flexibility to attendants and designated voice terminal users who are responsible for activating or deactivating individual hunt groups/splits at various times.

The system can have both Hunt Group Night Service and Trunk Group Night Service features at the same time. An incoming trunk call will be redirected to the trunk group's designated NSE. If this NSE happens to be a hunt group/split that happens to be in the Hunt Group Night Service split group, the call will be redirected to the hunt group/split's designated NSE.

Calls in progress, such as talking, on hold, or waiting in queue, on the hunt group/split will not be affected when the hunt group/split is put in the Hunt Group Night Service mode.

Once in the Hunt Group Night Service mode, all calls will be prevented from entering into the hunt group/split queue.

All new calls terminating on the hunt group/split in the Hunt Group Night Service mode will be redirected to its designated NSE.

When the hunt group/split queue becomes empty, all idle members will be put in a busy condition.

If Night Service is activated for a hunt group or split, and a power failure occurs, the hunt group or split will automatically return to the Night Service mode.

Interactions

The following features interact with the Night Service—Hunt Group feature.

- Automatic Call Distribution

  When Hunt Group Night Service is activated for a split and the night-service destination is a hunt group, the caller will hear the first forced announcement, if administered. The call is then redirected to the night service destination hunt group. When an agent in the night service hunt group becomes available, the call goes to that agent. If all agents in the destination hunt group are busy, the caller will hear the following, if assigned: delayed first announcement, music-on-hold or silence, and a second announcement.

- Call Coverage

  While Night Service is activated, the NSE's normal coverage criteria and path will apply to night service attempting to terminate at that NSE. If the NSE is a hunt group/split of any type, the hunt group/split's call coverage criteria and coverage path apply. The hunt group/split's coverage criteria and path can be different from that assigned to the voice terminals that are members of that hunt group/split.

  If a coverage point is a hunt group/split in night service, it is considered unavailable and the call will not be forwarded to the coverage point's NSE.

- Call Forwarding—All Calls

  If the hunt group/split is in the Hunt Group Night Service mode and the hunt group/split's NSE has Call Forwarding—All Calls activated, the night service calls terminating to that NSE will be forwarded to its designated extension.

  If the forwarded-to destination is a hunt group/split in the Night Service mode, the call will not be forwarded and will be terminated at the forwarding extension.

Administration

Hunt Group Night Service is administered on a voice terminal basis or attendant console. The following items require administration:

- Assign "hunt-ns" button(s) to designated voice terminal(s). Up to three hunt group buttons can be assigned to a combination of attendant consoles and voice terminals in each hunt group. The hunt group number must be assigned for each button. These buttons should be assigned to feature buttons that have an associated status lamp. The lamp lights when Hunt Group Night Service is activated. If the assigned button has no status lamp, no visual indication of the Hunt Group Night Service status is given.

- Assign "hunt-ns" button(s) to attendant console(s). Up to three buttons can be assigned to a combination of voice terminals and attendant consoles assigned for each hunt group. The hunt group number must be assigned for each button.

Hardware and Software Requirements

No additional hardware or software is required.

Night Service—Night Console Service

Description

Directs all calls for the primary and daytime attendant consoles to a night console.

Night Service—Night Console Service is activated when an attendant presses the Night button on the primary attendant console. Night Service is deactivated by pressing the Night button again. When Night Service is activated, all attendant-seeking calls and calls waiting in queue are directed to the night console.

Considerations

Night Service—Night Console Service calls to the attendant group are still handled by an attendant, even though the primary and daytime attendant consoles are out of service.

Only one night console is allowed in the system. The night console can be activated only when the primary and daytime consoles have been deactivated. The attendant activates the night console and deactivates all other consoles by pressing the Night button on the primary console.

The night console must be identical to, and have the same features as, the primary console. A daytime console can double as the night console.

With V1, if Night Service is activated and a power failure occurs, the system, when brought back up, will not automatically return to the Night Service mode. Instead, the system returns to the day mode. With V2, V3, or G1, if Night Service is activated and a power failure occurs, the system, when brought back up, automatically returns to the Night Service mode.

Interactions

Activation of Night Service for the attendant consoles also puts trunk groups into night service, except those trunk groups for which a night service button is administered. See the Night Service—Trunk Group feature description for details.

Administration

Night Service—Night Console Service is administered by the System Manager. The only administration required is the assignment of a night console and whether or not only Direct Inward Dialing (DID) Listed Directory Number (LDN) calls will go to the DID LDN night service extension.

Hardware and Software Requirements

Requires an attendant console. No additional software is required.

Night Service—Night Station Service

Description

Redirects incoming attendant-seeking trunk calls to designated extension numbers whenever the system is placed in Night Service.

This feature is activated under the following two conditions:

- The attendant (or voice terminal user, if the switch has no attendant) has pressed the Night button on the primary console.
- A night console is not assigned or not operational.

When the above conditions have been met, incoming calls to the attendant route as follows:

- Direct Inward Dialing (DID) Listed Directory Number (LDN) calls route to a designated DID LDN night extension.
- Internal calls to the attendant route to the DID LDN night extension.
- Incoming calls on trunk groups (other than DID trunk groups) which have the attendant as their destination route to the night destination specified for the trunk group or individual trunk (V2, V3, or G1). If no night destination is specified, the calls route to the DID LDN night extension.

When Night Station Service is activated, all trunk and internal calls to the attendant (other than calls redirected via Call Coverage or Call Forwarding All Calls) route to either the DID LDN night extension, the trunk group's specified night destination, or the individual trunk's specified night destination (V2, V3, or G1) as described above. A different extension number can be assigned as the night destination for each incoming central office, foreign exchange, or 800 Service trrunk group. Both the DID LDN night extension and the extension number assigned as a trunk group's night destination can be a voice terminal or an answering group, that is, Direct Department Calling group, Uniform Call Distribution group, or Terminating Extension Group.

Calls redirected to the attendant via Call Coverage or Call Forwarding All Calls do not route to the DID LDN night extension. These calls enter the attendant queue, and can be answered via the Trunk Answer From Any Station feature, if administered.

Considerations

Night Station Service provides for the answering of attendant-seeking calls when all attendant consoles are out of service due to Night Service activation.

When the Night Station Service feature is active but night station extension numbers have not been established, the Trunk Answer From Any Station feature can be activated.

With V2, V3, or G1, a Night-Serv button can be assigned to either an attendant or a voice terminal extension. This button, when pressed, puts the entire system in night service and incoming calls on all trunk groups (except DID LDN) route to the night destination specified for the trunk group.

With V3 or G1, an individual trunk group or hunt group can be put into night service by either an attendant or a voice terminal extension with the required button (Trunk Night Service or Hunt Night Service). When the button is pressed, all calls to that particular trunk group or hunt group are routed to the night service extension assigned to that group. A second depression of the same button deactivates night service for that trunk group or hunt group.

With V1, if Night Service is activated and a power failure occurs, the system, when brought back up, will not automatically return to the Night Service mode. Instead, the system returns to the day mode. With V2, V3, or G1, if Night Service is activated and a power failure occurs, the system, when brought back up, automatically returns to the Night Service mode.

If Night Service is activated and the DID LDN night extension is busy, an incoming DID LDN call receives busy tone (V1 or V2) or may be forwarded to another number (V3 and G1).

With V1, if Night Service is activated, and a call then returns to the console (for example, an incoming call transferred by the attendant that has not been answered and has timed out), the call will be dropped. It will not route to the DID-LDN night extension.

Interactions

The following features interact with the Night Service—Night Station Service feature.

- Call Coverage

  A call routed to the DID LDN night extension via Night Station Service does not go to coverage, even if the coverage criteria of the DID LDN night extension is met.

  Calls redirected to the attendant via Call Coverage do not route to the DID LDN extension.

  If a night extension has a coverage path in which Cover All Calls has been administered, all attendant-seeking calls will redirect to coverage and changes to the protocol for handling DID LDN calls (that is, forwarding attendant-seeking calls on- or off-premises from the night extension) will not work.

- Call Forwarding All Calls (V1)

  A call routed to the DID LDN night extension via Night Station Service does not forward to another extension, even if Call Forwarding All Calls has been activated at the DID LDN night extension.

  Calls redirected to the attendant via Call Forwarding All Calls do not route to the DID LDN extension.

- Inward Restriction

  Inward-restricted voice terminals can be administered for Night Station Service. Night Service features override Inward Restriction.

- Night Service—Trunk Answer From Any Station

  Night Service—Trunk Answer From Any Station and Night Service—Night Station Service can both be assigned within the same system, but cannot be assigned to the same trunk group.

- Remote Access

  The Remote Access extension number can be specified as the Night Station extension number on an incoming, non-DID, trunk group.

- Timed Reminder

  Timed Reminder Calls returning to a console which has been placed in Night Service and has an assigned DID LDN night extension will not be redirected to the DID LDN night extension, but will be dropped.

Administration

Night Station Service is assigned by the System Manager. The following items require administration:

- DID LDN night extension and permission to let DID LDN calls redirect to the DID LDN night extension.
- Trunk group night destination (per trunk group)
- Hunt group night destination (per hunt group) (V3 or G1)
- Night-Serv button
- Hunt Night Service button
- Trunk Night Service button.

Hardware and Software Requirements

No additional hardware or software is required.

Night Service—Trunk Answer From Any Station

Description

Allows voice terminal users to answer all incoming attendant-seeking calls when the attendant(s) is not on duty and when other voice terminals have not been designated to answer the calls.

The incoming call activates a gong, bell, or chime. A voice terminal user dials an access code and answers the call.

Trunk Answer From Any Station (TAAS) is activated only under the following three conditions:

- The attendant has pressed the Night button on the primary console.
- A night console is not assigned or not operational.
- The Night Station Service feature is not active.

Considerations

When Trunk Answer From Any Station is activated, any user can answer the attendant-seeking trunk call. Even though an attendant is not available, the call is still answered. This reduces the number of lost calls.

With V1, if Night Service is activated and a power failure occurs, the system, when brought back up, will not automatically return to the Night Service mode. Instead, the system returns to the day mode. With V2, V3, or G1, if Night Service is activated and a power failure occurs, the system, when brought back up, automatically returns to the Night Service mode.

Interactions

Inward-restricted voice terminals can activate TAAS for incoming trunk calls. Night Service features override Inward Restriction.

Calls that are redirected to the attendant via the Call Coverage and Call Forwarding All Calls features while the Night Station Service feature is activated can be answered via TAAS.

Night Service—Trunk Answer From Any Station and Night Service—Night Station Service can both be assigned within the same system, but cannot be assigned to the same trunk group.

Administration

Trunk Answer From Any Station (TAAS) is administered on a per-system basis by the System Manager. The following items require administration:

- Dial access code for TAAS (to answer a call)
- Port for the ringing device.

Hardware and Software Requirements

Requires a ringing device and one port on a TN742, TN746, or TN769 Analog Line circuit pack. No additional software is required.

Night Service—Trunk Group (V3 or G1)

Description

The Trunk Group Night Service feature allows an attendant or a designated voice terminal user to individually assign a trunk group or all trunk groups to the night service mode. Specific trunk groups (individually) assigned to Trunk Group Night Service are in the "Individual Trunk Night Service Mode." In this mode, incoming calls made on a specific trunk group will be redirected to its designated Night Service Extension (NSE). Incoming calls on the trunk groups not assigned to Trunk Group Night Service will be processed normally. The specific trunk groups can be assigned to Trunk Group Night Service by pressing the individual Trunk Night Service button(s) on the attendant console or a voice terminal.

All trunk groups can be assigned to the night service mode at the same time. In this arrangement, the trunk groups are in the System Night Service mode. Any incoming calls made on the trunk groups will be redirected to their designated NSE. All trunk groups can be assigned to System Night Service by pressing the System Night Service button on the principal attendant console or a designated voice terminal.

Considerations

The Trunk Group Night Service feature gives added flexibility to attendants and designated voice terminal users who are responsible for activating or deactivating all, or individual, trunk groups at various times.

All incoming calls on individual or system Night Service trunk groups will go to the trunk group's NSE unless the trunk group member has its own Trunk Group Member Night Destination, in which case the call will be redirected to that night destination instead of the trunk group's NSE.

Calls already in progress on a trunk group, such as talking, on hold, or waiting in queue on a trunk group, are not affected when the individual Trunk Group Night Service or System Night Service feature is activated by the attendant or a voice terminal user.

Trunk Group Night Service and System Night Service both work independently of each other. Activation or deactivation of one of these night service features does not affect the other. Specific situations are described below:

- When System Night Service is deactivated, trunks with individual Trunk Group Night Service still activated remain in night service.
- When System Night Service is activated, trunks controlled by individual Trunk Group Night Service buttons remain in day service.
- Trunks with individual Trunk Group Night Service can be taken out of Night Service even though the rest of the system remains in Night Service.
- Trunks with individual Trunk Group Night Service can be put into Night Service even though the rest of the system remains in day service.

- Trunk groups assigned to individual Trunk Group Night Service will not be reassigned to System Night Service when the System Night Service feature is activated. Those trunk groups that are not currently assigned to Trunk Group Night Service will be assigned to System Night Service.

If a trunk is added to a trunk group while that trunk group is in Trunk Group Night Service, the trunk is brought up in night service.

Individual Trunk Group Night Service does not apply to Direct Inward Dialing trunk groups.

If Night Service is activated for a trunk group, and a power failure occurs, the trunk group will automatically return to the Night Service mode.

If, for some reason, a voice terminal with a trunk-ns button remains out of service after a system reboot and later comes back in service, the trunk-ns lamp will show the trunk status within 10 seconds of coming back in service. For example, a voice terminal with a trunk-ns button may be unplugged when the system is rebooted. If the voice terminal is later plugged back in, the trunk status will be shown on the trunk-ns button within 10 seconds.

Interactions

The following features interact with the Night Service—Trunk Group feature.

- Listed Directory Number

  In the System Night Service mode, all incoming Listed Directory Number (LDN) calls, except those using Direct Inward Dialing (DID) trunks, which have activated night service will be redirected to their corresponding trunk group's NSE. Incoming LDN calls using DID trunks are directed to the Night Console Service, Night Station Service, or Trunk Answer From Any Station feature, respectively, whichever applies first. Non-LDN DID trunk calls terminate at the dialed extension.

- Call Forwarding—All Calls

  If the Trunk Group Night Service mode and the trunk group's NSE has Call Forwarding—All Calls activated, the night service calls terminating to that NSE will be forwarded to its designated extension.

Administration

Individual Trunk Group Night Service is administered on a voice terminal basis or attendant console. The following items require administration:

- Assign "trunk-ns" button(s) to designated voice terminal(s). Up to three buttons can be assigned to voice terminals in each trunk group. The trunk group number must be assigned for each button.

If a trunk-ns button is assigned for an existing trunk, it is updated immediately to show the status of the trunk.

- Assign "trunk-ns" button(s) to attendant console(s). Three buttons per attendant console are allowed. The trunk group number must be assigned for each button.

- Permission to let DID LDN calls redirect to the DID LDN night extension.

The system can have Trunk Group Night Service and split Night Service at the same time, but the call will be redirected to the trunk group's NSE before it goes to the hunt group/split's NSE.

Hardware and Software Requirements

No additional hardware or software is required.

Off-Premises Station

Description

Allows a voice terminal located outside the building where the switch is located to be connected to the system. If central office (CO) trunks are used, the voice terminal must be analog and must be FCC-registered.

Considerations

Off-Premises Stations are useful whenever it is necessary to have a voice terminal located away from the main location.

The maximum loop distance for Off-Premises Stations is 14,000 feet, without repeaters.

Interactions

The Distinctive Ringing feature might function improperly at an Off-Premises Station due to the distance. However, the Distinctive Ringing feature can be disabled when the Off-Premises Station is administered. If the Distinctive Ringing feature is not used with an Off-Premises Station, the terminal will receive 1-burst ringing for all calls.

Administration

Off-Premises Stations are administered by the System Manager.

Off-Premises Stations are administered the same as on-premises voice terminals with the following exceptions:

- For voice terminals used as Off-Premises Stations, the Off-Premises Station field must be administered as "yes".
- For voice terminals used as Off-Premises Stations, the R Balance Network field must be completed.

Hardware and Software Requirements

Requires cross-connecting capabilities and one port on a TN742 or TN769 Analog Line circuit pack or one port on a TN767 DS1 Tie Trunk circuit pack for each interface to be provided. No additional software is required.

Outbound Call Management (G1)

Description

Outbound Call Management (OCM) provides automated dialing of outgoing calls, connection of these calls to an agent when the calls are answered, and screening of these calls when they are not answered. OCM is used with the Automatic Call Distribution (ACD) feature to process large numbers of outgoing calls for a business such as a telemarketing center.

ACD splits and agents are used to perform the OCM functions. Please note that the OCM splits and OCM agents described in this document are actually ACD splits and agents being used for OCM purposes. In contrast to ACD, which is primarily used to process incoming calls, OCM is used to process large numbers of outgoing calls.

OCM uses the DEFINITY Communications System Generic 1 switch along with a separate adjunct processor to improve the efficiency of agents by automating outgoing calls. OCM relies heavily on the ACD and Agent Call Handling features and procedures to process the outgoing calls. With OCM, an agent does not have to manually make outgoing calls. The system, with the help of the adjunct processor, automatically makes the outgoing calls and delivers the call to an agent only after the call has been answered. This way, the agent does not waste time waiting for a call to be answered.

OCM Configuration

OCM is configured as shown in Figure 3-26. All agents in an OCM split must have both a voice terminal and a data terminal. Agents' voice terminals are connected to the switch and agents' data terminals are connected to both the adjunct and the synchronous host via a cluster controller as shown in Figure 3-26. The DEFINITY Generic 1 is connected to the adjunct (3B2) by a synchronous BX.25 OCM communications link. Asynchronous data connections may be direct or switched through the DEFINITY Generic 1.

Adjunct Functions

The main functions of the Adjunct in an OCM configuration are as follows:

- Managing calling lists
- Interfacing customer's host computer
- Initiating and pacing the dialing of the outgoing calls
- Managing the agent/terminal interface
- Generating OCM reports.

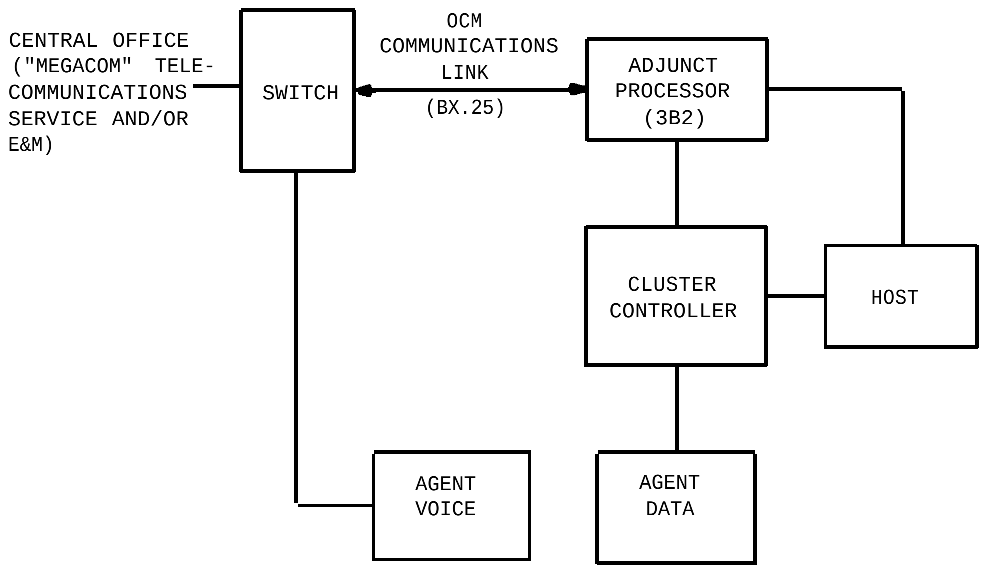


Figure 3-26. Outbound Call Management Configuration With Cluster Controller and Host

DEFINITY Generic 1 Functions

The main functions of the DEFINITY Generic 1 in an OCM configuration are as follows:

- Setting up calls
- Doing call classification
- Providing call status feedback to the adjunct
- Providing support features.

Optionally, an OCM system may also include other supporting adjuncts such as a Call Management System (CMS) and/or an Operations Support System (OPS).

OCM Operations

Agent Log-In

Before an agent can process OCM calls, the agent must be assigned to an OCM split (also called an "adjunct controlled split") and be logged into that split. The agent logs into the OCM split using the data terminal keyboard to enter the split extension and login information. The adjunct then sends the login request over the OCM communications link to the switch. The login is successful if:

- The requested split extension is valid and the split is an OCM split.

- The agent has been administered as a member of the split.
- The agent is off-hook/idle in the auto-answer mode.
- The correct number of login identification digits has been entered.

If the login is successful, the agent is now available to receive OCM calls. When an agent logs into an OCM split, the system locks the touch-tone pad on the agent's voice terminal. Therefore, the data terminal is used to do the following:

- Log in and log out of the OCM split. This can only be done on a data terminal.
- Change work modes. The same work modes that apply to ACD agents also apply to OCM agents.
- Make a call.
- Drop a call.
- Transfer a call.

Once an agent has logged into an OCM split, that agent cannot log into another split until he or she logs out of the OCM split.

What the Switch and Adjunct Do After an Agent Logs Into an OCM Split

After an agent logs into a split, the adjunct can initiate OCM calls in either of two ways. The adjunct can request OCM calls from a call list, or can request an OCM call as a result of a request from an agent's data terminal. The switch receives call requests from the adjunct over the OCM communications link. The switch then interprets the requests, responds back to the adjunct over the same link, and makes the call.

When the switch receives the request and makes the OCM call, the switch and adjunct perform as follows:

- If the request is from a call list, one of the following will occur:
  - If the called party answers, the adjunct is informed of the agent selected to handle the call and the call is transferred to an available agent by the switch. As soon as the adjunct knows which agent is selected, the adjunct also delivers data associated with the call to the agent's data terminal.
  - If the called party is either busy or does not answer after a predetermined number of rings, the switch will drop the call and notify the adjunct. The adjunct will then try the call again at a later time.
- If the request is from an agent, the following will occur:
  - A call is originated for that agent. Call progress tones are provided to the agent whether the call is answered or not. The agent can then drop the call if the called party does not answer or when the conversion is completed. Call Classification by the switch is not done automatically in this case.

During an active call, an agent may transfer the call to another OCM agent or non-OCM extension (such as a hunt group, Automatic Call Distribution (ACD) split, announcement, voice terminal extension, or attendant). Also, the agent may conference the call or consult with a supervisor using the data terminal keyboard. When the call is finished, either the agent or the called party may disconnect from the call. The agent may then enter the After Call Work mode to do additional work before being available to answer another call. (The After Call Work [ACW] mode is described later in more detail.)

Agent Answering Options

An agent can answer OCM calls by using either a headset, handset, or speakerphone.

The Automatic Answer option allows the agent to be connected directly to incoming calls without ringing. The agent hears zip tone through the headset, handset, or speakerphone and is automatically connected to the OCM call when the called party answers.

It is recommended that Automatic Answer be used with a headset. In this case, the agent hears zip tone through the headset and is then automatically connected to the call. (If the incoming trunk group is data restricted, the zip tone is not heard. If the agent's extension is data restricted, all parties on the call hear the zip tone. A headset user should not be assigned data restriction.)

Although possible, it is not recommended that a handset or speakerphone be used with Automatic Answer. For an agent with Automatic Answer and a handset or speakerphone to answer an OCM call, the handset or speakerphone must be off-hook (handset lifted or speakerphone turned on). While off-hook, the agent hears zip tone through the handset or speakerphone.

OCM Agent Call Work Modes

An agent can enter either of the three work modes described below:

- Auxiliary Work—An agent enters the Auxiliary Work mode to do non-OCM activities such as taking a break or going to lunch. This makes the agent unavailable for OCM calls. The agent remains in this work mode until the adjunct makes a request for a change in work mode for that agent.

- After Call Work—An agent enters the ACW mode to perform OCM-related activities. For example, an agent may need to fill out a form as a result of an OCM call. The agent is unavailable for OCM calls while in the ACW mode. An agent enters the ACW mode automatically upon disconnecting from an OCM call while in the Manual-In mode. Also, an agent may enter this mode as a result of a specific request from the adjunct. The agent remains in this work mode until the adjunct makes a request for a change in work mode for that agent.

- Manual-In—When an agent is in the Manual-In mode, he or she, upon disconnecting from an OCM call, is automatically entered into the ACW mode by the switch, and is not available for OCM calls. For this mode change to happen, the agent must be assigned the Automatic Answer option (described previously), logged into the OCM split, off-hook, and idle. The agent remains in the ACW mode until the adjunct makes a request for a change in work mode for that agent.

An agent can enter one of the previously described work modes as a direct result of a request by the OCM adjunct or as a result of the OCM call status. If an agent is given the capability, that agent can request a work mode change via the data terminal keyboard. The adjunct may make a request for a work mode change at any time. If the agent is not active on a call, the change is accepted immediately. If the agent is active on a call, the change is denied.

Voice Terminal Operations

When an agent logs into an OCM split, the system locks the touch-tone pad on the agent's voice terminal. Therefore, the agent's voice terminal is restricted from normal use. Going on-hook drops the current call and logs the agent out of the OCM split. Incoming non-OCM calls to an OCM agent that is logged into an OCM split are either redirected to coverage or receive busy treatment. Any voice terminal lamps (if provided) continue to operate normally.

An agent can place outgoing calls only through the OCM adjunct via the data terminal keyboard. Also, the adjunct may initiate an outgoing call to be connected to the agent after the called party answers.

OCM Split Queuing and Announcements

A small queue should be assigned to all OCM splits. OCM calls have already been answered by the called party before they are directed to an OCM split. OCM calls and regular ACD calls should never be directed to the same split. OCM split queues should be carefully considered and monitored. Also, they should not have a long queue length.

Announcements can be administered so that if an OCM split agent is not available for an OCM call, the call can be queued and the called party can receive an announcement. Announcements are strongly recommended in OCM applications when queuing is desired, because without them the called party may hear ringback tone and hang up. The called party should be connected to either an agent or an announcement immediately. Forced first announcements are not recommended for applications using OCM agents.

If an OCM call is directed to a specific OCM agent rather than an OCM split, and that agent is busy, the call cannot be queued. Therefore, the called party cannot be connected to an announcement. If this happens, the switch notifies the adjunct and drops the call.

Announcements can also be assigned so that the called party on an OCM call is automatically connected to an announcement when he or she answers the call (an agent is not needed in this case). When the announcement is over, the switch drops the call. This can be useful for applications such as promoting a sale or notifying customers of other upcoming events.

Two announcements can be assigned to each split. The second announcement can be administered so that it will repeat itself.

When an OCM call is directed to an OCM split, the call, depending on the administration of the split, will either try to access a split agent or will automatically be connected to the first announcement (Forced First Announcement), if assigned.

Forced First Announcement: The first announcement delay interval (0 to 99 seconds) indicates how long a call will remain in queue before the call is connected to the first announcement. If this interval is set to "0" seconds, the incoming call will automatically be connected to the first announcement, if available. The result is a "forced first announcement," and the call will not attempt to access an agent until after the first announcement is heard. Forced first announcements, although available and useful for ACD applications, are not recommended for applications using OCM agents.

When a forced first announcement is assigned, the system tries to connect the called party to the first announcement, with the results being one of the following:

- If the first announcement is available, the called party receives audible ringing followed by the first announcement. The system then tries to connect the call to an agent.
- If the announcement is busy and has no queue, the system will wait 10 seconds and then try to access the announcement again.
- If the announcement is busy and has a queue, one of the following happens:
  - If the queue is full, the system will wait 10 seconds and then try to access the announcement again.
  - If the queue is not full, the call enters the announcement queue and the called party receives audible ringing until the first announcement is heard. The system then tries to connect the call to an agent.
- If the announcement is not busy, but is still unavailable (it might have been deleted), then the system tries to connect the call to an agent.

Entering the Queue: When a forced first announcement is not assigned, the system will try to connect an incoming call to an available agent. If an agent is available, the call is connected to the agent. If all agents in the split are active, the call enters the split queue.

First Announcement: After a call enters a split queue, the called party receives audible ringing and the first announcement delay interval begins. (If there is no first announcement, the second announcement delay interval begins. If there is no second announcement, the call remains in queue until answered or removed from the queue.) If an agent becomes available during the first announcement delay interval, the call is connected to the available agent. Otherwise, the first announcement delay interval expires and the system tries to connect the called party to the first announcement, with the result being one of the following:

- If the first announcement is available, the called party receives audible ringing followed by the first announcement.
- If the announcement is busy and has no queue, the called party receives audible ringing and the first announcement delay interval is reset. The system will try to access the announcement again when the interval expires.

- If the announcement is busy and has a queue, one of the following happens:
  - If the queue is full, the called party receives audible ringing and the first announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
  - If the queue is not full, the call enters the announcement queue and the called party receives audible ringing until the first announcement is heard. The system then tries to connect the call to an agent.
- If the announcement is not busy, but is still unavailable (it might have been deleted), the second announcement delay interval begins and the system attempts to connect the call to the second announcement. If there is no second announcement, the call will remain in queue until answered or removed from the queue.

Second Announcement: After the first announcement has completed, the second announcement delay interval begins and the called party hears music, if provided. (If there is no second announcement, the call remains in queue until answered or removed from the queue.) If an agent becomes available during the second announcement delay interval, the call is connected to the available agent. Otherwise, the second announcement delay interval expires and the system tries to connect the incoming call to the second announcement, with the result being one of the following:

- If the second announcement is available, the called party receives audible ringing (only if the first announcement has not been heard) followed by the second announcement.
- If the announcement is busy and has no queue, the called party receives audible ringing and the second announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
- If the announcement is busy and has a queue, one of the following happens:
  - If the queue is full, the called party receives audible ringing (only if the first announcement has not been heard) and the second announcement delay interval is reset. The system will try to access the announcement again when the interval expires.
  - If the queue is not full, the call enters the announcement queue and the called party receives audible ringing (only if the first announcement has not been heard) until the second announcement is heard. The system then tries to connect the call to an agent.
- If the announcement is not busy, but is still unavailable (it might have been deleted), the call will remain in queue until answered or removed from the queue.

After the second announcement is heard, the called party hears music (if provided) or silence (if music is not provided), and one of the following occurs:

- If the split has been administered so that the second announcement is repeated, the system will attempt to connect the call to the second announcement after the delay expires.

- If the split has been administered so that the second announcement is not repeated, the call will remain in queue until answered or removed from the queue.

Forced Disconnect: At times, it may be desired to connect the called party directly to an announcement and then disconnect the call after the announcement has completed. This can be accomplished two ways:

- The incoming destination can be administered as an announcement extension. This way the calling party will hear the announcement and be disconnected. Also, the call is never queued for a split because it goes directly to the announcement.

- An announcement extension can be administered as a point in a split's coverage path. This way, calls that have been in the queue for a long period of time are forced to go directly to the announcement and are then disconnected.

Queue Status Indications

The system provides queue status indications for OCM calls based on the number of calls in queue and time in queue. These indications are provided via lamps assigned to the terminals or consoles of split agents or supervisors. In addition, an auxiliary warning lamp can be provided to track queue status based on time in queue and another for number of calls in queue. Also, display-equipped voice terminals and consoles can display the time in queue of a split's oldest call and the number of calls in that split's queue.

Two types of Queue Status Indications are provided:

- Number of Queued Calls

  The Number of Queued Calls status indication is based on the total number of calls in queue at a split. The status indication can be provided by an NQC (Number of Queued Calls) button with associated lamp on a voice terminal or console. Each split is assigned a Number of Queued Calls warning threshold of 1 to 99 calls. When this threshold is reached, the lamp associated with the NQC button flashes. If there are calls in the queue, but the threshold is not reached, the lamp lights steadily. If there are no calls in queue, the lamp goes dark.

  In addition to the NQC button(s), the Number of Queued Calls status indication can be provided by an auxiliary queue warning lamp. This lamp can be installed at any location convenient to the split agents. When the Number of Queued Calls warning threshold is reached, the auxiliary queue warning lamp lights.

- Oldest Queued Time

  The Oldest Queued Time status indication is based on the time in queue of the oldest call in a split queue. The status indication can be provided by an OQT (Oldest Queued Time) button with associated lamp on a voice terminal or console. Each split is assigned an Oldest Queued Time warning threshold of 0 to 999 seconds. When the oldest call in queue has been in queue for this length of time, the lamp associated with the OQT button flashes. If there are calls in the queue, but the threshold is not reached, the lamp lights steadily. If there are no calls in queue, the lamp goes dark.

  In addition to the OQT button(s), the Oldest Queued Time status indication can be provided by an auxiliary queue warning lamp. This lamp can be installed at any location convenient to the split agents. When the Oldest Queued Time warning threshold is reached, the auxiliary queue warning lamp lights.

Each NQC and OQT button is associated with a specific split. Display-equipped voice terminals and consoles can display queue status information for a split by pressing the OQT or NQC button. The same information is displayed no matter which of the two buttons is pressed. The split name (or extension if name is not assigned), Oldest Queued Time, and Number of Queued Calls are displayed for 5 seconds unless the displaying terminal or console receives an incoming call or the display is put into another mode. Otherwise, at the end of 5 seconds, the display returns to its previous condition. If the display has two lines, the queue status information is displayed on the second line.

Call Distribution Within an OCM Split

When the OCM adjunct initiates a call and the called party answers, the adjunct must then decide which member of the OCM split should receive the call. Calls are distributed to members of an OCM split by either the Direct Department Calling (DDC) feature or Uniform Call Distribution (UCD) feature. One of these methods of call distribution, described as follows, may be assigned to each split.

- Direct Department Calling

  If a split is administered for DDC, an incoming call rings the first available extension number in the administered sequence. If the first split agent in the sequence is active on a call (busy), or is not available because of activity in one of the work modes, the call routes to the next split agent and so on. In this case, incoming calls always try to complete at the first split agent in the administered sequence. Therefore, the calls are not evenly distributed among the split agents.

- Uniform Call Distribution

  If a split is administered for UCD, an incoming call will ring the available split agent that has not received an OCM split call for the longest period of time (the most-idle agent). In this case, incoming calls to a split will be distributed evenly among the split agents. For this reason, UCD hunting is usually preferred over DDC hunting.

Agent Request for Supervisor Assistance

An OCM agent can make a request for supervisor assistance on a call. This function is similar to an ACD Agent Request for Supervisor Assistance, except with OCM the agent makes the request using the data terminal keyboard.

Call Classification

Call Classification is used to place an OCM call in a specific category depending on the outcome of the call. Call Classification is done automatically by the switch or manually by the OCM agent (depending on whether the OCM call is the result of a request from the OCM call list or the result of a request from an agent).

The Call Classification categories are listed below:

- ***Busy—*** A call is classified as Busy when a busy tone is returned to the switch. If the call is classified as busy because the called party is busy, the call is dropped and the "Far End Busy" condition is reported to the adjunct. If the call is classified as busy because the agent or split is busy, the call is dropped and the Reorder condition is reported to the adjunct.

- ***No Answer—*** A call is classified as No Answer when the called party's phone rings for a specified time interval or number of rings without being answered. If the call is classified as No Answer, the call is dropped and the No Answer condition is reported to the adjunct.

- ***Answered—*** A call is classified as Answered when the called party goes off-hook (answers the call). If a call is classified as Answered, the Answered condition is reported to the adjunct. The adjunct then initiates the transfer of the call to the specified split or extension.

- ***Reorder—*** A call is classified as Reorder when the system detects a reorder tone on the call. If a call is classified as Reorder, the call is dropped and the Reorder condition is reported to the adjunct.

Other call conditions include the following:

- ***High and Dry—*** A call is considered High and Dry when no tones are detected after the switch has completed dialing. After a predetermined time-out interval, the call is considered Answered and is transferred to an agent for verification. The agent then classifies the call and notifies the adjunct of the classification via the data terminal keyboard.

  The time-out interval mentioned in the previous paragraph is the same as that used for off-premises data calls. The normal default is 6 seconds. For the OCM application, however, it is recommended that the time-out be set to at least 15 seconds for increased accuracy in detecting High and Dry calls.

- ***Special Information Tones—*** Calls that receive Special Information Tones from the network (for such things as numbers that do not exist, or network congestion) are considered Answered and are transferred to the agent for verification and classification.

OCM Call Disconnecting

An OCM agent should disconnect from an OCM call promptly after the call is completed, rather than relying on the called party to disconnect from the call. This keeps call facilities from being tied up longer than necessary. An OCM agent can disconnect from a call in either of the following ways:

- By pressing a function key on the data terminal keyboard—This can be done only if the OCM adjunct initializes the call, and is the equivalent of pressing the Release button on a voice terminal. The agent is dropped from the current call. The call is then torn down, unless there is more than one party left on the call. In this case, the system saves the call. After disconnecting from the call, the agent is placed in the After Call Work (ACW) mode.

- By going on-hook (hanging up) or turning the headset off—These methods are not recommended except in drastic situations, because they will log the agent out of the split if the agent has a multi-appearance voice terminal. Single-line voice terminals are not logged out by going on-hook.

Logging Out

An agent is normally logged out of an OCM split via the OCM adjunct. The OCM adjunct notifies the switch that the agent is no longer available for OCM calls. The switch then logs the agent out of the split, unless the agent is still active on a call. In this case, the call is not affected, the agent is still considered logged out (the agent will not receive more OCM calls), and full functionality is returned to the agent's voice terminal.

An agent is automatically logged out of an OCM split when he or she turns the headset off, when the OCM Communications Link goes down, or when an agent's extension is placed in an out-of-service state because of maintenance. If an agent is logged out of a split because of a failure in the OCM Communications Link, the adjunct will automatically try to log the agent back in when the link is reestablished.

Considerations

The OCM feature provides automated dialing of outgoing calls, delivery of these calls along with data screen information to agents when the calls are answered, and statistics on the calls. The automated dialing reduces dialing errors. Since only answered calls are delivered to agents, their productivity is increased by eliminating the time spent listening to call progress tones. Data screen information allows the agent to address the correct party and have information on the call at hand. The call statistics are important for proper management of the operation.

In a stand-alone OCM system (the switch and adjunct are used only to perform OCM operations), an agent who performs only outgoing OCM functions needs only a basic voice terminal. If the system is a shared system (performs operations other than OCM), the same agent may perform both OCM and ACD functions, although not at the same time. In this case, the agent should have adequate voice terminal equipment to perform ACD operations and data terminal to perform OCM functions.

An agent can be assigned to more than one OCM split if desired, but an agent can only log into one OCM split at a time. If the OCM Communications Link is down, an agent cannot log into an OCM split, but can log into an ACD split.

If an agent is administered as a member of an OCM split, he or she must log into that split via the data terminal keyboard. The number of digits in the login identification number must be equal to the systemwide maximum.

The maximum number of OCM agents is limited by the total number of calls that can be placed during a busy hour. The number of calls per busy hour depends on the average call holding time and the percentage of call attempts that result in answered calls. If the call holding time is long, less call attempts per agent are made, resulting in support for more agents. If the percentage of call attempts resulting in answered calls is high, more agents can also be supported.

An OCM agent can transfer a call to a number outside the switch only if a Distributed Communications System (DCS) link is used.

Interactions

The following features interact with the Outbound Call Management feature.

- AAR/ARS Partitioning

  If Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) is used with OCM, AAR/ARS Partitioning should be used with OCM so that OCM is guaranteed the use of certain trunks. Also, care should be taken during administration to properly assign Facility Restriction Levels (FRLs) so that OCM calls are not held up because of trunk availability.

- Abbreviated Dialing

  When an agent logs into an OCM split, the agent's Abbreviated Dialing buttons are disabled.

- Agent Call Handling

  An Agent Request for Supervisor Assistance is handled differently with OCM than it is with ACD. With OCM, the agent makes the request using the data terminal keyboard instead of the voice terminal.

- Attendant Control of Trunk Group Access

  Trunks used for OCM calls should not be controlled by an attendant.

- Automatic Call Distribution (ACD)

  An OCM agent can only be logged into one split at a time. If an OCM agent is logged into another split and tries to log into an OCM split, the login attempt is denied and the OCM adjunct is notified of the attempt.

The same agent status indications that are reported to the Call Mangement System (CMS) for ACD are also reported for OCM, with the exception of the "on ACD call" state. For OCM, this mode is reported as "on OCM call".

Two trunk status indications that are not available for ACD calls are reported to the CMS for OCM calls. These are "Seized OCM", which allows the CMS to differentiate between OCM calls and outgoing personal calls, and "Connected Outgoing", which allows the CMS to report success/failure attempts for OCM outgoing calls.

Forced Disconnect can be used just as it can be used for ACD. However, it is not recommended that a Forced Disconnect announcement be assigned for applications in which agents must handle the answered calls.

It is recommended that a Forced First Announcement not be used with OCM calls because it may cause a delay in the distribution of an OCM call to an agent. However, a Forced First Announcement can be assigned. If this is done and an announcement port is not available for an OCM call, the condition is reported to the adjunct.

- Automatic Circuit Assurance (ACA)

  Since calls are automatically made with OCM, it is possible that calls may not be dropped properly. ACA can help keep these calls from tying up trunks. The long holding time measurements provided by ACA are particularly useful with OCM. The split supervisor should be the party who is notified by an ACA referral call. Therefore, unless voice synthesis is provided, the split supervisor should have a display-equipped voice terminal.

- Bridged Call Appearance

  Since the primary function of an OCM agent is to answer OCM calls, OCM agents should not be assigned bridged call appearances. Another reason is that calls to a split cannot be bridged, and a logged-in OCM agent cannot receive personal calls anyway.

- Call Coverage

  Call Coverage should not be assigned to an OCM split. Call Coverage may, however, be assigned to the individual agent's voice terminal.

- Call Forwarding All Calls

  Call Forwarding All Calls cannot be activated after an agent has logged into an OCM split. If Call Forwarding All Calls is activated before the agent logs into an OCM split, only non-OCM calls will be forwarded. OCM calls are not affected.

- Call Pickup

  An OCM agent should not be a member of a Call Pickup group. Call Pickup is automatically deactivated when an agent logs into an OCM split.

- Class of Restriction

  Calls placed by OCM without switch call classification use the originator's Class of Restriction.

- Class of Service

  Calls placed by OCM without switch call classification use the originator's Class of Service.

- Forced Entry of Account Codes

  If the system is used only for OCM applications, Forced Entry of Account Codes should not be assigned. If the system is also used for other applications, Forced Entry of Account Codes may be used but should be assigned based on CORs so that an account code is never required on OCM calls.

- Intraflow and Interflow

  Intraflow and Interflow should not be assigned to OCM splits.

- Integrated Services Digital Network Primary Rate Interface (ISDN-PRI)

  OCM and Station Identification Number/Automatic Number Identification (SID/ANI) to Host Call Identification applications cannot co-reside on the same adjunct software, because there is only one OCM-SID/ANI link.

- Modem Pooling

  Although most OCM calls should be voice calls, OCM calls can possibly be data calls. An off-premises data call will be connected to a pooled modem (if necessary) on being answered.

- Move Agents From CMS

  The CMS cannot be used to move an OCM agent from one split to another if the agent is logged into a split. If an attempt is made to move an agent who is logged into an OCM split, the move is denied and the appropriate error message is sent back to the CMS.

- Night Service

  Night Service should not be used on OCM splits.

- Ringback Queuing

  Ringback Queuing is not supported on calls that are placed by the OCM adjunct.

- Send All Calls

  Send All Calls cannot be activated after an agent has logged into an OCM split. If Send All Calls is activated before the agent logs into an OCM split, only non-OCM calls will be sent to coverage. OCM calls are not affected.

- Service Observing

  Service Observing may be used with OCM calls the same as with other calls, except the observer cannot activate the Service Observing after logging into an OCM split.

- Station Message Detail Recording (SMDR)

  When a call is classified via Call Classification as "answered," the time the call is answered is entered in the SMDR call record. If the call is not classified correctly, it is directed to an agent who will subsequently drop the call. The SMDR record is output when the call is dropped. This short duration call should be interpreted as an ineffective call attempt.

  OCM calls are marked in the SMDR call record with condition code "B."

- Ten-Digit to Seven-Digit Conversion

  Enough tie trunks should be provided to handle OCM calls to private network endpoints.

Administration

OCM is administered by the System Manager. The administration requirements for OCM are the same as those for ACD with the following additions:

- Since OCM is an optional feature, the OCM-SID/ANI option must be administered as "yes." The ACD option must also be administered as "yes".
- Trunk groups used with OCM must be administered as to whether or not they are capable of receiving answer supervision. Answer supervision is always assumed with ISDN trunks and tie trunks.
- The answer supervision time-out parameter must be administered to the maximum allowable value (250 seconds) for trunks which expect to receive answer supervision.
- The queuing option for trunk groups should not be administered.
- Each split used for OCM calls must be administered as an adjunct-controlled split.
- The time-out interval used when no tones are detected from off-premises during dialing (Off-Premises Tone Detect Timeout Interval) must be administered.
- Agents' voice terminals should be assigned automatic answer.
- OCM hunt groups should be assigned small queues and announcements.
- The OCM Communications Link must be administered for communication between the switch and the adjunct.

Hardware and Software Requirements

A 3B2 Starcaller is required for use as the OCM adjunct.

A port on a TN765 Processor Interface circuit pack is required for the OCM Communications Link for direct RS-232 connection. Alternately, a Modular Processor Data Module (MPDM) and digital port can be used.

TN748C Tone Detector circuit packs are required for call classification and answer detection.

Each announcement requires one port on a TN750 Integrated Announcement circuit pack or announcement equipment and one port on a TN742 Analog Line circuit pack. The four analog announcements should be assigned on the TN742 ports since the TN742 can only ring four ports at a time.

Each auxiliary queue warning level lamp requires one port on a TN742 Analog Line circuit pack. A 21C-49 indicator lamp may be used as a queue warning level lamp. This lamp is about 2 inches in diameter and has a clear beehive lens. The lamp operates on ringing voltage and can be mounted at a location convenient to the split.

ACD software is required. If a CMS is to be used, CMS software is required.

PC/PBX Connection

Description

Brings the voice terminal and personal computer (PC) together into an integrated voice and data workstation. The PC can be an AT&T PC or other IBM* compatible PC.

Three software/hardware packages are available for the AT&T PC (or an IBM compatible PC):

- Package 1—Provides many phone services (such as keyboard dialing, customized phone features, personal phone directory, directory dialing, and message retrieval) and data services (such as terminal emulation, file transfer, and script programs). The hardware of the workstation includes a PC, a 7404D digital voice terminal, and a cartridge plugged into the voice terminal to provide communications between the voice terminal and the PC.

- Package 3—Provides the same phone and data services as Package 1 plus additional features (such as call log, higher file transfer rates, and the ability to take notes on calls). The hardware for a Package 3 workstation includes a PC, a digital telephone (7400-type), and an expansion board installed in the PC to provide communications between the voice terminal and the PC.

- Package 5—Provides terminal emulation which allows an AT&T PC6300 or compatible computer to emulate a 3278/3279 terminal. Package 5 is a software enhancement for Package 3 and works with the Package 3 hardware and software.

Considerations

By providing PC users with the voice and data capabilities of a fully integrated voice/data workstation, the PC/PBX Connection feature makes communications more efficient. Also, PC users with the PC/PBX Connection feature are linked for easy access to other PCs, modem pooling, and on- and off-site computers.

Interactions

None.

* Trademark of International Business Machines Corp.

Administration

The PC/PBX Connection feature is administered on a per-extension basis by the System Manager. A PC is assigned to the system just as any other station would be. That is, the station type is administered as "pc". An additional field is then specified for the type of digital voice terminal to be connected to the PC.

Hardware and Software Requirements

A port on a TN754 or TN784 (G1) Digital Line circuit pack is required for each PC to be connected. See the previous descriptions for software/hardware packages 1, 3, and 5 for additional information.

Permanent Switched Calls (V2, V3, or G1)

Description

Maintains a call between two data endpoints that should always be connected while the system is active. The specified calls are automatically placed when the system is started or restarted, and remain active until the system becomes inactive.

Data endpoints consist of digital line ports, data line ports, netcons, and DS1 tie trunks administered for "avd" or "data" communication types.

If a Permanent Switched Call (PSC) is inadvertently dropped, the system automatically reestablishes the call. The system attempts to reestablish all nonactive PSCs at 2-minute intervals. These attempts continue until all calls are completed.

Considerations

PSCs make the system responsible for placing and maintaining calls that should be present while the system is active. Only data calls can be placed in this manner.

The system can support up to 18 PSCs, indicated in a PSC list. Each PSC listed can contain up to 36 characters of dialing information (see the Data Call Setup feature description for the dialing format).

Interactions

The following features interact with the Permanent Switched Calls feature.

- Call Forwarding

  The called endpoint should not have Call Forwarding activated since the endpoint should be a final destination.

- Data Restriction

  All PSCs should be administered with Data Restriction set to prevent imposing system tones on the call. Such tones interfere with data transmission.

Administration

PSCs are assigned on a per-system basis by the System Manager. The following items require administration:

- Call List—Establish or change the list of PSCs.

- Classes of Restriction—Determine Classes of Restriction so that only PSC endpoints are allowed to call other PSC endpoints. Other users should be denied permission to call a PSC endpoint.

Before a PSC extension is entered on the call list, the System Manager must check the Property Management System (PMS) (V3 or G1) and Station Message Detail Recording (SMDR) extensions so that the same extension is not assigned twice.

A PSC between the Interface 3 circuit pack and a data module does not appear in the call list, but is administered as a link assignment.

A PSC can be dropped temporarily, for maintenance purposes, by disabling the call in administration. The call remains in the list, but is dropped until it is enabled.

Hardware and Software Requirements

PSCs do not require additional hardware or software.

Personal Central Office Line

Description

Provides a dedicated trunk for direct access to or from the public network for multi-appearance voice terminal users.

Each Personal Central Office Line (PCOL) can have an appearance at up to four multi-appearance voice terminals. Users assigned this feature press the PCOL feature button to answer and place calls—dial access is not provided. The status lamp associated with the PCOL button indicates the busy or idle status of the trunk.

An incoming PCOL call rings all voice terminals assigned the feature (ringing can be either audible or silent, depending on administration). The PCOL button status lamp flashes even if all call appearances at the voice terminal are active. If a call appearance is idle, the status lamp associated with that appearance also flashes.

Central office (CO), foreign exchange (FX), and Wide Area Telecommunications Service (WATS) trunks can be assigned to this feature.

PCOLs are not assigned a Class of Restriction.

Considerations

PCOLs are useful to users such as executives, dispatchers, or buyers with a high volume of calls going outside the system, and businesses with specialized incoming calls (such as a service department).

The system will support 25 (V1) or 40 (V2, V3, or G1) PCOLs. These lines (trunks) are not included in the trunk groups supported by the system. They are, however, included in the 200-trunk (V1, V2, or V3) or 400-trunk (G1) system limit.

Interactions

The following features interact with the Personal Central Office Line feature.

- Abbreviated Dialing

  Abbreviated Dialing can be used with the PCOL feature. However, the accessed lists are associated with the individual voice terminals.

- Bridged Call Appearance

  If a user is active on his or her primary extension number on a PCOL call, bridged call appearances of that extension number cannot be used to bridge onto the call. The call can only be bridged onto if another voice terminal is a member of the same PCOL group and has a PCOL button.

- Call Coverage
- AUDIX cannot be in the coverage path of a PCOL group.
- Hold

  When a user, active on a PCOL call, puts the call on hold, the status lamp associated with the PCOL button does not track the busy/idle status of the PCOL.

- Leave Word Calling

  Leave Word Calling messages can be stored for a PCOL group. The messages are retrieved by an authorized systemwide message retriever. When a message is stored, the remote Automatic Message Waiting lamp assigned for the PCOL group lights. One remote Automatic Message Waiting lamp is allowed per group.

- Station Message Detail Recording

  The Station Message Detail Recording (SMDR) feature can be activated for PCOL calls, but the SMDR record will not specifically identify the call as PCOL. A PCOL call can, however, be identified by the trunk access code used on the call. The call will be recorded to the extension number assigned to the voice terminal where the call was originated or answered.

- Temporary Bridged Appearance

  When a PCOL is shared (assigned to a group), any group member can bridge onto a PCOL call through the Temporary Bridged Appearance feature. The Privacy—Manual Exclusion feature can be activated on such a call if the voice terminal is assigned an Exclusion button.

- Transfer

  A PCOL can be transferred to an extension that does not have a button for that PCOL.

The following features cannot be used with the PCOL feature:

- Automatic Route Selection
- Call Forwarding All Calls
- Ringback Queuing.

Administration

PCOLs are administered by the System Manager. The following items require administration:

- Group number

- Group type (CO, FX, or WATS)
- Group name (optional, used for display purposes)
- Data Restriction activation
- SMDR activation
- Call Coverage path (redirection criteria can be Don't Answer and Cover All Calls)
- Extension numbers of voice terminals assigned to PCOL group (up to four terminals can share a PCOL)
- PCOL button (per terminal assigned to the PCOL group)
- Exclusion button (optional on a per-terminal basis)
- Remote Automatic Message Waiting lamp (one allowed per PCOL group)
- Audible or silent ringing.

The following items can be administered for the CO, FX, or WATS trunk used for the PCOL:

- Circuit pack port number
- Trunk type
- Trunk name (for display purposes)
- Trunk access code (nondialable, used to identify the trunk for SMDR)
- Outgoing dialing type
- CO disconnect timing
- Terminating area code
- Prefix for code conversion
- Toll table index for code conversion
- Prefix 1 (needed for CO and FX trunks if the prefix 1 is needed for toll calls).

Hardware and Software Requirements

Requires one port on a TN747 CO Trunk circuit pack for each CO, FX, or WATS trunk assigned as a PCOL. No additional software is required.

Personalized Ringing (V2, V3, or G1)

Description

Allows users of certain voice terminals to uniquely identify their own calls. Each user can choose one of a number of possible ringing patterns.

The eight ringing patterns are tone sequences consisting of different combinations of three tones. The eight different combinations are listed below. The tones are heard in the sequence given for each combination.

- 750 Hz, 750 Hz, 750 Hz (normal ringing)
- 1060 Hz, 1060 Hz, 1060 Hz
- 530 Hz, 530 Hz, 530 Hz
- 530 Hz, 1060 Hz, 1060 Hz
- 1060 Hz, 1060 Hz, 530 Hz
- 1060 Hz, 530 Hz, 530 Hz
- 1060 Hz, 530 Hz, 1060 Hz
- 530 Hz, 1060 Hz, 530 Hz

Each ringing pattern requires 0.6 second (0.2 second for each tone) in the 5.2 seconds ringing cycle. This 0.6 second of personalized ringing occurs at the given time during the ringing cycles of the following types of calls (times indicated are in seconds):

- Internal voice terminal, internal tie trunk, and remote access calls

  0.6 on, 0.6 personalized ringing, 4.0 off

- Attendant extended, attendant originated, and incoming trunk calls, including external tie trunk calls

  0.2 on, 0.4 off, 0.6 personalized ringing, 4.0 off

- Automatic Callback, Priority Calling, and Ringback Queuing Callback calls

  0.1 on, 0.1 off, 0.1 on, 0.3 off, 0.6 personalized ringing, 4.0 off

- Intercom Calls (7404D and 7407D voice terminals only)

  0.6 personalized ringing, 4.6 off

One of the eight ringing patterns can be specified for each eligible voice terminal (7303S and 7305S) by the System Manager. In addition, the 7404D, 7406D, 7407D, and 7103A programmable voice terminal users have the capability of setting their own ringing pattern. The 7404D user can select the desired ringing pattern via the given menu options. The 7406D or 7407D user can select the desired ringing pattern by using the Select Ring and PR

(Personalized Ringing #) buttons. The 7103A programmable voice terminal user can select one of four ringing patterns via a slide switch on the voice terminal.

Considerations

With Personalized Ringing, users working closely in the same area can each specify a different ringing pattern. This enables the users to distinguish their own ringing voice terminal from other voice terminals in the same area.

Up to eight different ringing patterns are available.

Interactions

The normal ringing cycles are altered as described in the Description of this feature.

Administration

Personalized Ringing is administered for the 7303S and 7305S voice terminals on a per-voice terminal basis by the System Manager. Administration consists of assigning one of the eight ringing patterns to each eligible voice terminal. Also, a 7404D, 7406D, 7407D, or 7103A programmable voice terminal user can specify his or her own ringing pattern. The user specified ringing pattern for a 7404D, 7406D, or 7407D, however, is lost in the event of a power failure.

Hardware and Software Requirement

No additional hardware or software is required.

Power Failure Transfer

Description

Provides service to and from the local telephone company central office (CO), including Wide Area Telecommunications Service (WATS), during a power failure.

Considerations

Power Failure Transfer provides certain voice terminals with the capability to access the local CO and to answer certain incoming calls during a power failure. These voice terminals can be used to make or answer important or emergency calls.

Each of 5 to 35 (maximum) voice terminals can be connected to a separate CO trunk for the Power Failure Transfer feature. The Power Failure Transfer feature is available in multiples of five.

Local CO trunks (including incoming WATS lines) can be used for Power Failure Transfer.

The 500-type (rotary dial) or 2500-type (touch-tone) voice terminals must be used for Power Failure Transfer. Rotary dialing must be used if the local CO accepts dial pulses only. When a Version 2 system is not in the power failure mode, Power Failure Transfer terminals (500-type rotary dial) can be used as regular extensions.

Interactions

During the Power Failure Transfer mode, no other system features can be activated.

With V1 if Night Service is activated and a power failure occurs, the system, when brought back up, will not automatically return to the Night Service mode. Instead, the system returns to the day mode. With V2, V3, or G1, if Night Service is activated and a power failure occurs, the system, when brought back up, automatically returns to the Night Service mode.

Administration

None required.

Hardware and Software Requirements

One emergency transfer panel is required for every five or six trunks assigned to Power Failure Transfer, depending on the transfer panel used. Two emergency transfer panels are available:

- Z1A Panel—Each unit serves up to six power failure transfer terminals. A ground-start key is required at each preselected voice terminal when ground-start trunks are used.

- Porta Systems* Model 574-5 Panel—Each unit serves up to five failure transfer terminals. The unit provides automatic ground start or loop start.

No additional software is required.

* Trademark of PORTA SYSTEM Corp.

Priority Calling

Description

Provides a special form of call alerting between internal voice terminal users. The called voice terminal user receives a distinctive 3-burst alerting signal.

An active single-line voice terminal user who receives a Priority Calling call will hear a distinctive 3-burst priority Call Waiting tone.

A multi-appearance voice terminal user receives the Priority Calling call on an idle call appearance. If all call appearances, including the call appearance normally reserved for call origination, are active, the caller receives busy tone. If the call appearance normally reserved for call origination is the only idle call appearance, an incoming priority call will ring at that call appearance.

A user activates priority calling by dialing a Priority Calling access code or pressing a Priority button, followed by the desired extension number.

Whether or not a user can activate Priority Calling is determined by the user's Class of Service.

Considerations

With Priority Calling, a voice terminal user can ring another voice terminal with a distinctive signal that tells the called party the incoming call requires immediate attention. The called party can then handle the call accordingly.

Call Coverage Consult calls and callback calls from Automatic Callback and Ringback Queuing are Priority Calling calls.

Interactions

The following features interact with the Priority Calling feature.

- Automatic Callback and Ringback Queuing

  Callback calls do not redirect, do not forward, and cannot be picked up by a Call Pickup group member.

- Bridged Call Appearance

  A Bridged Call Appearance receives ringing on a priority call the same as the called primary extension.

- Call Coverage

  Priority Calling calls do not redirect to coverage unless the caller activates Go To Cover. If the call redirects, it remains a Priority Call, and the covering user receives a distinctive 3-burst ringing signal.

- Call Forwarding All Calls

  Priority Calling calls (except callback calls) will forward, and the forwarded call remains a Priority Calling call.

  Priority calls cannot be forwarded to an off-premises extension.

- Call Waiting Termination

  A Priority Calling call will wait on an active single-line voice terminal even if the Call Waiting Termination feature is not assigned to the voice terminal. The active single-line voice terminal user receiving the call hears a distinctive 3-burst priority Call Waiting tone.

- Consult

  A Consult call acts as a priority call and will wait at a single-line voice terminal, even if the single-line voice terminal does not have Call Waiting Indication assigned.

- Dial Access to Attendant

  A Priority Calling call cannot be originated to the attendant. However, the attendant can originate Priority Calling calls.

- Distributed Communications System (DCS)

  On a DCS tandem call to a single-line voice terminal, the called party does not receive priority ringing if the calling party activates Priority Calling after he or she has already made the call. The called party in this situation only receives priority ringing if the calling party activates Priority Calling prior to dialing the extension.

- Ringing

  Single-line voice terminals (2500 series) can be administered so that distinctive signals are not provided.

Administration

Priority Calling is administered by the System Manager. The following items require administration:

- Priority Calling access code

- Permission to activate Priority Calling (per Class of Service).

Hardware and Software Requirements

No additional hardware or software is required.

Privacy—Attendant Lockout

Description

Prevents an attendant from reentering a multiple-party connection held on the console unless recalled by a voice terminal user.

Considerations

Privacy—Attendant Lockout provides privacy for parties on a multi-party call held on the console. The held parties can hold a private conversation without being interrupted by the attendant.

Interactions

The following features interact with the Privacy—Attendant Lockout feature.

- Trunk-to-Trunk Transfer

  Privacy—Attendant Lockout does not function when a call using the Trunk-to-Trunk Transfer feature is held on the console.

- Individual Attendant Access (V2, V3, or G1)

  Privacy—Attendant Lockout applies only to attendant group calls. Individual attendant calls are not affected.

Administration

Privacy—Attendant Lockout is administered on a per-system basis by the System Manager. The only administration required is to administer whether or not attendant lockout is active.

Hardware and Software Requirements

No additional hardware or software is required.

Privacy—Manual Exclusion

Description

Allows multi-appearance voice terminal users to keep other users with appearances of the same extension number from bridging onto an existing call.

Exclusion is activated by pressing the Exclusion button on a per-call basis. If the Exclusion button is pressed while other users are bridged onto the call, the other users are dropped from the call. The Privacy—Manual Exclusion feature is automatically deactivated when the Exclusion button is pressed a second time or when the party who activated Privacy—Manual Exclusion is dropped from the call.

Privacy—Manual Exclusion is used with the Personal Central Office Line, Terminating Extension Group, and Bridged Call Appearance features.

Considerations

Privacy—Manual Exclusion prevents users who have an appearance of another terminal's extension from bridging onto that extension.

Interactions

The following features interact with the Privacy—Manual Exclusion feature.

- Bridged Call Appearance

  When Privacy—Manual Exclusion is activated, all other users are prevented from bridging onto the active call.

Administration

Privacy—Manual Exclusion is administered on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Exclusion button to the desired voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Property Management System Interface (V3 or G1)

Description

Provides a communications link between the System 75 and a customer-owned Property Management System (PMS). The PMS allows a customer to control certain features used in both a hospital-type and a hotel/motel-type environment.

The communications link allows the PMS to interrogate the system and allows information to be passed between the system and the PMS. Routine operations related to the following features are simplified through this message exchange capability:

- Message Waiting Notification
- Controlled Restriction
- Housekeeping Status
- Check-In/Check-Out
- Room Change/Room Swap.
- Names Registration (G1)
- Guest Information Input/Change (G1)
- Support of 5-Digit Extension Numbers.

Message Waiting Notification, Controlled Restriction, and Housekeeping Status are optional features. A customer may elect to operate each of these features from the system only or to operate each of these features from either the system or a PMS.

Check-In/Check-Out, Room Change/Room Swap, Names Registration (G1), and Guest Information Input/Change (G1), are controlled from the PMS as long as the communications link between the system and the PMS is operational. If the link is not operational (the link is down), these features are affected as follows:

- Control of Check-In/Check-Out transfers to the system. With the system in control, Check-In/Check-Out operations are performed via feature buttons.
- Control of Guest Information Input/Change (G1) transfers to the system. With the system in control, Guest Information Input/Change operations are performed by the System Manager via system administration commands.
- The system does not support Room Change/Room Swap as such. However, the equivalent of Room Change/Room Swap is executed through the system by activating Check-Out followed by Check-In.
- Names Registration information, which is normally sent automatically from the PMS to the switch, can be entered manually at the switch by the System Manager.

The PMS Interface provides the following:

- A communications protocol for controlling message exchange between the system and a PMS
- An application module for controlling the operation of the PMS features
- Status data on all guest/patient rooms for selected features

The protocol is full-duplex, asynchronous and provides the mechanisms for setting up a data session with the PMS, message exchange control, error identification, and recovery. The interface supports standard data rates (1200 bps in V3 and 1200, 2400, 4800, or 9600 bps in G1).

With G1, two protocol modes are provided; the "normal" protocol mode as described above, and the "transparent" protocol mode. The transparent protocol mode spports ASCII character transmission and is required for G1 PMS features such as Names Registration and Guest Information Input/Change. G1 systems may be administered to use either the normal or transparent protocol mode.

The application module of the PMS Interface implements requested features and provides backup procedures if the communications link between the PMS and the system is down. Whether or not the communications link with the PMS is operational, the system always maintains the following data for each room:

- Whether the room is vacant or occupied
- Whether the voice terminal's Message lamp is on or off
- Whether a Controlled Restriction is active at the voice terminal and, if so, which one.
- The guest's name and coverage path (G1).

When the link to the PMS is down, the system automatically activates Check-In/Check-Out for the attendant console and front desk terminal with display capability, and continues to support PMS features that are activated from guest/patient room voice terminals.

When the link is again operational, the system sends one of the following messages to the PMS:

- No room status changes occurred during loss of communications.
- Room status changes did occur during loss of communications; therefore, a status data exchange is needed to synchronize the system and the PMS databases.
- The system failed momentarily, destroying its record of room status; therefore, a room status data exchange (full transfer of data from the PMS to the system) is needed to synchronize the system and the PMS databases.

Also, when the PMS link is down or if a PMS is not used, the system maintains a log, called an audit trail report, of all events that would normally be sent to the PMS.

The audit trail data (accessed via the System Access Terminal [SAT] [V3] or the Manager I terminal [G1]) is a sequential listing of all PMS transactions executed by the system when the PMS data link was down. Also included in the audit trail are some error events that may have occurred when the link was either up or down. If a printer is configured in the system, copies of the audit trail data will help the administrative staff to restore the room status of system and the PMS.

In addition to the PMS audit trail report, if the system has an operational PMS log printer and the PMS link is down, Housekeeping Status changes will be printed as they occur. The Housekeeping Status report will contain the following information:

- Room number
- Feature Access Code (FAC) dialed
- Any additional information digits that were dialed
- Reason for the entry (error message)
- Time the error occurred.

In addition to the PMS log printer, a PMS Journal/Schedule printer is also available. The PMS Journal/Schedule printer prints reports on Automatic Wakeup activity. Emergency Access to the Attendant activity, and scheduled reports.

A supporting function called Room Data Image synchronizes the switch and PMS databases after a PMS link goes down and comes back up. The information included in the Room Data Image is as follows:

- Room extension
- Whether the room is occupied or vacant
- Message Waiting lamp status
- Controlled Restriction status
- Guest's name (G1 only)
- Call Coverage path (G1 only).

Message Waiting Notification

Message Waiting Notification requests are originated from attendant consoles, front desk terminals, or PMS terminals. When a request is entered, the PMS sends a message to the system to change the state of the Message lamp associated with a certain extension number. If the Message lamp has been turned on by the Audio Information Exchange (AUDIX) or Leave Word Calling, the PMS cannot be used to turn the lamp off. However, in the transparent mode, certain events may cause the switch to inform the PMS that the Message lamp has been turned on by AUDIX or Leave Word Calling.

Any console or terminal used to activate and deactivate Message Waiting Notification must be assigned a Console Permissions Class of Service. The affected extension must have a Client Room Class of Service.

Controlled Restriction

When the Controlled Restriction feature is activated through the PMS, the PMS sends a message to the system to assign one of the following restrictions to the voice terminal in a guest/patient room:

- No restriction
- Outward restriction
- Total restriction
- Station-to-station restriction
- Termination restriction
- Combined outward and termination restriction
- Combined Outward and Station-to-Station restriction
- Combined Termination and Station-to-Station restriction.

If a PMS is not used or if the communications link is down, the attendant or front desk user can still set the Controlled Restriction for a voice terminal, because activation of this feature is independent of the PMS. When the communications link is again operational (if a PMS is used), the system asks for a database exchange and all status changes are sent to the PMS. At this time, controlled restrictions can still be assigned.

If user-controlled restrictions are activated or deactivated from the switch, the PMS receives a message with this information.

Housekeeping Status

The housekeeping staff can enter status information using voice terminals located in guest/patient rooms or using designated terminals. Up to ten Housekeeping Status Access Codes can be assigned in the system.

- Room Voice Terminal Access Code

  After the Access Code is dialed, the system accepts up to six additional information digits. These information digits can be used for items such as maid identification.

- Designated Voice Terminal Access Code

  After the Access Code is dialed, the system waits for the room extension to be entered and then will accept up to six additional information digits.

If a PMS is used, the system notifies the PMS when Housekeeping Status information is entered. If a PMS is not used, if the communications link is down, or if a PMS is connected, but housekeeper information is not sent (the Housekeeper Information Configuration field on the Hospitality-Related System Parameters form is administered as "act-nopms"), then Housekeeping Status information is written to a log, can be accessed through the SAT (V3) or Manager I terminal (G1), and can be sent to a printer to obtain a hard copy. If the system has a PMS log printer and the PMS link is down, each entered event will be printed as it occurs.

If a PMS is used, but goes down, and a PMS log printer is not operational, the Housekeeping Status access codes cannot be used.

Check-In/Check-Out

A Check-In request deactivates the Outward Controlled Restriction level on the terminal in a guest/patient room. A Check-Out request deactivates any Controlled Restrictions and changes the Controlled Restriction level to Outward Restriction, checks for any messages, clears the wakeup request (if there is one), and deactivates Do Not Disturb (if activated).

If a PMS is not used or if the communications link is down, Check-In and Check-Out can be activated from an attendant console or a front desk terminal with display capability and console permission. Two buttons are required, Check-In and Check-Out. Pressing either button places the display in the respective mode and allows the touch-tone buttons to be used for entering data (rather than for placing calls).

The user exits the Check-In or Check-Out mode by pressing another button associated with the display (such as the Normal Mode button). This restores the display and the touch-tone buttons to normal operation.

With G1, a Check-In/Check-Out request also sends information for the Names Registration feature to the switch. This information includes the guest's name (up to 15 characters), room extension, and Call Coverage path. The switch must be assigned the "transparent" communications protocol mode for this information to be transferred between the switch and the PMS. If the PMS link is down, and Check-In is done from an attendant console or display-equipped front desk terminal, the guest's name and coverage path information is not automatically updated at Check-In.

If a guest/patient room has both a voice and a data extension, the Check-Out request applies only to the voice extension.

Room Change/Room Swap

These features are provided only through a PMS and must be activated from a PMS terminal. When either Room Change or Room Swap is activated, the PMS sends a message to the system. When Room Change is activated, data pertaining to the old room, including a pending wakeup request, the guest's name (G1, transparent mode), and the guest's Call Coverage path (G1, transparent mode) is moved to the new room. When Room Swap is activated, the data pertaining to the two rooms are swapped. When either feature is activated, if the occupancy status is inconsistent, the system sends an error message to the PMS.

Names Registration (G1)

Automatically sends a guest's name and room extension from the PMS to the switch at Check-In, and automatically removes this information at Check-Out. In addition, the guest's call coverage path (for example, voice mail or hotel operator) will also be sent from the PMS to the Switch during Check-In. The guest's call coverage arrangement is set to the administered "Default Call Coverage Path for Client Rooms" at check-out. The Names Registration feature is described in detail elsewhere in this chapter.

Guest Information Input/Change (G1)

Guest Information Input/Change allows guest information (name or coverage path) to be entered or altered subsequent to the check-in message. Hotel personnel can change this information at the PMS and it is automatically sent to the switch.

The Guest Information Input/Change function is used in those situations when the guest's name associated with an extension must be changed, input of a guest's name must be made after the check-in sequence has taken place, or a change in call coverage arrangement must be made.

Considerations

The PMS Interface feature provides a collection of features needed in a hospital or hotel/motel environment: Message Waiting Notification, Controlled Restriction, Housekeeping Status, Check-In/Check-Out, Room Change/Room Swap, Names Registration, and Guest Information Input/Change. All of these features, except Room Change/Swap and Check-In/Check-Out, can operate through the system with or without a PMS. When these features are entered through a PMS, the system provides the communications interface needed for their correct operation.

A customer may elect to use Leave Word Calling or Integrated Message Center Service for the hospital or hotel/motel staff and Message Waiting Notification for guests/patients. However, if Message Waiting Notification is not used, Integrated Message Center Service can be used for both.

A PMS extension cannot be removed while the PMS link is active.

PMS link parameter changes do not go into effect until the PMS link is reset.

With G1, the "normal" protocol mode allows extensions of up to 4 digits in length. The "transparent" protocol mode allows extensions of up to 5 digits in length.

When a "save translations" is done on a G1 system with the transparent protocol mode active, station names with Client Room Class of Service are saved as "blank" and coverage paths are saved as the "Default Coverage Path for Client Rooms".

Interactions

The following features interact with the Property Management System Interface feature.

- Attendant Console or Front Desk Terminal

  The Controlled Restriction, Check-In/Check-Out, and Message Waiting Notification features can be activated at an attendant console or a front desk terminal with console permission. Also, the attendant console can receive visual notification of the status of the communications link between the system and the PMS.

- Audio Information Exchange (AUDIX) Interface

  Message lamps activated by this feature cannot be deactivated by feature buttons or by feature messages from the PMS.

- Automatic Wakeup

  An Automatic Wakeup request for a guest/patient room is set or canceled as a result of Room Change/Room Swap or Check-Out.

- Do Not Disturb

  A Do Not Disturb request for a guest/patient room is set or canceled as a result of a different Controlled Restriction, Room Change/Room Swap, or Check-Out.

- Leave Word Calling

  Message lamps activated by this feature cannot be deactivated by Manual Message Waiting feature buttons.

  With a G1 system using the transparent protocol mode, any Leave Word Calling messages that are present when a check-in request is processed are deleted.

  If Room Change is activated, Leave Word Calling messages for the old room will not be moved to the new room. If Room Swap is activated, Leave Word Calling messages for the two rooms will not be swapped. Therefore, use of the Leave Word Calling feature should not be encouraged in guest/patient rooms.

- Restriction—Controlled

  Controlled Restriction for a group of users extensions, when activated from the switch, is not conveyed to the PMS. Also, the PMS is not able to remove such restrictions by sending feature messages.

Administration

The following Feature-Related System Parameters may be administered:

- Message Waiting Notification—One of two choices must be administered:
  - Active with no PMS message exchange (act-nopms)
  - Active with PMS message exchange (act-pms)
- Controlled Restriction—One of two choices must be administered:
  - Active with no PMS message exchange (act-nopms)
  - Active with PMS message exchange (act-pms)
- Housekeeping Status Information—One of two choices must be administered:
  - Active with no PMS message exchange (act-nopms)
  - Active with PMS message exchange (act-pms)
- If active is selected, the following additional administration is required:
  - The number of additional information (Housekeeper Identification) digits from 0 to 6 that can be dialed.
- The extension numbers assigned to the PMS Journal/Schedule printer and PMS log printer, if used, and the extension number assigned to the PMS. Before an extension is assigned, the System Manager should check to make sure that the extension is not already assigned as a Station Message Detail Recording (SMDR) or Permanent Switched Call (PSC) extension.
- Seconds Before PMS Link Idle Timeout—Specifies the number of seconds that the system will wait before it concludes that the PMS is not sending data across the transmission link—choice is a number of seconds from 5 to 20.
- Milliseconds Before PMS Link Acknowledgment Timeout—Specifies the maximum time the system expects acknowledgment from the PMS that a message was received correctly—choice is a number of milliseconds from 100 to 500.
- PMS Link Maximum Retransmissions—Specifies the maximum number of times that the system will retransmit a message in response to a negative acknowledgement or send an inquiry for an acknowledgement from the PMS for a message before giving up on on the message transmission—choice is a number from 1 to 5.
- PMS Link Maximum Retransmission Requests—Specifies the maximum number of times that the system will accept requests from the PMS to resend a reply (acknowledgement or negative acknowledgement) that the system did not receive before giving up on the incoming message—choice is a number from 1 to 5.

- PMS Protocol—Specifies the communication protocol mode used between the switch and the PMS. The choices are either "normal" or "transparent"
- Default Coverage Path for Client Rooms—Specifies the coverage path value that is set for an extension when the switch receives a "check-out" message while in the "transparent" communication protocol mode, or when a save translation is stored for extensions with a Client Room Class of Service—choice is a number from 1 to 600.

In addition to system parameters, the following items can be administered:

- Message Waiting Notification activate and deactivate buttons—per attendant console and front desk terminal
- Check-In and Check-Out buttons—per attendant console and front desk terminal
- Console Permission Class of Service needs to be assigned to the front desk terminal.
- Any console or terminal used to activate and deactivate Message Waiting Notification must be assigned a Console Permission Class of Service. The affected extension must have Client Room Class of Service.

Hardware and Software Requirements

A PMS, if used, can be connected through a Modular Processor Data Module (MPDM) and port on a Digital Line circuit pack or through an Asynchronous Data Unite (ADU) and a port on a Data Line circuit pack. A Digital Terminal Data Module (DTDM) (with a null modem), 7400A data module, and 7400B data module can also be used for the PMS link. Journal/Schedule and PMS log printers can be used and also require at least an MPDM and a port on a Digital Line circuit pack or an ADU and a port on a Data Line circuit pack. The Journal/Schedule and PMS log printer functionality can be on the same or two distinct printers.

Optional Hospitality Services software is required to provide the Property Management System Interface feature.

Queue Status Indications (V3 or G1)

Description

Provides indications of queue status for Automatic Call Distribution (ACD) calls based on the number of calls in queue and time in queue. These indications are provided via lamps assigned to the terminals or consoles of split agents or supervisors. In addition, an auxiliary warning lamp can be provided to track queue status based on time in queue. Also, display-equipped voice terminals and consoles can display the time in queue of a split's oldest call and the number of calls in that split's queue.

Two types of Queue Status Indications are provided:

- Number of Queued Calls

  The Number of Queued Calls status indication is based on the total number of calls in queue at a split. The status indication can be provided by an NQC (Number of Queued Calls) button with associated lamp on a voice terminal or console. Each split is assigned a Number of Queued Calls warning threshold of 1 to 99 calls. When this threshold is reached, the lamp associated with the NQC button flashes. If there are calls in the queue, but the threshold is not reached, the lamp lights steadily. If there are no calls in queue, the lamp goes dark.

  In addition to the NQC button(s), the Number of Queued Calls status indication can be provided by an auxiliary queue warning lamp. This lamp can be installed at any location convenient to the split agents. When the Number of Queued Calls warning threshold is reached, the auxiliary queue warning lamp lights.

- Oldest Queued Time

  The Oldest Queued Time status indication is based on the time in queue of the oldest call in a split queue. The status indication can be provided by an OQT (Oldest Queued Time) button with associated lamp on a voice terminal or console. Each split is assigned an Oldest Queued Time warning threshold of 0 to 999 seconds. When the oldest call in queue has been in queue for this length of time, the lamp associated with the OQT button flashes. If there are calls in the queue, but the threshold is not reached, the lamp lights steadily. If there are no calls in queue, the lamp goes dark.

  In addition to the OQT button(s), the Oldest Queued Time status indication can be provided by an auxiliary queue warning lamp. This lamp can be installed at any location convenient to the split agents. When the Oldest Queued Time warning threshold is reached, the auxiliary queue warning lamp lights.

Each NQC and OQT button is associated with a specific split. Display-equipped voice terminals and consoles can display queue status information for a split by pressing the OQT or NQC button. The same information is displayed no matter which of the two buttons is pressed. The split name (or extension if name is not assigned), Oldest Queued Time, and Number of Queued Calls are displayed for 5 seconds unless the displaying terminal or console receives an incoming call or the display is put into another mode. Otherwise, at the end of 5 seconds, the display returns to its previous condition. If the display has two lines, the queue status information is displayed on the second line.

In addition to providing queue status information for splits, the Queue Status Indications feature can be used to provide status information for attendant groups or other hunt group types (Direct Department Calling and Uniform Call Distribution). The feature works the same with attendant groups as it does with splits, except the button names are different and the display shows "attendant" instead of the split name or extension, and all status information applies to the attendant group queue. The attendant buttons are the AQT (Attendant group's Queued Time) and the AQC (Attendant group's Queued Calls) buttons.

Considerations

The Queue Status Indications feature allows split agents, split supervisors, and attendants to monitor queue activity. This information is extremely useful in that it allows the agents, supervisors, and attendants to better manage their time.

An NQC, OQT, AQC, and/or AQT button can be assigned to any multi-function voice terminal or console.

Interactions

The following features interact with the Queue Status Indications feature.

- Attendant Display and Voice Terminal Display

  The timer and the queue status information may be displayed at the same time. When this happens, the timer occupies the last eight display positions and the number of queued calls is not displayed. This applies only to one-line displays. With a two-line display, the timer is displayed on the first line and the queue status information is displayed on the second line.

- Move Agent From CMS

  When the Call Management System (CMS) is used to move an agent from one split to another, all split associated buttons (including NQC and OQT buttons) become associated with the new split.

Administration

The Queue Status indications feature is administered by the System Manager. The following items require administration:

- Buttons:
  - NQC (Number of Queued Calls)
  - OQT (Oldest Queued Time)

— AQT (Attendant Queued Time)

— AQC (Attendant Queued Calls)

- Number of Queued Calls warning threshold (1 to 99 calls) (per split or attendant group)
- Oldest Queued Time warning threshold (0 to 600 seconds) (per split or attendant group)
- Port number assigned to auxiliary queue warning lamp (per split or attendant group).

Hardware and Software Requirements

Each auxiliary queue warning lamp requires one port on a TN742, TN746, or TN769 Analog Line circuit pack. A 21C-49 indicator lamp may be used as an auxiliary queue warning lamp. This lamp is approximately 2 inches in diameter and has a clear beehive lens. The lamp operates on ringing voltage and can be mounted at a location convenient to the group.

ACD software is required.

Recall Signaling

Description

Allows a single-line voice terminal user, who is active on a call, to place the party on hold and obtain recall dial tone by pressing the Recall button or by flashing the switchhook. The user can then place another call or activate a feature, and return to the held party by pressing Recall twice or by flashing the switchhook twice.

Considerations

Recall Signaling provides a single-line voice terminal user with the ability to place a call on hold and use the voice terminal for other operations. The user can then return to the held call.

Recall Signaling cannot be used to answer a waiting call (Version 1 only).

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Recent Change History (G1)

Description

Allows the user to view or print out a history report of the most recent administration and maintenance changes. This report may be used for diagnostic or information purposes.

The system maintains a log in a software buffer of the most recent administration and maintenance commands, up to a maximum of 250. The log is called the **transaction log.** The commands must be "data affecting" and successfully entered to be saved in the transaction log. The "data affecting" commands are called **data commands.**

The transaction log can be displayed or printed as a report by entering the **list history** or **list history print** command at the Manager I terminal or a remote terminal by the following users:

- Local Customer Administrator
- Local Craft
- Remote Customer Administrator
- Remote Craft.

Commands

A command is made up of multiple words, typed on the Manager I keyboard, that instruct the system to do a task. The system command structure is made up of an Action, Object, and Qualifier format.

The first command word entered is the **action.** It specifies the operation to be performed (add, display, change, remove, etc.).

The second command word entered is the **object.** It specifies the specific object to be operated on (station, trunk group, hunt group, etc.).

The third command word entered is the **qualifier.** The Qualifier is one or more words or digits used to further identify or complete the Object. Depending on the Object used, a Qualifier may or may not be used. Some commands do not have a qualifier, such as the Dial Plan and Feature Access Codes.

Data Commands

Only those administration and maintenance commands that change the data state associated with any object and qualifier are maintained in the log; the commands are called **data commands.**

Administration data commands affect translation data; maintenance data commands affect state information. For example, the **change station 3600** command will change the state of the translation data and will be classified as a data command and entered in the log. However, the command **display station 3600** will not change the state of the translation data and will not be entered in the log.

The following are the commands that are classified as data commands and are saved in the transaction log:

- add, change, remove, duplicate
- set, reset
- busyout, release
- clear
- enable, disable
- test
- wp (write physical)
- recycle

The following are the commands that are not classified as data commands and are not saved in the transaction log:

- list, display, status
- monitor
- get
- rp (read physical)
- save
- load, restore

Transaction Log and History Report

Other associated data is saved in the transaction log along with the data commands, and this data is:

- date, time
- port, login
- action, object, qualifier

A history report of the transaction log data can be displayed or printed by the system administrator by entering the **list history** or **list history print** command. The data commands are displayed or printed in last in, first out order, up to a maximum of 250 entries.

An example of a recent change history report is shown in Figure 3-27, and the following is a brief description of the report entries:

- Date—The date the data command was entered; for example, "07/18".
- Time—The time the data command was entered; for example, "12:34".
- Port—The port, or group of ports, the user was connected to. The users are defined as:
  - SAT
  - INADS
  - SMDR
  - EPN
  - NET

  Table 3-E shows the way the software correlates the port number to the user that is displayed under Port on the report.

Table 3-E. Software Port Correlations

Port No.	Access Method	Intended Use	Displayed
0	MB (EPN)	SAT	EPN
1	MB (EPN)	(not used)	EPN
2	MB (EPN)	(not used)	EPN
3	Netcon		NET
4	Netcon		NET
5	Netcon		NET
6	Netcon		NET
7	MTP	SAT	SAT
8	MTP	INADS	INAD
9	MTP	SMDR	SMDR

Legend: EPN—Expansion Port Network
INADS—Initialization and Administration System
MB—Maintenance Board
MTP—Maintenance Tape Processor
Netcon—Network Controller
SAT—Manager I Terminal
SMDR—Station Message Detail Recording

- Login—The system login of the user entering the data command, for example, "craft."
- Actn—The first command word entered; specifies the operation to be performed; for example, "add, change, remove."

- Object—The second command word or words entered; specifies the specific object to be acted on; for example "station, trunk group." (If the object is multiple words, only the first word will be displayed. All succeeding words will be treated as qualifiers.)

- Qualifier—The third command word or words entered; one or more words or digits used to further identify or complete the object; for example, "1120" (the station number). Some commands do not have a qualifier, such as "dialplan."

- Date of Translation Loaded—The time and date that the translation is saved on tape. When a translation is saved on tape, by entering the **save translation** command, the time and date of the save is logged on the tape. Whenever the system is cold started or rebooted, the transaction log is loaded from the tape and the time and date are included on the Recent History Report; for example, "19:53 Wed Mar 15, 1990".

```
                                History
                 Date of Translation Loaded: 19:53 Wed Mar 15, 1990

Date   Time    Port  Login  Actn  Object        Qualifier
----   ----    ----  ----   ----  ------        --------
07/18  12:34   EPN   cust   add   station       1120
07/18  12:23   EPN   cust   cha   dialplan
07/16  09:44   SAT   craft  rel   station       504
07/16  09:22   SAT   craft  busy  station       504
07/15  15:25   SAT   cust   cha   station       507
07/15  15:19   SAT   cust   cha   system-param  features
07/15  15:18   NET   inads  dup   station       20001 start 30001 board c11 count 8
07/15  15:16   SAT   cust   add   station       507
07/15  15:15   SAT   cust   add   station       506
07/15  15:09   SAT   cust   add   station       505
07/15  15:06   SAT   cust   cha   station       504
07/15  15:04   SAT   cust   add   station       504
07/15  15:02   SAT   cust   add   station       503
07/15  15:01   SAT   cust   add   station       502
07/15  14:56   SAT   cust   add   station       501
07/15  14:23   SAT   cust   cha   dialplan
```

Figure 3-27. Recent Change History Report

Considerations

A maximum of 250 data commands are stored in the transaction log.

The Permission Administration Form shows the command permission categories that a user can access, for example, "Administer Features." There are no permission restrictions for access to the Recent Change History reports. The local and remote customer and craft administrators have unrestricted access.

The data commands and associated data fields in the transaction log are limited in length to keep entries on the report short. This aids in visual searching of the report. The following shows the field size limits:

Field	Bytes (Digits)
date	5
time	5
port	4
login	7
action	4
object	12
qualifier	36

Interactions

The following features interact with the Recent Change History feature.

- Call Processing
- There are no interactions with any call processing features.
- Other Users

  When a user requests a Recent Change History report, it takes a little time to read all the pages of the report. If, during this time, other users are entering data commands and altering the transaction log, the oldest entries in the transaction log may have been overwritten by the data commands entered by these other users.

- Set Time Command

  The use of the "Set Time" maintenance command to change the system clock can make the Recent Change History report look as if it is not in true last-in, first-out order.

Administration

None Required.

Hardware and Software Requirements

No additional hardware or software is required.

Recorded Announcement

Description

Provides a recorded announcement to the following types of calls:

- Direct Inward Dialing (DID) calls that cannot be completed as dialed
- Incoming Private Network Access calls that cannot be completed as dialed
- Direct Department Calling and Uniform Call Distribution calls that have been in queue for an assigned interval
- Automatic Call Distribution (ACD) calls that have been in queue for an assigned interval (V3 and G1 only)
- Any call whose destination is a Recorded Announcement extension (V3 and G1 only)
- Incoming calls to a user.

With Version 1 and Version 2 systems, as many as ten recorded announcements can be provided. Each announcement requires separate announcement equipment and a port on an analog circuit pack. With Version 3 and DEFINITY Generic 1, the TN750 circuit pack can provide up to 64 integrated announcements.

When a call is directed to an integrated announcement (V3 or G1), the system checks to see if a port on a TN750 circuit pack is available. If a port is available, the call is immediately connected to the announcement. If all 16 ports are already connected to other callers, the call waits in one of the 50 queue slots for a port. When a port becomes available, the waiting call will be connected to that port if no other callers are seeking the same announcement or if four other callers are seeking the same announcement.

For additional information on how Recorded Announcement functions, see the following features:

- Automatic Call Distribution (V3 or G1)
- Direct Department Calling and Uniform Call Distribution
- Intercept Treatment.

Considerations

Recorded Announcements can be used to perform many tasks such as letting users know a call cannot be completed as dialed, letting callers know their call is in queue, or that all lines are busy. By letting Recorded Announcements perform these tasks, attendants and other users are free to perform other operations.

With Version 1 and Version 2 systems, as many as ten recorded announcements can be provided. Version 3 or DEFINITY Generic 1 systems, allow as many as 64 recorded announcements.

With Version 2 or 3, or DEFINITY Generic 1, when a DID call cannot be completed as dialed and goes to an announcement, a second DID call going to the same announcement will wait in queue for the announcement if the announcement is one of the first five announcements. Announcements six through ten will not work this way (the calling party hears nothing).

The TN750 Announcement circuit pack, available with Version 3 and DEFINITY Generic 1, has a simultaneous call capacity of 80 calls (16 ports with 5 calls on each port). Therefore, it is possible, if the same announcement is playing on all 16 ports, to have as many as 80 callers listening to the same announcement at the same time. The TN750 Announcement circuit pack provides a total of 4 minutes and 16 seconds of announcement time.

Any announcement connected to the system via an analog port can only play to one caller at a time.

A user cannot enter into an announcement session for recording/playback/deleting if any of the following are true:

- A user is already in an announcement session.
- An announcement is already being played (on port 0 which is used for recording).
- The announcement circuit pack board is not in normal mode (uploading/downloading mode).

Interactions

Recorded Announcement is used in conjunction with the Automatic Call Distribution, Intercept Treatment, Direct Department Calling, and Uniform Call Distribution features.

Administration

Recorded Announcement is administered by the System Manager. With Versions 1 and 2, each announcement must be assigned an analog port number and a name.

With Version 3 or DEFINITY Generic 1, each announcement can be assigned an extension, a type, a name, and whether or not it has a queue. The type of Recorded Announcement can be either analog or integrated. If the type is analog, then the queue length and analog port must be administered for each announcement. If the type is integrated (the announcement is recorded directly onto the TN750 circuit pack), then each announcement must be assigned an integrated announcement board number and the System Manager must specify whether the announcement is protected against being overwritten and deleted.

Hardware and Software Requirements

With Versions 1 and 2, each announcement requires announcement equipment and one port on a TN742 or TN769 Analog Line circuit pack. No additional software is required.

With Version 3 or DEFINITY Generic 1, announcements can be either analog or integrated. Each analog announcement requires announcement equipment and one port on a TN742 Analog Line circuit pack. Each integrated announcement requires one port on a TN750 Integrated Announcement circuit pack. Since the integrated announcement is recorded onto the circuit pack, no announcement equipment is required. No additional software is required.

Recorded Telephone Dictation Access

Description

Permits voice terminal users, including Remote Access and incoming tie trunk users, to access dictation equipment.

The dictation equipment is accessed by dialing an access code or extension number (depending on how the feature is administered). After the dictation equipment is accessed, the start/stop function can be voice- or dial-controlled. Other functions such as initial activation and playback are controlled by additional dial codes. The specific dial codes depend on the dictation equipment selected.

Considerations

This feature provides dictation equipment which users can access at their own convenience. Dictation can be recorded, corrected, and played back by the user.

Interactions

The Recorded Telephone Dictation Access feature cannot be used with the following features:

- Automatic Route Selection
- Conference—Attendant
- Conference—Terminal.

Administration

Recorded Telephone Dictation Access is administered on a per-system basis by the System Manager. The following items require administration:

- One port on an Analog Line circuit pack (per dictation machine) and an extension number

  or

- One port on an Auxiliary Trunk circuit pack and a trunk access code.

Hardware and Software Requirements

Requires telephone dictation machines and, depending on the type of machine, one port on a TN742 Analog Line circuit pack or one port on a TN763 Auxiliary Trunk circuit pack for each machine assigned. No additional software is required.

Remote Access

Description

Permits authorized callers from the public network to access the system and then use its features and services.

Remote Access users can dial into the system using central office (CO), foreign exchange (FX), or 800 Service trunks. The Remote Access feature is assigned an extension number, as any voice terminal. When a call is received on a trunk group dedicated to Remote Access, the system routes the call to the assigned extension number. If Direct Inward Dialing (DID) is provided and if the Remote Access number is within the range of numbers that can be accessed by DID, then the Remote Access feature can be accessed through the DID feature.

After access to the feature, the user hears system dial tone, and, for system security, may be required to dial a Barrier code. If a valid Barrier code is dialed, the user again hears dial tone, and can place calls the same as an on-premises user.

The destination of incoming, non-DID, trunk calls can be an attendant or an extension number. The destination is specified on each individual trunk group. When the trunk group is dedicated to Remote Access, the Remote Access extension number is specified. In this case, the user does all dialing. If an attendant is needed on a call, the user dials the public network telephone number assigned, the Barrier code, and "0" (the attendant access code). To provide attendant-assisted calling, service can be arranged so the attendant handles calls during the day, but Remote Access applies after normal business hours. This is accomplished by setting the trunk group destination as "0" (the attendant), and specifying the Remote Access extension number as the Night Station number. incoming calls route to the attendant unless the Night button on the primary console is pressed. When Night Service is in effect, incoming calls route to Remote Access.

Considerations

Remote Access provides a caller with access to the system and its features from the public network. An executive can make business calls from home or use the Recorded Telephone Dictation Access feature to dictate a letter. Remote Access may also be used from any extension on the switch. This allows authorized users to access system features from any voice terminal extension.

Ten Barrier codes, each with a different Class of Restriction (COR), can be administered. The Barrier codes can be from four to seven digits, but all codes must be the same length. Barrier codes not only provide system security but also define the calling privileges through the administered COR.

Ringback Queuing cannot be used on a Remote Access call since the system does not have access to the calling (outside) number.

Any feature requiring recall dial tone (for example, Hold and Transfer) cannot be accessed remotely.

The Remote Access caller must use a touch-tone voice terminal, or equivalent.

After a Digital Terminal Data Module's (DTDM's) baud rate is changed from 9600 to 1200, the DTDM cannot be accessed by Remote Access until an internal call is made to the DTDM. A Remote Access user attempting the call before an internal call is made will receive intercept treatment.

Note: AT&T has designed the Remote Access feature incorporated in this product that, when properly administered by the customer, will enable the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes, and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of Remote Access features. In such events, applicable tariffs require that the customer pay all network charges for traffic. AT&T cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

Interactions

The following features interact with the Remote Access feature.

- Abbreviated Dialing

  Since Abbreviated Dialing lists are associated with specific voice terminals, Remote Access cannot be used to access Abbreviated Dialing.

- Authorization Codes (V3 or G1)

  When a remote access caller dials the assigned remote access number and establishes a connection to the system, the system may request the caller to dial an Authorization code in addition to a Barrier code. The Authorization code defines his or her calling privileges within the system.

- Class of Restriction

  COR restrictions do not block access to the Remote Access feature.

- Night Service—Night Station Service

  The Remote Access extension number can be specified as the Night Station extension number on an incoming, non-DID, trunk group.

Administration

Remote Access is administered by the System Manager. The following items require administration:

- Extension number
- Barrier code length (from four to seven digits or blank [no barrier codes])
- Barrier codes
- COR (per Barrier code)
- Trunk groups
- Authorization codes (V3 or G1).

Note: Remote Access numbers and Barrier code assignments should be kept confidential to prevent unauthorized persons from accessing the system.

Hardware and Software Requirements

If Remote Access is not available via DID, dedicated trunks must be provided. No additional software is required.

Report Scheduler and System Printer (G1)

Description

Allows the System Manager to schedule selected administration commands to be printed by an asynchronous printer. Reports are scheduled at 15-minute intervals for any combination of days of the week. Most list, display, or test commands may be scheduled.

Reports may be scheduled, changed, listed, and removed via the system's Manager I Terminal.

The Basic Call Management System (BCMS) feature uses the Report Scheduler and System Printer feature to print BCMS reports.

Scheduling (Adding) Reports

The System Manager can schedule a report on the Report Scheduler by using the "schedule" command line option on the Manager I Terminal (for example, **list configuration all [schedule].** The system then verifies the command to be scheduled, and the Report Scheduler screen is displayed as shown in Figure 3-28. By setting the Print Interval field (described below) to "scheduled" or "deferred", the additional fields shown in Figures 3-29 and 3-30 appear. If the Report Scheduler is full, the error message "Maximum number of reports scheduled; cannot schedule new report" is displayed on the Manager I terminal.

When using the "schedule" command line option to schedule a report, the Report Scheduler form contains the following fields.

- **Job Id:** (display only) Shows the report identification number (1 through 50), provided by the system.
- **Command:** (display only) Shows the command to be executed.
- **Print interval:** This field has three options: immediate, scheduled and deferred. Figure 3-28 shows a Report Scheduler screen with the "immediate" option. Figure 3-29 shows a Report Scheduler screen with the "scheduled" option. Figure 3-30 shows a Report Scheduler screen with the "deferred" option.

  The "scheduled" option is used to schedule a report to be printed at a later time.

  The "deferred" option is used to schedule a report to be printed once at a later time.

  The "immediate" option is used if the System Manager would like to print the report immediately. If the printer link is not up, the scheduler will attempt to bring up the link and print the report. If the link is already up, the scheduler will mark the report for printing during the current 15-minute time interval. If the printer link cannot be established, the report will be placed at the head of the queue and will be printed the next time the link is established.

  If the printer link fails before the report has completed printing, no attempt will be made to print the report when the link is finally established.

The immediate option allows one-shot printing of reports.

- **Days of Week:** If the "scheduled" or "deferred" option of the Print Interval field is chosen, the System Manager will be prompted for the days of the week and time of day for the report to be printed. A maximum of one day of the week may be selected for "deferred" reports.
- **Print time:** Reports may be scheduled at 15-minute intervals within a given hour (0,15,30,45).

```
list configuration all
                                                        Page 1 of 1
                              REPORT SCHEDULER



           Job Id: 10                Job Status: none

          Command: list configuration all

   Print Interval: immediate
```

Figure 3-28. Report Scheduler Screen Form (With Immediate Print Interval)

```
list configuration all
                                                            Page 1 of 1
                              REPORT SCHEDULER



              Job Id:  10                      Job Status:  none

             Command:  list configuration all

      Print Interval:  scheduled

          Print Time:  21:15

      Sun: n  Mon: y  Tue: n  Wed: y  Thu: n  Fri: y  sat: n
```

Figure 3-29. Report Scheduler Screen Form (With Scheduled Print Interval)

```
list configuration all
                                                            Page 1 of 1
                              REPORT SCHEDULER



              Job Id: 10                       Job Status:  none


             Command: list configuration all

      Print  Interval: deferred

          Print Time: 21:15

      Sun: n  Mon: y  Tue: n  Wed: n  Thu: n  Fri: n  sat: n
```

Figure 3-30. Report Scheduler Screen Form (With Deferred Print Interval)

Changing Scheduled Reports

The System Manager may change a scheduled report using the **change report-scheduler** command. When this command is entered, the Report Scheduler screen is displayed, as shown in Figure 3-31. This screen is similar to the Report Scheduler screen displayed with the "schedule" command line option, but has an additional field. This is the Job Status field which shows one of the following:

- **print-next**—Indicates that the report is scheduled to be printed in the current time interval.
- **printing**—Indicates that the report is currently being printed.
- **printed** —Means that the report has been successfully printed.
- **waiting**—Means that the report is not scheduled for any activity during the current 15-minute time interval.

If the Print Interval of a report is changed so that its scheduled time now falls inside the current 15-minute time interval, the report will not be printed in the interval. Instead, the report will be printed during its next scheduled time interval.

If a report is scheduled for a given time period, other than the current 15-minute interval, and has its Print Interval field changed from "scheduled" to "immediate", the report will be printed immediately.

```
change report-scheduler 10
                                                      Page 1 of 1
                              REPORT  SCHEDULER



             Job Id: 10                Job Status: printed

            Command:  list configuration all

      Print Interval: scheduled

          Print Time: 21:15

      Sun:_n_Mon:_y_Tue:_n_Wed:_y_Thu:_n_Fri:_y_sat:_n_
```

Figure 3-31. Screen Form Used To Change Report Scheduler

Removing Scheduled Reports

The System Manager may remove a scheduled report using the **remove report-scheduler** command.

If the Job Status of the report is "print-next", "printed", or "waiting" (that is, not being printed), it will be removed immediately. If the report is being printed ("printing" state), not only will the command be removed, but the printer link will be torn down as well. The link will be brought up during the next 15-minute time interval or if an "immediate" report is scheduled, whichever comes first.

Listing Scheduled Reports

The System Manager may display a list of the scheduled reports on the Manager I terminal, or its printer, using the **list report-scheduler** command. A sample list is shown in Figure 3-32. The reports are displayed in the order they will be printed. The id of the user who scheduled the command is also displayed. This field is used to identify who scheduled the command.

Reports that are scheduled for immediate execution will be listed at the top of the queue.

Reports with the same scheduled printing time are displayed according to their order in the report scheduler queue. The first report in the queue will be displayed first.

The Job status field indicates the status of a report. There are four possible values; waiting, print-next, printing, and printed.

The System Manager may send the output of the **list report-scheduler** command to the printer attached to the Manager I terminal by using the "print" option.

Establishing the Printer Link

The system will attempt to bring up the link to the Report Scheduler printer at the beginning of each 15-minute time interval, provided there are reports to be printed, or when an immediate report is to be printed. After all reports for which the link was brought up have been printed, the system will tear down the link to preserve system resources.

```
                        REPORT  SCHEDULER

Job Id       Days  (smtwtfs)          Time    User    Status     Type
    Command

  4           immediate               18:53   BCMS    printing    immediate
     list bcms split 7 time hh:mm

  2               nynynyn             19:00   BCMS    waiting     scheduled
     list bcms split 2 time hh:mm

  7               nynnnnn             19:15  BCMS     waiting     deferred
     list bcms system

  23              nyyyyyn             22:45  BCMS     waiting     scheduled
     list bcms agent 4000 day         09/15

    Note:   hh:mm is used to indicate field size but is not displayed.
```

Figure 3-32. Screen Form Used To List Report/Scheduler Information

Considerations

With the Report Scheduler and System Printer, the System Manager can schedule most "list", "test", and "display" administration commands to be printed at various times on an asynchronous printer. By scheduling these reports to print automatically at the desired times, the System Manager saves valuable time which can be used to perform other administrative duties.

The System Manager can schedule a maximum of 50 individual reports. The system has a single asynchronous printer connection dedicated for use by the report scheduler. Other printers in the system include those connected to the Manager I terminal, the Station Message Detail Recording (SMDR) printer, and the Journal Printer. These are not used by the Report Scheduler feature.

Reports scheduled for the same time and day are printed according to their order in the Report Scheduler queue. The first report in the queue will be printed first.

In order to present the least possible impact on system performance, it is recommended that reports be scheduled at off-peak hours and staggered so that they are not all scheduled to be printed at the same time.

Reports that are added to the scheduler queue, and are scheduled to be printed during the current time interval, will not be printed until the next scheduled time.

If a system error is encountered while trying to print a scheduled report, the error will be printed on the report, just as it would be displayed for the same command on the Manager I terminal screen.

Interactions

There is only one processor board EIA port available for asynchronous output. The port cannot be administered for both SMDR and the Report Scheduler System Printer on the System-Parameter Feature form. Also, the Report Scheduler System Printer and the Journal Printer used with hospitality features cannot share the same printer.

Administration

The System Manager may schedule, list, change, and remove the desired reports as previously described in this description. Before these procedures can be done, however, the System Manager must supply printer information on Page 4 of the System-Parameters Features form by entering the following information:

- **Printer Extension:** "EIA" for the EIA port or a valid data module extension if the EIA port is not to be used.

  The System Manager must specify the printer link by selecting either the EIA port, if available, or a data-module extension. If the data-module extension is chosen, the System Manager must have previously administered the extension using the **add data-module** command.

- **EIA Device Bit Rate:** The speed of the printer (1200, 2400, 4800 or 9600 baud). Default is 1200.

- **Lines per Page:** The number of printed lines per page (24 to 132). Default is 60.

Hardware and Software Requirements

The asynchronous printer can be connected to the switch using either of the following methods:

- The printer can be connected directly to the EIA port on the switch's processor board. In this case the appropriate cable is required.
- The printer can be connected to the switch with a Modular Processor Data Module (MPDM) and a port on a TN754 or TN784 (G1) Digital Line circuit pack.
- The printer can be connected to the switch with an Asynchronous Data Unit (ADU) and a port on a TN726 Data Line circuit pack.

There is a single EIA port in DEFINITY Generic 1. There may be contention between the SMDR and the Report Scheduler feature for use of this port. If the Report Scheduler feature is using the EIA port and you would like to enable the SMDR feature, it is recommended that you disconnect the system printer from the EIA port and use a data module for its connection, freeing the port for use by SMDR.

The EIA port on the processor board is not available in a duplicated system. Therefore, the data connection to the Report Scheduler System Printer must interface through a data module. When a processor switch occurs, the link to the printer is dropped and re-established.

An AT&T 475 or AT&T 572, which uses a serial interface, or compatible printer, may be used as the System Printer. A Personal Computer (PC) may be connected to the system printer port for collection of data; however, a serial interface on the PC must be provided for the connection.

Report Scheduler and System Printer software is required.

Restriction—Controlled

Description

Allows an attendant or voice terminal user with console permission (V3) to activate and deactivate the following restrictions for an individual voice terminal or a group of voice terminals:

- Outward—The voice terminal(s) cannot be used for placing calls to the public network. Such call attempts receive intercept tone.
- Total—The voice terminal(s) cannot be used for placing or receiving calls. Direct Inward Dialing calls are routed to the attendant or a recorded announcement. All other calls receive intercept tone.
- Station-to-Station (V3)—The voice terminal cannot receive or place station-to-station calls. Such call attempts receive intercept treatment.
- Termination (V3)—The voice terminal cannot receive any calls. Incoming calls are routed to the attendant, are redirected via Call Coverage, or receive intercept treatment.

To activate the desired Controlled Restriction, the attendant or voice terminal user with console permission (V3 or G1) dials the feature access code for either the extension or the group, followed by either 1 for Outward, 2 for Total, 3 for Termination, or 4 for Station-to-Station, and then dials the voice terminal extension number (Attendant Control—Extension) or the Class of Restriction (COR) for a group of voice terminals (Attendant Control—COR).

Considerations

Controlled Restriction gives the attendant control of outward, total, station-to-station, and termination restriction for voice terminals or groups of voice terminals.

All voice terminals with the same COR are affected by a group restriction.

Interactions

The following features interact with the Controlled Restriction feature.

- Call Coverage

  Controlled Restrictions are not checked for covering users.

- Call Forwarding

  Controlled Restrictions for the forwareded-to extension are only checked when Call Forwarding All Calls is activated. Once calls are redirected, the forwarded-to extension's restrictions are not checked.

- Class of Restriction

  Both Class of Restriction and Controlled Restrictions are checked when a call is authorized.

- Uniform Dial Plan

  Calls dialed through the Uniform Dial Plan are not restricted by Outward Restriction.

Administration

Controlled Restriction is administered on a per-system basis by the System Manager. The following items require administration:

- Controlled Restriction Activation and Deactivation access codes. Separate access codes are needed for each type of controlled restriction.

- Type of Intercept Treatment for each type of controlled restriction.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction—Miscellaneous Terminal

Description

Restricts callers at specified voice terminals from accessing certain other voice terminals.

Miscellaneous Terminal Restrictions can be used whenever it is undesirable for users at certain voice terminals to access other specific voice terminals.

Considerations

The Miscellaneous Terminal Restriction is controlled by the Class of Restriction (COR) assigned to the calling voice terminal user and to the voice terminal being called. Any COR can be administered to allow or deny access to any other COR. Restricted calls are routed to intercept tone.

Interactions

A voice terminal user with authorization to access an Abbreviated Dialing Privileged Group Number List or Privileged System Number List can place calls to any number on that list. COR assignments are not checked.

Administration

Miscellaneous Terminal Restriction is administered via the Class of Restriction feature by the System Manger. The only administration required is the permission for each COR to access other CORs.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction—Miscellaneous Trunk

Description

Restricts users at specified voice terminals from accessing certain trunk groups, such as Wide Area Telecommunications Service (WATS).

For a detailed description of Miscellaneous Trunk Restrictions, see the Class of Restriction (COR) description.

Considerations

Miscellaneous Trunk Restriction can be used whenever it is necessary to restrict users at certain voice terminals from accessing specific trunk groups.

The Miscellaneous Trunk Restriction is controlled by the COR assigned to the calling voice terminal user and to the trunk group being accessed. Any COR can be administered to allow or deny access to any other COR. Restricted calls are routed to intercept tone.

Interactions

The following features interact with the Restriction—Miscellaneous Trunk feature.

- Abbreviated Dialing

  A voice terminal user with authorization to access an Abbreviated Dialing Privileged Group Number List or a Privileged System Number List can place calls to any number on that list. COR assignments are not checked.

- Automatic Route Selection

  This feature overrides the Miscellaneous Trunk Restriction feature. Permission or denial of Automatic Route Selection calls is determined by the Facility Restriction Level.

Administration

Miscellaneous Trunk Restriction is administered via the Class of Restriction feature by the System Manger. The only administration required is the permission for each COR to access other CORs.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction—Toll/Code

Description

Restricts users at specified voice terminals from placing public network calls to certain numbers within the local area code, to certain foreign (nonlocal) area codes, and to service codes (such as 411 for directory assistance and 911 for emergency service).

Code Restriction applies when a code-restricted system user accesses a code-restricted trunk group and dials a number string. The system checks the code-restriction tables. If the number is found, the call is permitted. If not, the caller receives intercept tone.

Toll Restriction applies as follows:

- When a toll-restricted system user accesses a trunk group and dials a number string containing a 0 or 1 as the first or second digit, the system checks the Allowed Calls List. If the number is found, the call is permitted. If not, the caller receives intercept tone. (In areas where area codes can also serve as office codes, the system requires the prefix 1 on area code calls to differentiate them from local calls. In this case, local calls with a 0 or 1 as the second digit are not subject to toll restriction.)

- When any system user accesses a toll-restricted trunk group and dials a number string containing a 0 or 1 as the first or second digit, the system checks the Allowed Calls List. If the number is found, the call is permitted. If not, the caller receives intercept tone.

If the system is connected to a central office (CO) that uses a step-by-step switch, all seven of the digits that are normally dialed for a local call may not be required by the CO to route the call. For example, the CO may only require the last five of the normally dialed seven digits. If all seven digits are dialed, the step-by-step switch uses digit absorption to absorb the unneeded digits. Digit absorption can be provided within the system to emulate the absorption at the CO. This prevents users from bypassing code and toll restriction by dialing unneeded digits. For example, assume that the CO absorbs leading 7s before processing a number and that a toll-restricted user wants to call someone in area code 201. The user could dial 77-1-201 plus seven more digits. The Toll Restriction feature would not recognize the call as a toll call and the CO would route the call. With digit absorption, the 77 is absorbed by the system before Toll Restriction is used. Thus, the call would be denied, as intended. Up to five digit absorption lists can be assigned.

Considerations

Toll and/or Code Restriction is used whenever it is necessary to restrict users at certain voice and data terminals from making calls to certain COs, area codes, and/or service codes.

The Allowed Calls List can include up to ten CO codes (that is, the first three digits of a 7-digit number), area codes, and/or service codes that toll-restricted system users will be permitted to access.

Two code-restriction tables are established. One table lists certain CO codes within the local area code and the other lists certain foreign area codes and service codes. Code-restricted users are permitted to access the codes listed in the code-restriction tables.

If a caller is toll-restricted and a trunk group is code-restricted, or vice versa, the toll restriction applies.

Interactions

The Automatic Route Selection feature overrides Toll and Code Restriction. Permission or denial of Automatic Route Selection calls is determined by the Facility Restriction Level.

Administration

Toll or Code Restriction (but not both) is administered by the System Manager to each foreign exchange and CO trunk on a trunk group basis. A trunk group may also be administered to have neither toll nor code restriction by leaving the Restriction field blank.

Toll or Code Restriction (but not both) is administered by the System Manager to the following by the Class of Restriction:

- Attendant consoles as a group
- Incoming tie trunks on a trunk group basis
- Voice terminals on a per-terminal basis
- Data Modules on a per-module basis.

Other items that can be administered are as follows:

- Allowed Calls List containing up to ten codes that toll-restricted users will be permitted to access
- Code-restriction table listing central office codes within the local area code that code-restricted users will be permitted to access
- Code-restriction table listing foreign area codes and service codes that code-restricted users will be permitted to access
- Digit absorption lists containing absorption treatment of each digit 0 through 9.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction—Voice Terminal—Inward

Description

Restricts callers at specified voice terminals from receiving public network, attendant-originated, and attendant-extended calls. A denied call is routed to intercept tone, a recorded announcement, or the attendant.

Calls can redirect to an inward-restricted voice terminal. The Class of Restriction (COR) of the originally called extension number is the only one checked.

Considerations

Inward Restriction is used whenever it is necessary that users at certain voice terminals receive only internal calls from other voice terminals.

Interactions

The following features interact with the Restriction—Voice Terminal—Inward feature.

- Controlled Restriction

  When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

- Night Service

  The Trunk Answer From Any Station and Night Station Service features, if assigned to an inward-restricted voice terminal, override the Inward Restriction.

- Tie Trunk Access

  Incoming dial repeating tie trunk calls can be completed directly to an inward-restricted extension number. However, such calls cannot be extended by an attendant to an inward-restricted voice terminal.

- Transfer

  Incoming trunk calls can be transferred from an unrestricted extension number to an inward-restricted extension number.

Inward Restriction is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction—Voice Terminal—Manual Terminating Line

Description

Restricts callers at specified voice terminals from receiving calls other than those from an attendant. All other calls are routed to intercept tone, a recorded announcement, or an attendant. The voice terminal user can originate calls and activate features.

Calls can redirect to a voice terminal assigned this feature. The Class of Restriction (COR) of the originally called extension number is the only one checked.

Considerations

Manual Terminating Line Restriction is used whenever it is necessary to have users at certain voice terminals receive only calls from an attendant.

Interactions

The following features interact with the Restriction—Voice Terminal—Manual Terminating Line feature.

- Controlled Restriction

  When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

- Night Service

  The Trunk Answer From Any Station or Night Station Service feature, if assigned to a restricted voice terminal, overrides Manual Terminating Line Restriction.

Administration

The Manual Terminating Line Restriction feature is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction—Voice Terminal—Origination

Description

Restricts callers at specified voice terminals from originating calls. Voice terminal users can receive calls.

If a voice terminal user attempts to place a call, intercept tone is received. A voice terminal can, however, activate certain features by dialing the assigned feature access codes.

Considerations

Origination Restriction is used whenever a voice terminal is to be used only for answering incoming calls.

Interactions

When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

Administration

The Origination Restriction feature is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction—Voice Terminal—Outward

Description

Prevents specified voice terminal users from placing calls to the public network. Calls can be placed to other voice terminal users, to the attendant, and over tie trunks.

Considerations

Outward Restriction is used whenever it is desired that a voice terminal make only internal calls.

The attendant or an unrestricted voice terminal user can extend a call to an outside number for the outward-restricted voice terminal user.

Interactions

When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

Administration

The Outward Restriction feature is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction—Voice Terminal—Termination

Description

Restricts voice terminal users on specified extension numbers from receiving any calls. The restricted users can, however, originate calls.

Considerations

Termination Restriction is used whenever a voice terminal is to be used only for making calls.

Interactions

When the system authorizes a call, Controlled Restrictions are checked as well as those restrictions assigned by COR.

Administration

The Termination Restriction feature is administered by the System Manager to voice terminals by the COR.

Hardware and Software Requirements

No additional hardware or software is required.

Ringback Queuing

Description

Places outgoing calls in an ordered queue (first-in, first-out) when all trunks are busy. The voice terminal user is automatically called back when a trunk becomes available. The voice terminal receives a distinctive 3-burst alerting signal (Priority Calling) when called back.

With V1, when an all-trunks-busy condition exists within a trunk group, a multi-appearance voice terminal user receives reorder (fast-busy) tone after dialing is complete. To access Ringback Queuing, the user presses the Automatic Callback button. The system acknowledges the availability of the queue by returning a confirmation tone (three short bursts of tone).

With V2, V3, or G1, if a multi-appearance voice terminal user has an idle Automatic Callback button and tries to access an all-trunks-busy trunk group, Ringback Queuing is automatically activated, the lamp associated with the Automatic Callback button lights, and confirmation tone is heard. The multi-appearance voice terminal user must also be authorized to make the call, Ringback Queuing must be allowed on the trunk, and the trunk queue must not be full.

Ringback Queuing is automatic for a single-line voice terminal. After dialing is complete, the user hears confirmation tone if the queue is available. No action is required by the voice terminal user. The user hangs up and waits for callback.

The callback call is automatically placed to the terminal when a trunk becomes available. When the user answers the callback call, the original call automatically continues. Redialing is not required.

Queuing can be specified for any non-Distributed Communications System (DCS) outgoing only trunk group, or for the outgoing direction of a non-DCS 2-way trunk group.

Considerations

With Ringback Queuing, users do not have to keep trying to access a trunk group when all trunks in the group are busy. This feature provides for the caller of a busy trunk group to automatically be called back when a trunk becomes available.

Queuing can reduce the number of trunks required.

The system allows a maximum of 100 (V1) or 120 (V2, V3, or G1) calls in queue for all the trunk groups in the system.

A single-line voice terminal can have only one call waiting at a time; therefore, Ringback Queuing is denied to these voice terminals if a call is already waiting.

A multi-appearance voice terminal can have one callback call associated with each Automatic Callback button assigned to the terminal.

A queue request will be canceled for the following reasons:

- A trunk is not available within 30 minutes.
- The voice terminal user does not answer the callback call within the administered interval (2 to 9 ringing cycles).
- The voice terminal is busy when the callback call is attempted.
- The voice terminal user dials the Ringback Queuing cancellation code or presses the Automatic Callback button associated with the queued call.

Incoming tie trunk calls cannot queue on an outgoing trunk group. The system does not know the calling number and cannot originate the callback call.

The system checks the busy/idle status of the trunk group just once (immediately after the trunk access code is dialed). If at this time all of the trunks are busy, the call is put into queue, even if a trunk has become available by the time the caller has completed dialing the number. This occasionally results in the caller being called back immediately after receiving confirmation tone and going on-hook.

At times, a trunk appears to be available, but outgoing calls are still placed in a queue. In this case, a trunk is not free, but is being reserved for a previous Automatic Callback request.

Interactions

If Ringback Queuing is provided, Automatic Callback must also be provided. Automatic Callback is administered through the Class of Service.

Ringback Queuing affects the following features:

- Automatic Route Selection (ARS) (V2, V3, or G1)

  If a multi-appearance voice terminal user has an Automatic Callback button, makes an ARS call, and all trunks are busy, Ringback Queuing is activated automatically.

- Bridged Call Appearance

  Ringback Queuing is not provided on calls originated from a bridged call appearance.

- Call Coverage

  Callback calls do not redirect even if Send All Calls is activated.

- Call Forwarding All Calls

  Callback calls are not forwarded.

- Call Pickup

  Callback calls cannot be picked up.

- Conference or Transfer

  A single-line voice terminal cannot receive a callback call while it has a call on hold and can have only one active call at a time.

- Remote Access

  A callback call cannot be made to a Remote Access user because the system does not know the calling number.

Administration

Ringback Queuing is administered by the System Manager. The following items require administration:

- Callback call no-answer time-out (from 2 to 9 ringing cycles)
- Automatic Callback button (per multi-appearance voice terminal)
- Ringback Queuing cancellation code
- Queue length (per outgoing trunk group).

Hardware and Software Requirements

No additional hardware or software is required.

Ringer Cutoff

Description

Allows the user of a multi-appearance voice terminal to turn certain audible ringing signals on and off. Visual alerting is not affected by this feature.

When this feature is enabled, only Priority ring (3-burst ringing), Intercom ring, and Manual Signaling will ring at the voice terminal. One-burst, two-burst, and redirection notification will not ring. When this feature is disabled, the voice terminal will have normal ringing.

The following table summarizes what call types are and are not affected by the activation of Ringer Cutoff:

Call Type	Ring Type	Will The Voice Terminal Ring If Ringer Cutoff Is Active?
Voice Terminal to Voice Terminal	1-burst	no
Attendant to Voice Terminal	2-burst	no
Internal Tie to Voice Terminal	1-burst	no
APLT Trunk to Voice Terminal	1-burst	no
Redirect Notification	ring-ping	see Note
Trunk to Voice Terminal	2-burst	no
Priority Call to Voice Terminal	3-burst	yes
Intercom Call to Voice Terminal	Intercom	yes
Manual Signaling	Manual Signal	yes

Note: With V1, if Ringer Cutoff and Redirect Notification are both active, the voice terminal will not ring. With V2, V3, and G1, if Ringer Cutoff and Redirect Notification are both active, the ring-ping is still heard, unless the Call Coverage criteria is administered as busy.

There are occasions when a user does not wish to be disturbed by the arrival of incoming calls, and does not want the call to be immediately redirected to coverage. For example, an executive may have a secretary who has bridged call appearances of his or her extension. If the executive does not wish to be disturbed, this feature can be used to allow the secretary a chance to answer the incoming call before it redirects to coverage. The bridging user (the secretary) is not affected by the executive's activation of Ringer Cutoff.

If a bridging user has Bridged Call Alerting administered for his or her voice terminal (and does not have Ringer Cutoff administered), the bridging user will receive ringing for the principal's call.

If a primary extension and all other users with bridged appearances of the primary extension activate Ringer Cutoff, an incoming call will silently alert all of those voice terminals before the call redirects to coverage.

To activate Ringer Cutoff, the user pushes the voice terminal's Ringer-Cutoff button. The associated green status lamp then lights. This lamp remains lighted until the feature is deactivated. If the feature is activated while the voice terminal is ringing with a 1-burst ring, 2-burst ring, or redirection notification, the ringer is silenced.

To deactivate Ringer Cutoff, the user pushes the active Ringer-Cutoff button on his/her voice terminal. The green status lamp associated with the button then goes dark. If the selected call is in any ringing state, the ringer returns to the proper audible ring. If there are no calls ringing at the voice terminal, the ringer remains silent.

Considerations

The Ringer Cutoff feature allows a user to turn off audible ringing on his or her voice terminal.

Each multi-appearance voice terminal user may have one Ringer-Cutoff button on his or her voice terminal.

Interactions

The following features interact with the Ringer Cutoff feature.

- Automatic Callback

  Even if the Ringer Cutoff feature has been activated, the Automatic Callback call will return to the user's voice terminal with the normal 3-burst ring.

- Bridging

  A bridging user is not affected by a primary extension's activation of Ringer Cutoff; nor is the primary extension affected by the activation of Ringer Cutoff by bridging user.

- Call Forwardinging All Calls

  If Ringer Cutoff and Call Forwarding All Calls are active, the user will not receive redirect notification, even if the "redirection notification" is administered for that extension.

- Distinctive Ringing

  Activation of Ringer Cutoff only turns off the ringing of one-burst ring, two-burst ring and redirection notofication. Intercom ringing, Priority ringing, and Manual Signaling are not turned off by the feature.

- Intercom (Automatic and Dial)

  Even if the Ringer Cutoff feature has been activated, Intercom calls will still ring the user's voice terminal.

- Manual Signaling

  Even if the Ringer Cutoff feature has been activated, Manual Signaling will still ring the user's voice terminal.

- Ringback Queuing

  Even if the Ringer Cutoff feature has been activated, the return call for Ringback Queuing will still ring the user's voice terminal.

- Priority Calling

  Even if the Ringer Cutoff feature has been activated, Priority Calls will still ring at the user's voice terminal.

- Send All Calls

  When Ringer Cutoff and Send All Calls are both active, the user will not receive redirect notification, even if the "redirection notification" is administered for that extension.

Administration

The Ringer Cutoff feature is administered on a per-voice terminal basis by the System Manager. The only administration required is a Ringer-Cutoff button, which can be assigned to any multi-appearance voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

Rotary Dialing (V2, V3, or G1)

Description

Allows rotary dialing voice terminals to be used with a System 75.

When a number is dialed at a rotary dialing voice terminal, the voice terminal outpulses at a rate of 10 pulses per second. Each digit dialed sends out the corresponding number of pulses. For example, dialing a 7 results in 7 pulses being sent from the voice terminal. Version 2, Version 3, or DEFINITY Generic 1 software recognizes that the voice terminal is rotary when the user lifts the handset, and expects to receive dial pulses instead of tones.

Considerations

With Rotary Dialing, existing rotary dialing voice terminals can be used in situations where very simple call processing functions are required.

Any functions requiring the * and # symbols cannot be performed on a rotary dialing voice terminal.

Interactions

None.

Administration

Rotary Dialing voice terminals must be administered as a 500 set.

Hardware and Software Requirements

No additional hardware is required. V2, V3, or G1 software is required.

Send All Calls

Description

Allows users to temporarily direct all incoming calls to coverage regardless of the assigned Call Coverage redirection criteria. Send All Calls also allows covering users to temporarily remove their voice terminals from the coverage path.

Send All Calls is activated by pressing the Send All Calls button or by dialing the Send All Calls access code. It is deactivated by pressing the button a second time or by dialing the deactivate access code.

Details of how Send All Calls is used in conjunction with Call Coverage are given in the Call Coverage feature description elsewhere in this chapter.

Considerations

Send All Calls gives a user the option to have all incoming calls sent directly to coverage. This is useful when a user needs to be away from his or her desk temporarily.

Interactions

Send All Calls is used only in conjunction with the Call Coverage feature.

Administration

Send All Calls is administered by the System Manager. The following items require administration:

- Send All Calls button (per voice terminal)
- Activate and Deactivate access codes for Send All Calls (per system)
- Send All Calls coverage criteria (per coverage path).

Hardware and Software Requirements

No additional hardware or software is required.

Senderized Operation

Description

Reduces the time necessary to place calls to distant locations equipped to receive touch-tone signals and allows end-to-end signaling to remote computer equipment.

The number dialed and end-to-end signaling digits from voice terminals and trunks are detected by the system and regenerated for transmission over outgoing trunks. The distant end associated with the trunk must be equipped to receive touch-tone signals.

Considerations

This feature provides quicker service to remote touch-tone receiving facilities.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Service Observing (V3 or G1)

Description

Allows a specified user, such as a supervisor, to observe a call that involves other users while the call is in progress. While observing a call, the specified user can toggle between a listen-only and a listen/talk connection to the call.

In this feature description, Service Observing is described as it pertains to the Automatic Call Distribution (ACD) feature. However, Service Observing can also be used in non-ACD applications. If this feature is being used for non-ACD applications, this description remains the same but requires the following translations:

- The split supervisor is the user who is observing the call.
- The agent is the user whose call is being observed.

Service Observing can be activated by a split supervisor or any other user that has a Service Observe button. Service Observing cannot be activated by feature access code. To activate Service Observing, the split supervisor presses the Service Observe button followed by the extension number of the agent whose calls are to be observed. To deactivate Service Observing, the split supervisor can either hang up, select another call appearance, or press the Disconnect or Release button.

When Service Observing is activated, the split supervisor is in the listen-only mode. Each additional press of the Service Observe button causes the split supervisor to toggle between a listen-only and a listen/talk connection to the call. The split supervisor can observe consecutive calls without having to re-activate Service Observing. In other words, as long as a split supervisor has activated Service Observing for a specific agent, the split supervisor can observe that agent's calls until Service Observing is deactivated.

An optional warning tone can be administered (on a per-system basis) to let the agent and the calling party know that the split supervisor is observing the call. The warning tone is a 440-Hz tone. A 2-second burst of this tone is heard before the split supervisor is connected to the call. A half-second burst of this tone is heard every 12 seconds while a call is being observed. The warning tone is heard by all parties on the observed call.

It is possible for a split supervisor to activate Service Observing for an agent's calls, even though the agent is not active on a call. In this case the split supervisor enters the "waiting" mode until the agent receives an ACD call. When the agent receives an ACD call, the split supervisor is bridged onto the call.

If an agent makes an outgoing trunk call, and is being observed by the supervisor, Service Observing begins when dialing is completed. For central office (CO) trunks with answer supervision, dialing is considered completed when answer supervision is returned. For CO trunks without answer supervision, dialing is considered completed when answer supervision timeout occurs (see Note).

Note: The use of Service Observing features may be subject to federal, state, or local laws, rules or regulations and may be prohibited pursuant to the laws, rules, or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable law, rules and regulations before using these features.

Considerations

With Service Observing, a split supervisor can monitor a split agent while the agent is active on an ACD call. This allows the supervisor to ensure that calls are being handled properly. The supervisor can also assist the agent with the call if necessary.

Although an agent can be a member of more than one split, an agent can only be observed by one supervisor at a time.

Each split supervisor can have only one Service Observe button.

If the agent whose calls are to be observed has a Class of Restriction that does not permit Service Observing, the split supervisor cannot observe that user's calls.

The following types of calls cannot be observed by the split supervisor:

- A call with a 6-party conference
- A call being service observed by another split supervisor
- A call that is being Busy Verified
- A call that has Data Privacy, Data Restriction, or Privacy—Manual Exclusion activated.

If two agents with different supervisors are being service observed, and one agent calls the other, the originator's supervisor observes the call, and the other supervisor is in the waiting mode.

An attendant cannot be a service observer.

While service observing someone, the only buttons allowed to be pressed are as follows:

Call Appearance	Bridged Appearance
Position Busy	Auxiliary Work
Auto-ckt Assure	Queue Status (NQC, OQT, AQC, and AQT)
Release (ACD)	System Night Service
Service Observing	

Interactions

The following features interact with the Service Observing feature.

- Busy Verification of Terminals and Trunks

  A split supervisor cannot service observe an agent's call that is being bridged onto by busy verification. Also, an agent's call that is being bridged onto by service observing cannot be busy verified.

- Call Coverage

  A split supervisor cannot service observe a call that has been answered by a covering user until the called agent bridges onto the call.

- Call Park

  A split supervisor cannot park a call while service observing the call.

- Call Pickup

  A split supervisor cannot service observe a call that has been answered by a member of a pickup group until the called agent bridges onto the call.

- Call Waiting

  A call cannot wait on a single-line voice terminal that is being service observed.

- Conference

  The split supervisor cannot use this feature when Service Observing is activated.

  If an agent conferences a call while being observed and the number of parties in the call is less than six, the split supervisor is put into the waiting mode. The supervisor is bridged onto any call the agent becomes active on before the conference is complete. When the conference is complete, the supervisor is again bridged onto that call.

  If an agent conferences a call while being observed and the number of parties in the call is six, including the split supervisor, the conference is denied.

- Hold

  The split supervisor cannot use this feature when Service Observing is activated.

  If an agent places a call on hold while being observed, the split supervisor is put into the waiting mode.

- Leave Word Calling

  Leave Word Calling cannot be used by any party on a call that is being service observed.

- Privacy—Manual Exclusion

  A split supervisor cannot service observe an agent that has activated Privacy—Manual Exclusion.

- Transfer

  The split supervisor cannot use this feature when Service Observing is activated.

  If an agent transfers a call while being observed, the split supervisor is put into the waiting mode. The supervisor is bridged onto any call that the agent becomes active on before the transfer is complete.

Administration

Service Observing is administered by the System Manager. The following items require administration:

- Optional Warning Tone (per system)
- Service Observe Button (per voice terminal)
- Class of Restriction (assigned on a per-agent basis to determine whether or not the agent can be service observed).
- Service Observing and/or Automatic Call Distribution must be administered as a feature.

Hardware and Software Requirements

No additional hardware or software is required.

Single-Digit Dialing and Mixed Station Numbering (V3 or G1)

Description

Allows easy access to internal hotel/motel services and provides the capability to associate room numbers with guest room voice terminals.

The following dial plan types are provided:

- Single-Digit Dialing
- Prefixed Extensions
- Mixed Numbering.

Single-Digit Dialing

A single-digit extension number can be assigned to internal hotel/motel services such as room service. These single-digit extension numbers can be assigned to an individual voice terminal or to a group of voice terminals used, for example, to service the front desk.

Prefixed Extensions

A prefixed extension is made up of a prefix (or first digit) and an extension number with up to five digits. The prefix identifies the call type and specifies the number of digits that will follow. System 75 collects the dialed digits, removes the prefix digit, and uses the extension number for any further processing.

Assume that the following dial plan has been administered for a hotel/motel system:

First Digit	Length					
	1	2	3	4	5	6
0	ATT					
1		TAC				
2	EXT					
3	EXT					
4	EXT					
5			EXT			
6				PEXT		
7					PEXT	
8	TAC					
9	TAC					
*		FAC				
#		FAC				

This example dial plan will allow the following call types:

- Single-digit access to the hotel/motel attendant (0)
- Ten trunk access codes (TACs) beginning with the digit 1 (10 through 19)
- Single-digit access to three hotel/motel services using the digits 2, 3, and 4
- Nonprefixed access to as many as 100 hotel/motel staff extensions (500 through 599)
- Room extensions for as many as 100 floors
  - Access to floors 1 through 9 (prefix digit 6 + [100 through 999])
  - Access to floors 10 through 99 (prefix digit 7 + [1000 through 9999])
- Toll calling access by dialing TAC 8
- Toll calling access by dialing TAC 9
- Two-digit feature access codes (FACs) beginning with * and # and followed by a second digit.

The system identifies a Prefixed Extension number through translation processing. Without the prefix digit, the same group of digits could belong to any call type. In the preceding dial plan example, the digits 71234 will be identified as extension 1234 preceded by the prefix 7. If 1234 is dialed, the system will interpret it as the 2-digit trunk access code 12 because a 4-digit extension number beginning with a 1 is not defined.

Mixed Numbering

A dial plan with mixed numbering has the following characteristics:

- Extension numbers can have from one to five digits and can begin with any digit from 1 to 9 (the digit 0 will define access to the hotel/motel attendant).
- The first digit, in combination with the number of digits dialed, defines the call type that corresponds to the dialed numbers.

The flexibility of mixed numbers, administrative staff extension numbers, service extension numbers (Single-Digit Dialing), TACs, and FACs may have common leading digits. To differentiate between two numbers with the same leading digit but with different lengths, the system applies a 3- to 4-second interdigit time-out.

Assume that the following dial plan has been administered:

First Digit	Length 1	2	3	4	5	6
0	ATT					
1	EXT	EXT	EXT	EXT+		
2	EXT	EXT	EXT	EXT+		
3	EXT	EXT	EXT	EXT+		
4	EXT	EXT	EXT	EXT+		
5	EXT	EXT	EXT	EXT+		
6	EXT	EXT	EXT	EXT+		
7	EXT	EXT	EXT	EXT+		
8	TAC					
9	TAC					
*		FAC				
#		FAC				

+ Time-outs are applied after the first, second, and third digits.

This dial plan example will allow the following dial access:

- Single-digit access to the hotel/motel attendant (0)
- Single-digit access to seven hotel services (extensions 1 through 7)
- Two-digit access to 70 hotel/motel services (extensions 10 through 70)
- Guest room extensions for floors 1 through 7 (extensions 100 through 799)
- Toll calling access by dialing TAC 8
- Toll calling access by dialing TAC 9

- Two-digit FACs by dialing * or # plus another digit.

Using the preceding dial plan example, the digit 2 can be assigned as the extension number for a hotel/motel service, 22 as an extension number for an administration staff member, and 222 as the extension number for guest room 222. Interdigit time-outs will be required after the first and second digits.

Time-out intervals can be canceled if the user dials # after dialing all required digits.

Considerations

Single-Digit Dialing allows easy access to hotel/motel services.

Mixed Station Numbering allows guest room numbers and room extensions to be the same. Dialing time is a little longer, however, because of the required interdigit time-out interval.

Prefixed extensions eliminate the need for interdigit time-outs, but require dialing an extra digit.

Prefixed extensions greater than five digits in length (including the prefix) cannot be assigned to intercom lists.

A TAC and an extension number can only share a first digit if the extension number is shorter than the TAC.

Although extensions with the same first digit can have different lengths, data channel extensions should have the maximum number of digits possible in order to avoid netcon time-out problems.

Extension numbers and FACs can share the same first digit with the extension number being longer, but these extension numbers will only work within the switch. They will not work as remote Uniform Dial Plan extensions.

Interactions

The following features interact with the Single-Dgit Dialing and Mixed Station Numbering feature.

- Attendant Display and Voice Terminal Display

  If prefixed extensions are used in the system's dial plan, the prefix is not displayed when the extension is displayed. With V3, the Return Call button cannot be used to dial prefixed extensions, because this button causes the system to dial the displayed number, which does not contain the entire extension. With G1, the Return Call button can be used to dial prefixed extensions, because the G1 system will dial the prefix, even though it is not displayed.

- Property Management

  If Prefixed Extensions are assigned in the system, the prefix digit is removed before messages containing the extension number are sent to the Property Management System (PMS).

  Five-digit extensions cannot be exchanged with a PMS until modifications are made to the PMS interface.

- Uniform Dialing Plan (UDP)

  The following limitations apply to a Distributed Communications System (DCS) environment:

  — Extension numbers that differ in length from the UDP cannot be distributed to other switches.

  — If the first two digits of an extension number correspond to the floor number, floors cannot be serviced by more than one switch.

Administration

The System Manager will define the dial type (extensions, prefixed extensions, TACs, and FACs) when the dial plan is administered for the system.

For each first digit (1 through 9, *, and #), a dial type can be defined for each length up to six digits. The digit 0 will always be defined as access to the hotel/motel operator (attendant).

Hardware and Software Requirements

No additional hardware or software is required.

SMDR Account Code Dialing

Description

Allows certain calls to be associated with a particular project or account number. This is accomplished by dialing specified account codes before making outgoing calls. This information is recorded by the Station Message Detail Recording (SMDR) feature and can be used later for accounting and/or billing purposes.

To associate an account code with a particular call, a user first dials an SMDR access code. The user then dials the desired account code, which can contain up to 5 digits (V1) or 15 digits (V2, V3, or G1). The user then dials the desired trunk access code (TAC), Automatic Alternate Routing (AAR) access code, or Automatic Route Selection (ARS) access code.

SMDR Account Code Dialing is optional in Version 1. The user may or may not dial an account code, and the call is not affected. With Version 2, Version 3, or DEFINITY Generic 1, SMDR Account Code Dialing can be optional or mandatory (forced). With Version 2, Version 3, or DEFINITY Generic 1, forced entry of account codes can be assigned for any of the following:

- All Toll Calls

  Toll Calls are defined as those calls which have a 0 or 1 as one of the first two digits of the called number, except service calls (for example, 911 and 411), directory assistance calls, and 800 Service calls.

  This affects all calls made by AAR, ARS, or TAC.

- Toll Calls Made By Users With a Specific Class of Restriction (COR)

  If forced entry of account codes is assigned to a specific COR, any voice terminal assigned that COR must dial an account code before making toll calls.

- All Calls Made on a Trunk Group With a Specific COR

  Any trunk group that is assigned a COR with forced entry of account codes cannot be accessed until an account code is dialed. If a call is being routed via AAR or ARS, account code checking is not done on the trunk group's COR.

With Version 3 or DEFINITY Generic 1, any time an account code is required and the user does not enter an account code, intercept tone is heard. An account code is never required for the following:

- Attendant originated call
- Busy verification of a trunk by an attendant or voice terminal user
- Distributed Communications System (unless required by the trunk group's COR)
- Personal Central Office Lines

- Remote Access Without Barrier Codes
- Trunk-to-Trunk Connections.

Considerations

SMDR Account Code Dialing provides an easy method of allocating the costs of specific calls to the correct project, department, etc. Call information is recorded by the SMDR feature for this purpose.

Account Code length can be up to 5 digits (V1) or 15 digits (V2, V3, or G1).

The validity of the entered account codes cannot be checked by the system.

Interactions

The following features interact with the SMDR Account Code Dialing feature.

Authorization Codes (V3 or G1)

With V3, authorization codes will be recorded on SMDR printouts if the account code length does not exceed five digits.

With G1, authorization codes will be recorded on all SMDR printouts except for the 59-character, Local Storage Unit (LSU), and Integrated Services Digital Network (ISDN) LSU formats, without regard to account code length. Authorization codes will be recorded on SMDR printouts in the LSU and ISDN LSU formats if the account code length does not exceed five digits.

Automatic Alternate Routing (V2, V3, or G1) and Automatic Route Selection (V2, V3, or G1)

If a trunk group is accessed via AAR or ARS, the trunk group's COR is not used to determine if an account code needs to be entered.

Busy Verification of Terminals and Trunks (V2, V3, or G1)

An attendant or voice terminal user is never required to enter an account code when making a busy verification.

Call Forwarding All Calls (V2, V3, or G1)

If a user is required to enter an account code to call a particular destination, the calls cannot be forwarded to that destination.

Last Number Dialed

The SMDR access code and account code dialed are stored as part of the Last Number Dialed. However, some digits may be lost due to the limit on the number of digits stored for this feature.

- Station Message Detail Recording (SMDR)

  SMDR does not record the correct account code if the length of the account code is changed during an active call. For example, if the account code length is 5, a user dials 12345, and the account code length is changed during the call to 2, the SMDR record shows only the first 2 digits (12) of the account code.

Administration

SMDR Account Code Dialing is administered by the System Manager. The following items require administration for forced entry of account codes (V2, V3, or G1):

- Whether or not all toll calls require account code entry (per system)
- Whether or not each individual COR requires account code entry.

Hardware and Software Requirements

No additional hardware is required. Optional SMDR Account Code Dialing software is required.

Station Message Detail Recording

Description

Records detailed call information on all incoming and outgoing calls on specified trunk groups and sends this information to a Station Message Detail Recording (SMDR) output device. Internal calls are not recorded. The SMDR output device provides a detailed printout that can be used by the System Manager to compute call costs, allocate charges, analyze calling patterns, and keep track of unnecessary calls.

Call detail information is provided on trunk groups, loudspeaker paging, and code calling access administered for SMDR. SMDR provides detailed call information for the following types of calls:

- Outgoing Calls—Calls originated by a system voice terminal user or attendant going out on a trunk group.
- Incoming Calls—Calls incoming on a trunk group and terminating at a system voice terminal or attendant console.
- Tandem Calls—Calls incoming on a trunk group and outgoing on another trunk group.
- Ineffective Call Attempt—Calls originated by a system voice terminal user blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy. With G1, this includes the unavailable incoming or outgoing trunks due to trunk usage allocation for Integrated Services Digital Network (ISDN) Call By Call Service Selection trunks and incoming calls rejected by the switch due to Network Specific Facility (NSF) mismatch.
- Calls made using the Loudspeaker Paging Access and Code Calling Access features.

SMDR Data Formats

This part covers the two formats sent to the SMDR output device, call detail and date record formats.

Call Detail Record Format

The call detail record format provides detailed information concerning an incoming or outgoing call. Call detail records are generated during call processing and are sent to the SMDR output device in American Standard Code for Information Interchange (ASCII). SMDR data transmitted to the Applications Processor (AP) (not available with XEV2 or G1) is not included because the system link to the AP uses BX.25 protocol and transmits more than SMDR data.

The following list describes the SMDR data collected for each call and the number of digits in each field. All information is right adjusted in the respective field, unless otherwise indicated. The list describes the data fields associated with an SMDR output device such as TELESEER® Station Message Detail Recorder (SMDR) unit, printer, 94A Local Storage Unit

(LSU), 3B2 Call Detail Recording Utility (CDRU), or customer-provided equipment. As an option the customer may elect to remote this data to a central collection point via private line.

- Access Code Dialed (up to 3 digits [V1, V2, V3, and G1] or 4 digits [G1 24-word formats])

  This field is used only for outgoing calls. This field can be the Automatic Route Selection access code, Automatic Alternate Routing access code, or the access code of a specific trunk group. This field does not exist in the ISDN 18-word record formats (G1).

- Access Code Used (up to 3 digits [V1, V2, and V3] or 4 digits [G1])

  This field is used only for outgoing calls and only when the trunk group used is different from the trunk group access code dialed. This field contains the access code of the actual trunk group that the call was routed over. When the dialed and used access code is the same, this field will be blank (unless one of the ISDN record formats is used. In this case, the field always shows the access code of the used trunk group, even if it is the same as the dialed access code).

- Account Code (up to 5 digits in V1 and 15 digits in V2, V3, or G1)

  This field is optional but can contain a number to associate call information with projects or account numbers. Account Codes must be prefixed with an access code which is either a fixed digit or a series of digits. The access code is administrable on the Feature Access Code form. On outgoing calls, the access code must be dialed before the trunk access code, Automatic Alternate Routing access code, or Automatic Route Selection code. Information in this field is right adjusted. These account codes allow the System Manager to associate calling information with projects or account numbers. The access code is not recorded because it is not part of the SMDR account code.

  Account code dialing is optional in Version 1. The user may or may not dial an account code and the call is not affected. With Version 2, Version 3, or DEFINITY Generic 1, account code dialing can be optional or mandatory (forced). Forced account code entry is set on a per-Class of Restriction (COR) basis. If the trunk group used for a call requires an account code and one is not dialed, the call is denied. Forced account code entry can also be assigned for all toll calls in which the first or second digit is a 0 or 1. Service calls, directory assistance calls, and Wide Area Telecommunications Service (WATS) calls are excluded.

  If the ISDN 18-word format SMDR record is used, a maximum of 12 account code digits may be in the record. If the account code is longer than 12 digits, the least significant digits are dropped.

- Attendant Console (2 digits) (G1) (24-Word Record Only)

  This field contains the attendant console number of the attendant that handled the call in a record that is marked as being attendant handled.

- Authorization Code (7 digits) (V3 or G1)

  This field contains the 4- to 7-digit authorization code used to make the call. With V3, the authorization code is output only if the administered account code length is less than six digits. With G1, on the 94A LSU and 3B2 CDRU 18-word records, the authorization code is output only if the administered account code length is less than six digits in length. With G1, on the 59-character record, the authorization code is never recorded.

- BCC (1 digit) (G1) (24-Word Record Only)

  This field contains the Bearer Capability Class (BCC) for ISDN calls. The BCC is a single digit. Either of the following digits may appear in this field.

  — 0 — Voice Grade Data and Voice

  — 1 — Mode 1 (56 kbps synchronous data)

  — 2 — Mode 2 (less than 19.2 kbps synchronous or asynchronous data)

  — 3 — Mode 3 (64 kbps data for LAPD protocol)

  — 4 — Mode 0 (64 kbps data clear)

- Calling Number (up to 4 digits in V1, 5 digits in V2, V3, or G1, and 10 digits in G1 24-word format)

  This field contains the extension number of the originating voice terminal user or the trunk group access code used for an incoming or tandem call. With the G1 24-word format, this field contains the SID/ANI information on incoming ISDN calls. Information in this field is right adjusted.

- Condition Code (1 character)

  These codes reflect special events relating to the call. The condition codes apply to the printer, TELESEER SMDR unit, 94A LSU, and 3B2 CDRU. These condition codes are listed and defined in Table 3-F on the next page. Condition codes for the 59-character SMDR record are different from the codes in Table 3-F, but can be mapped to these codes as shown below:

CONDITION CODE MAPPING FOR 59-CHARACTER RECORD	
59-Character Condition Code	**Code From Table 3-F**
A	1
D	4
G	7
H	8
I	9
L	C
N	E

Table 3-F. Condition Codes

Condition Codes	Description
1	Identifies an attendant-handled call or an attendant-assisted call (except conference calls).
4	Identifies a call of about 10 hours. On such a call, a call record with this condition code and a duration entry of 9 hours, 59 minutes, and 1 to 9 tenths of a minute is produced after the first period. A similar call record with this condition code is produced after each succeeding 10-hour period. When the call does terminate, a final call record with a different condition code identifying the call type is produced.
7	Identifies calls served by the Automatic Alternate Routing or Automatic Route Selection feature.
8	Identifies calls which have been served on a delayed basis via the Ringback Queuing feature.
9	Identifies an incoming or tandem call.
A	Identifies an outgoing call.
B	Identifies an adjunct-placed outgoing call (G1 Only).
C	Identifies a conference call. A separate call record with this condition code is produced for each incoming or outgoing trunk serving the conference connection. The only voice terminal recorded for a conference call is the conference call originator.
E	Identifies an ineffective call attempt due to facilities not being available, such as all trunks are busy and either no queuing exists or the queue is full on an outgoing call, or the called voice terminal is busy or unassigned for an incoming call attempt. With G1, this also identifies an ISDN Call By Call Service Selection call that is unsuccessful because of an administered trunk usage allocation plan.
F	Identifies an ineffective call attempt due to insufficient calling privileges of the originator (assigned per Facility Restriction Level). With G1, this also identifies ISDN calls rejected by the switch due to an NSF mismatch.

Note: When more than one condition applies to a call, the overriding code is shown in Table 3-G.

When two condition codes apply on the same call, one will override the other. The matrix in Table 3-G defines the overrides. To illustrate how to use this matrix, assume that condition codes 7 and A apply to the same call. The matrix contains ten horizontal rows (1, 4, 7, 8, 9, A, B, C, E, and F) and ten vertical columns (1, 4, 7, 8, 9, A, B, C, E, and F). To find the condition code that overrides, look at the point of intersection between row 7 and column A. In this case, condition code 7 overrides. This can also be found by looking at the point where row A and column 7 intersect.

Table 3-G. Condition Code Override Matrix

	CONDITION CODE									
	1	4	7	8	9	A	B	C	E	F
CONDITION CODE 1	NA	4	1	NA	9	1	NA	C	E	NA
4	4	NA	4	4	4	4	4	4	NA	NA
7	1	4	NA	7	9	7	B	C	E	F
8	NA	4	7	NA	NA	8	NA	C	E	NA
9	9	4	9	NA	NA	NA	NA	C	E	F
A	1	4	7	8	NA	NA	B	C	E	F
B	NA	4	B	NA	NA	B	NA	B	E	F
C	C	4	C	C	C	C	B	NA	NA	NA
E	E	NA	E	E	E	E	E	NA	NA	NA
F	NA	NA	F	NA	F	F	F	NA	NA	NA

- Dialed Number (up to 15 digits)

  This field contains the outside number dialed by a system user.

  The # sign ("E" with the 94A LSU format) may be printed in this field in the following cases for both ARS and Trunk Access Code (IAC) calls.

  - When the user dials a feature access code that starts with a #
  - When the user dials # at the end of digit dialing (for example, for WATS and IDDD calls)
  - When the inter-digit timeout occurs before the answer supervision timeout, even if the user has not dialed the # sign.

- Duration (4 digits)

  All calls are timed. The timing is recorded in hours (0 through 9), minutes (00 through 59), and to the nearest tenth of a minute (0 through 9).

- Facility Restriction Level (FRL) (1 digit)

  FRLs, numbered 0 through 7, are associated with the Automatic Alternate Routing and Automatic Route Selection features and define calling privileges. The information contained in this field is as follows:

  — If the call is an outgoing call and an authorization code (V3 or G1) is not used to make the call, this field contains the originating voice terminal user's FRL.

  — If the call is an outgoing call and an authorization code (V3 or G1) is used to make the call, this field contains the FRL associated with the dialed authorization code.

  — If the call is an incoming or tandem call, this field contains the FRL assigned to the incoming trunk group.

  — If the call is an incoming tandem tie trunk call, this field contains either the FRL assigned to the tandem tie trunk or the Traveling Class Mark (TCM) sent with the tandem tie trunk call, depending on which was used to complete the call. On G1 ISDN calls, this field always contains the TCM, if it was received.

  — If the Feature-Related System Parameters have been administered to have "Disconnect Information in Place of FRL", the following call disconnect data is printed in this field in place of the FRL data:

Data	Meaning
0	Don't know who dropped first
1	We dropped first
2	The CO dropped first
3	Maintenance got the trunk

- Feature Flag (1 digit) (G1 only)

  The digit in this field indicates whether or not the switch has received answer supervision from the network. A "4" in this field indicates that network answer supervision has been provided. Otherwise, a "0" is present in this field.

  Answer Supervision is indicated for non-interworked ISDN calls, E&M trunks (digital or analog), Ground Start trunks with battery reversal, calls placed by adjuncts over any of these trunks (for example, OCM), and calls that received data modem answer tone.

  The answer supervision flag is interpreted as follows:

  — For ISDN trunks, if the answer supervision field contains a "0", the call interworked with non-ISDN trunks and the duration was calculated but does not have the degree of accuracy of a strictly ISDN call. Thus this field shows whether the call was interworked or went through a strictly ISDN. network.

— If the answer supervision field contains a “4” or “5”, the call went over a strictly ISDN network and the duration marked is accurate.

— For non-ISDN CO, FX, and WATS trunks that have the Answer Supervision field marked with a “4” or “5”, and are receiving answer supervision from the network, the duration is accurate.

— For Tie, Tandem and Access trunks that have the Answer Supervision field marked with a “4” or "5", the duration can only be assumed to be accurate if the PBX in question is the “network egress” PBX in a private network or a stand alone PBX. The duration is also accurate if all of the trunks the call goes over provide answer supervision.

When the call duration is not accurate (a 0 appears in the Answer Supervision field), the calls have often been timed via an administered time-out on a switch or from an earlier point in the call than when they actually got answered, because the switch could not determine when the call was answered.

- INS (3 digits) (G1 only)

This field specifies the ISDN Network Service (INS) requested for a call. This field applies only to ISDN calls. Each Network Specific Facility is translated into an INS according to Table 3-H below:

Table 3-H. Network Specific Facility to INS Mapping

Network Specific Facility	INS Value
Network Operator	324
Presubscribed Common Carrier Operator	325
Software Defined Network (SDN)	352
MEGACOM 800 Telecommunications Service	353
MEGACOM Telecommunications Service	354
INWATS	355
Maximum Banded WATS	356
AT&T Lonq Distance Service	358
ACCUNET Digital Service	357
OUTWATS Band 0 OUTWATS Band 1 . . . OUTWATS Band 255	33 34 . . . 288

- Inter-exchange Carrier (IXC) Code (1 digit hexadecimal representation) (3 digits with a G1 ISDN format)

  — Non-ISDN Formats

  IXC codes, numbered 1 through 15 (1 through F hexadecimal), are associated with the Version 2, Version 3, or DEFINITY Generic 1 Automatic Alternate Routing and Automatic Route Selection features and depict the carrier used on the call. This information is sent to the SMDR output device in ASCII code as a hexadecimal representation (for example, ASCII "F" equals “15“).

  An IXC access number is used to access a specific common carrier for a call. This number is of the form 10XXX, 950—1 XXX, or NXX—XXXX, where N is any digit 2 through 9 and X is any digit 0 through 9. The IXC access numbers applicable at a given location are associated with an IXC code on the IXC form. When ARS is used, and a routing pattern inserts one of the administered IXC codes, the associated IXC code is recorded. If no IXC access number is used, a 0 is recorded. In this case, either an inter-exchange carrier is not used on the call or the carrier is selected at the central office.

  — ISDN Formats (G1 Only)

  With a G1 ISDN record format, this field is a 3-digit field that identifies the actual IXC used on an ISDN call. This information is determined from the routing pattern administration. On AAR adn ARS calls, the 3-digft IXC value is administered in the routing pattern for all ISDN calls. If a user dials an IXC code with a 10XXX format as administered on the IXC Codes form, the SMDR device will put only the last 3 digits in the SMDR record. If a user dials a 7-digit IXC code, this field will contain a zero.

- Incoming Circuit Identification (2 digits with V1, V2, or V3) (3 digits with G1)

  This field contains the member number of a trunk within a trunk group used for an incoming call.

- Node Number (2-digits) (G1) (24-Word Records Only)

  This field identifies the DCS node number of a switch within a DCS arrangement. This number should be assigned the same number as the administered PBX-ID.

- Outgoing Circuit Identification (2 digits with V1, V2, or V3) (3 digits with G1)

  This field contains the member number of the trunk within a trunk group used for an outgoing call.

- Time in Queue (2 digits) (G1)

  The system does not use this field for recording time in queue. This field is used, however, for the last digits of an account code that exceeds 12 digits. It is also used in 18-word ISDN format records to display the first 2 digits of the INS code. This field is always blank in the 24-word records.

- MA-UUI (1 digit) (G1)

  Message Associated User-to-User Signaling (MA-UUI) is shown in the field which keeps track of the number of ISDN messages containing user data sent on an outgoing call. Data in this field can range from 0 to 9 and is found only on 24-word records.

The call detail information sent to the TELESEER SMDR unit is shown in Figure 3-33 (V1), Figure 3-34 (V2 or V3), Figure 3-35 (G1), and Figure 3-36 (G1 with ISDN).

The call detail information sent to the printer is shown in Figure 3-37 (V1), Figure 3-38 (V2 or V3), Figure 3-39 (G1), and Figure 3-40 (G1 with ISDN).

The call detail information sent to the 94A LSU and 3B2 CDRU is shown in Figure 3-41 (V1), Figure 3-42 (V2), Figure 3-43 (V3), Figure 3-44 (G1), and Figure 3-45 (G1 with ISDN).

The call detail information included in the 59-character Direct Output Record is shown in Figure 3-46 (V1), Figure 3-47 (V2 or V3), and Figure 3-48 (G1).

The call detail information included in the G1 24-word unformatted record and the G1 24-word expanded record is shown in Figures 3-49 and 3-50, respectively.

ASCII Character Position	Data Field Description
01-03	Space
04	Time Hour- (tens)
05	Time Hour- (units)
06	Time Minute (tens)
07	Time Minute (units)
08	Space
09	Duration Hour
10	Duration Minute (tens)
11	Duration Minute (units)
12	Duration Minute (tenths)
13	Space
14	Condition Code*
15	Space
16-18	Access Code Dialed†
19-21	Access Code Used†
22	Space
23-37	Dialed Number†
38	Space
39-42	Calling Number†
43	Space
44-48	Account Code†
49	Space
50-56	Space or Account Code† §
57	Space
58-59	Space or Account Code† §
60	Space
61	FRL or Account Code† §
62-64	Space
65-66	Incoming Circuit ID‡
67-69	Space
70-71	Outgoing Circuit ID‡
72-76	Space
77	Carriage Return
78	Line Feed
79-81	null

* Refer to Table 3-F.

† Data is right justified and padded with blanks.

‡ Data is right justified and padded with Os.

§ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-33. SMDR Data Format—TELESEER SMDR Unit (V1)

ASCII Character Position	Data Field Description
01-03	Space
04	Time Hour- (tens)
05	Time Hour- (units)
06	Time Minute (tens)
07	Time Minute (units)
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Condition Code*
13-15	Access Code Dialed†
16-18	Access Code Used†
19-33	Dialed Number†
34-38	Calling Number†
39-53	Account Code†
54	FRL
55	IXC
56-57	Incoming Circuit ID‡
58	Space
59-60	Outgoing Circuit ID‡
61-62	Space
63-69	Space (V2) Authorization Code or 6th through 12th Digits of Account Code (V3)
70-76	Space
77	Carriage Return
78	Line Feed
79-81	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with Os.

Figure 3-34. SMDR Data Format—TELESEER SMDR Unit (V2 or V3)

ASCII Character Position	Data Field Description
01-03	Space
04	Time Hour- (tens)
05	Time Hour- (units)
06	Time Minute (tens)
07	Time Minute (units)
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Condition Code*
13-15	Access Code Dialed†
16-18	Access Code Used†
19-33	Dialed Number†
34-38	Calling Number†
39-53	Account Code†
54	FRL
55	IXC
56-58	Incoming Circuit ID‡
59-61	Outgoing Circuit ID‡
62	Feature Flag
63-69	Authorization Code
70-71	Time in Queue
72-76	Space
77	Carriage Return
78	Line Feed
79-81	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with Os.

Figure 3-35. SMDR Data Format—TELESEER SMDR Unit (G1)

ASCII Character Position	Data Field Description
01-03	Space
04	Time Hour- (tens)
05	Time Hour- (units)
06	Time Minute (tens)
07	Time Minute (units)
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Condition Code*
13-15	IXC†
16-18	Access Code Used†
19-33	Dialed Number†
34-38	Calling Number†
39-53	Account Code†
54	INS (third digit)
55	FRL
56-58	Incoming Circuit ID‡
59-61	Outgoing Circuit ID‡
62	Feature Flag
63-69	Authorization Code
70-71	INS (first and second digits)
72-76	Space
77	Carriage Return
78	Line Feed
79-81	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with Os.

Figure 3-36. SMDR Data Format—TELESEER SMDR Unit With ISDN (G1)

ASCII Character Position	Data Field Description
00-02	Space
03	Time Hour- (tens)
04	Time Hour- (units)
05	Time Minute (tens)
06	Time Minute (units)
07	Space
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Space
13	Condition Code*
14	Space
15-17	Access Code Dialed†
18	Space
19-21	Access Code Used†
22	Space
23-37	Dialed Number†
38	Space
39-42	Calling Number†
43	Space
44-48	Account Code†
49	Space
50-56	Space or Account Code† §
57	Space
58-59	Space or Account Code† §
60	Space
61	FRL or Account Code† §
62-64	Space
65-66	Incoming Circuit ID‡
67-69	Space
70-71	Outgoing Circuit ID‡
72	Space
73	Carriage Return
74	Line Feed

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with Os.
§ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-37. SMDR Direct Output Format From System to Printer (V1)

ASCII Character Position	Data Field Description
01	Time Hour- (tens)
02	Time Hour- (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration Hour
07	Duration Minute (tens)
08	Duration Minute (units)
09	Duration Minute (tenths)
10	Space
11	Condition Code*
12	Space
13-15	Access Code Dialed†
16	Space
17-19	Access Code Used†
20	Space
21-35	Dialed Number†
36	Space
37-41	Calling Number†
42	Space
43-57	Account Code†
58	Space
59-65	Space (V2) Authorization Code or 6th through 12th Digits of Account Code (V3)
66-69	Space
70	FRL
71	Space
72	IXC
73	Space
74-75	Incoming Circuit ID‡
76-77	Space
78-79	Outgoing Circuit ID‡
80-82	Space
83	Carriage Return
84	Line Feed

* Refer to Table 3-F.

† Data is right justified and padded with blanks.

‡ Data is right justified and padded with Os.

Figure 3-38. SMDR Direct Output Format From System to Printer (V2 or V3)

ASCII Character Position	Data Field Description
01	Time Hour- (tens)
02	Time Hour- (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration Hour
07	Duration Minute (tens)
08	Duration Minute (units)
09	Duration Minute (tenths)
10	Space
11	Condition Code*
12	Space
13-15	Access Code Dialed†
16	Space
17-19	Access Code Used†
20	Space
21-35	Dialed Number†
36	Space
37-41	Calling Number†
42	Space
43-57	Account Code†
58	Space
59-65	Authorization Code
66	Space
67-68	Time in Queue
69	Space
70	FRL
71	Space
72	IXC
73	Space
74-76	Incoming Circuit ID‡
77	Space
78-80	Outgoing Circuit ID‡
81	Space
82	Feature Flag
83	Carriage Return
84	Line Feed

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with Os.

Figure 3-39. SMDR Direct Output Format From System to Printer (G1)

ASCII Character Position	Data Field Description
01	Time Hour- (tens)
02	Time Hour- (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration Hour
07	Duration Minute (tens)
08	Duration Minute (units)
09	Duration Minute (tenths)
10	Space
11	Condition Code*
12	Space
13-15	IXC†
16	Space
17-19	Access Code Used†
20	Space
21-35	Dialed Number†
36	Space
37-41	Calling Number†
42	Space
43-57	Account Code†
58	Space
59-65	Authorization Code
66	Space
67-68	INS
69	Space
70	INS (third digit)
71	Space
72	FRL
73	Space
74-76	Incoming Circuit ID‡
77	Space
78-80	Outgoing Circuit ID‡
81	Space
82	Feature Flag
83	Carriage Return
84	Line Feed

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with Os.

Figure 3-40. ISDN SMDR Direct Output Format From System to Printer (G1)

ASCII Character Position	Data Field Description
01	Time Duration-Hours
02	Time Duration-Minutes (tens)
03	Time Duration-Minutes (units)
04	Time Duration-Minutes (tenths)
05	Condition Code*
06-08	Access Code Dialed†
09-11	Access Code Used†
12-26	Dialed Number†
27-30	Calling Number†
31-35	Account Code†
36-44	Space or Account Code† §
45	FRL or Account Code† §
46	Space
47-48	Incoming Circuit ID
49	Space
50-51	Outgoing Circuit ID
52-54	Space
55	Carriage Return .
56	Line Feed
57-59	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
§ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-41. SMDR Direct Output Format From System to 94A Local Storage Unit System or 362 CDRU (V1)

ASCII Character Position	Data Field Description
01	Time Duration-Hours
02	Time Duration-Minutes (tens)
03	Time Duration-Minutes (units)
04	Time Duration-Minutes (tenths)
05	Condition Code*
06-08	Access Code Dialed†
09-11	Access Code Used†
12-26	Dialed Number†
27-30	Calling Number†
31-35	Account Code†
36-44	Space or Account Code† §
45	FRL or Account Code† §
46	Calling Number (fifth and most significant digit)
47-48	Incoming Circuit ID
49	Space
50-51	Outgoing Circuit ID
52-53	Space
54	IXC
55	Carriage Return
56	Line Feed
57-59	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
‡ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-42. SMDR Direct Output Format From System to 94A Local Storage Unit System or 3B2 CDRU (V2)

ASCII Character Position	Data Field Description
01	Time Duration-Hours
02	Time Duration-Minutes (tens)
03	Time Duration-Minutes (units)
04	Time Duration-Minutes (tenths)
05	Condition Code*
06-08	Access Code Dialed†
09-11	Access Code Used†
12-26	Dialed Number†
27-30	Calling Number†
31-35	Account Code (First Five Digits)†
36-42	Authorization Code or 6th through 12th Digits of Account Code†
43-44	Space or 13th and 14th Digits of Account Code
45	FRL or 15th Digit of Account Code†
46	Calling Number (fifth and most significant digit)
47-48	Incoming Circuit ID
49	Space
50-51	Outgoing Circuit ID
52-53	Space
54	IXC
55	Carriage Return
56	Line Feed
57-59	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.

Figure 3-43. SMDR Direct Output Format From System to 94A Local Storage Unit System or 3B2 CDRU (V3)

ASCII Character Position	Data Field Description
01	Time Duration-Hours
02	Time Duration-Minutes (tens)
03	Time Duration-Minutes (units)
04	Time Duration-Minutes (tenths)
05	Condition Code*
06-08	Access Code Dialed†
09-11	Access Code Used†
12-26	Dialed Number†
27-30	Calling Number†
31-35	Account Code (First 5 Digits)†
36-42	Authorization Code or 6th Through 12th Digits of Account Code†
43-44	Time in Queue or 13th and 14th digits of account code†
45	FRL or 15th Digit of Account Code†
46	Calling Number (fifth and most significant digit)
47-48	Incoming Circuit ID §
49	Feature Flag
50-52	Outgoing Circuit ID §
53	Incoming Circuit ID (third digit)
54	IXC
55	Carriage Return
56	Line Feed
57-59	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
§ Data is right justified and padded with zeros.

Figure 3-44. SMDR Direct Output Format From System to 94A Local Storage Unit System or 3B2 CDRU (G1)

ASCII Character Position	Data Field Description
01	Time Duration-Hours
02	Time Duration-Minutes (tens)
03	Time Duration-Minutes (units)
04	Time Duration-Minutes (tenths)
05	Condition Code*
06-08	IXC †
09-11	Access Code Used†
12-26	Dialed Number†
27-30	Calling Number†
31-35	Account Code (First 5 Digits)†
36-42	Authorization Code or 6th Through 12th Digits of Account Code†
43-45	INS†
46	Calling Number (fifth and most significant digit)
47-48	Incoming Circuit ID §
49	Feature Flag
50-52	Outgoing Circuit ID §
53	Incoming Circuit ID (third digit)
54	FRL†
55	Carriage Return
56	Line Feed
57-59	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.
§ Data is right justified and padded with zeros.

Figure 3-45. ISDN SMDR Direct Output Format From System to 94A Local Storage Unit System or 3B2 CDRU (G1)

ASCII Character Position	Data Field Description
00-02	Space
03	Time-Hours (tens)
04	Time-Hours (units)
05	Time-Minutes (tens)
06	Time-Minutes (units)
07	Space
08	Duration-Hours
09	Duration-Minutes (tens)
10	Duration-Minutes (units)
11	Duration-Minutes (tenths)
12	Space
13	Condition Code*
14	Space
15-17	Access Code Dialed†
18-20	Access Code Used†
21	Space
22-36	Dialed Number†
37	Space
38-41	Calling Number†
42	Space
43-47	Account Code (First 5 Digits)†
48-56	Space or 6th Through 14th Digits of Account Code†
57	FRL or 15th Digit of Account Code
58	Carriage Return
59	Line Feed
60-62	null

* Refer to Table 3-F.

† Data is right justified and padded with blanks.

Figure 3-46. SMDR 59-Character Direct Output Format (V1)

ASCII Character Position	Data Field Description
00	Time-Hours (tens)
01	Time-Hours (units)
02	Time-Minutes (tens)
03	Time-Minutes (units)
04	Duration-Hours
05	Duration-Minutes (tens)
06	Duration-Minutes (units)
07	Duration-Minutes (tenths)
08	Condition Code*
09-11	Access Code Dialed†
12-14	Access Code Used†
15-29	Dialed Numbert†
30-34	Calling Number†
35-49	Account Code†
50	FRL
51	IXC
52-53	Incoming Circuit ID
54	Space
55-56	Outgoing Circuit ID
57	Space
58	Carriage Return
59	Line Feed
60-62	null

* Refer to Table 3-F.

† Data is right justified and padded with blanks.

Figure 3-47. SMDR 59-Character Direct Output Format (V2 or V3)

ASCII Character Position	Data Field Description
00	Time-Hours (tens)
01	Time-Hours (units)
02	Time-Minutes (tens)
03	Time-Minutes (units)
04	Duration-Hours
05	Duration-Minutes (tens)
06	Duration-Minutes (units)
07	Duration-Minutes (tenths)
08	Condition Code*
09-11	Access Code Dialed†
12-14	Access Code Used†
15-29	Dialed Number†
30-34	Calling Number†
35-49	Account Code†
50	FRL
51	IXC
52-54	Incoming Circuit ID
55-57	Outgoing Circuit ID
58	Carriage Return
59	Line Feed
60-62	null

* Refer to Table 3-F.
† Data is right justified and padded with blanks.

Figure 3-48. SMDR 59-Character Direct Output Format (G1)

ASCII Character Position	Data Field Description
01	Time Hour (tens)
02	Time Hour (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Duration-Hours
06	Duration-Minutes (tens)
07	Duration-Minutes (units)
08	Duration-Minutes (tenths)
09	Condition Code*
10-13	Access Code Dialed†
14-17	Access Code Used†
18-32	Dialed Number†
33-42	Calling Number†
43-57	Account Code†
58-64	Authorization Code†
65-66	Time in Queue†
67	FRL
68-70	Incoming Circuit ID †
71-73	Outgoing Circuit ID †
74	Feature Flag
75-76	Attendant Console†
77-80	incoming Trunk Group Access Code†
81-82	Node Number
83-85	INS†
86-88	IXC†
89	BCC
90	MA-UUI
91-100	Reserved
101	Carriage Return
102	Line Feed
103-105	Null

* Refer to Table 3-F.

† Data is right justified and padded with blanks.

Figure 3-49. 24-Word ISDN Unformatted SMDR Record Format (G1)

ASCII Character Position	Data Field Description
01	Time Hour (tens)
02	Time Hour (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration-Hours
07	Duration-Minutes (tens)
08	Duration-Minutes (units)
09	Duration-Minutes (tenths)
10	Space
11	Condition Code*
13-16	Access Code Dialed†
17	Space
18-21	Access Code Used†
22	Space
23-37	Dialed Number†
38	Space
39-48	Calling Number†
49	Space
50-64	Account Code†
65	Space
66-72	Authorization Code†
73	Space
74-75	Time in Queue†
76	Space
77	FRL
78	Space
79-81	Incoming Circuit ID †
82	Space
83-85	Outgoing Circuit ID †
86	Space
87	Feature Flag
88	Space

Figure 3-50. 24-Word ISDN Expanded SMDR Record Format (G1)

ASCII Character Position	Data Field Description
89-90	Attendant Console†
91	Space
92-95	Incoming Trunk Group Access Code†
96	Space
97-98	Node Number
99	Space
100-102	INS†
103	Space
104-106	IXC†
107	Space
108	BCC
109	Space
110	MA-UUI
111	Space
112-130	Reserved
131	Carriage Return
132	Line Feed
133-135	Null

* Refer to Table 3-F.

† Data is right justified and padded with blanks.

Figure 3-50. 24-Word ISDN Expanded SMDR Record Format (G1) (Contd)

Date Record Format

Three formats are available for date records, one for the 94A LSU or 3B2 CDRU (Figure 3-51), one for the printer (Figure 3-52), and one for the TELESEER SMDR unit (Figure 3-53). The records sent to the TELESEER SMDR and printer contain. the date only while the records sent to the 94A LSU or 362 CDRU contain both date and time.

ASCII Character Position	Data Field Description
01-02	Hour*
03	Colon (:)
04-05	Minute*
06	Blank
07-08	Month*
09	Slash (/)
10-11	Day*
12	Carriage Return
13	Line Feed
14-16	Null

* Leading zero added if needed.

Figure 3-51. Date Record Format to 94A LSU or 3B2 CDRU

ASCII Character Position	Data Field Description
01-02	Month*
03	Space
04-05	Day*
06	Carriage Return
07	Line Feed
08-10	Null

* Leading zero added if needed.

Figure 3-52. Date Record Format to Printer

ASCII Character Position	Data Field Description
00-01	Month*
02-03	Day
04	Carriage Return
05	Line Feed
06-08	Null

* Leading zero added if needed.

Figure 3-53. Date Record Format to TELESEER SMDR Unit

Set Time and Date

The system clock must be set for daylight savings time. Changing the time and date ensures that SMDR records have the correct time and date for the records being kept. The time and date can be changed using the System Access Terminal or Manager I terminal.

If the time is changed while calls are in progress, the actual call durations for these calls are not reflected in the SMDR record.

SMDR Output Devices

SMDR data is collected by the system and is continually sent to an output device for processing. An output device could be a TELESEER SMDR unit, printer, 94A LSU, 3B2 CDRU AP (not available with XEV2, or G1), host computer, or customer-provided equipment. Only one output device can be used for a V1, V2, V3, XEV2, or XEV3 system. A G1 system can have two output devices.

With XEV2, XEV3, and G1, a standard RS-232C interface is provided by the system's processor circuit pack. This allows for direct connection of the SMDR output device to the system. If this port is not used or if the system is a V1 or V2 system, additional interface equipment is required as described in the Hardware and Software Requirements part of this feature description. With G1, the system can support two SMDR output devices. One of these devices can use the direct RS-232C connection; the other device will require the additional interface equipment.

When a G1 system has two SMDR output devices, one device is administered as the Primary SMDR Output Device: the other device is administered as the Secondary SMDR Output Device. The Secondary output device can be a device such as the 94A LSU or 3B2 CDRU (discussed later). This Secondary Output Device can be used for various purposes. For example, it could be used to provide information to an NCOSS (Network Control Operations Support System) for assessing network performance or helping to find network problems. The following information applies to the port used for the Secondary SMDR output device in G1 systems:

- Data going to the secondary port should be the same as that going to the primary port. However, SMDR records sent to the secondary port can only be in the 94A LSU or 3B2 CDRU format or unformatted.

- If the system experiences problems in sending records to the primary SMDR Output Device, the system will discontinue sending records to the secondary port for 2 minutes. The secondary port should be run at the highest possible speed in order to prevent loss of information on the port.

- If more than 175 records have not been sent to the primary SMDR port, the secondary port is busied out for 2 minutes. This makes system resources available to send data to the primary SMDR port before the data is lost. The system will continue to busy out the secondary port for 2-minute intervals until less than 175 records remain to be sent to the primary port.

- The primary and secondary ports work independently. Each port will work even if the link to the other port is down. If a link is down for more than a minute, some data may be lost. However, the most recent 200 records are stored for the primary port even when a loss of records occurs. When the link comes back up, these records are output on a first-in, first-out basis.

The system can store up to 200 SMDR records which are sent to the output devices.

A 1200 bits/second rate may only be used over the cable for the TELESEER SMDR unit, 94A LSU, or printer, if the system line size is less than 1000. If the system line size is greater than 1000, a rate of at least 2400 must be used with the 18-word SMDR records. If the system line size is greater than 600, a rate of at least 2400 must be used with the 24-word SMDR records. The link to the AP (not available with XEV2, or G1) should operate at 9600 baud.

The time stamp on calls recorded by SMDR is normally applied at the end of the call.

The following paragraphs give a brief description of each output device. In addition to the devices described below, the system supports output devices that require a 59-character SMDR call detail record.

TELESEER SMDR Unit

The TELESEER SMDR unit is an output device that stores two types of information regarding each call record: call record details and summary totals. Call details consist of the following:

- Time of call
- Duration of call
- Account code
- Type of call
- Extension of call

- Dialed number
- Date of call.

Summary totals are running totals of the call records that fall into the following categories:

- Time of day (on an hourly basis)
- Cost (eight ranges)
- Duration (seven time ranges)
- Date
- Department by cost center and extension
- Call type
- Account code
- Access code/trunk number/trunk group number
- Printed call categories
- Recorded call categories.

The TELESEER SMDR unit can store up to 28,000 call records, 500 extension numbers, and 2000 account codes. SMDR records sent to the TELESEER SMDR unit are 80 bytes or 640 characters long.

The TELESEER SMDR unit provides four different types of reports: Summary, Account Code Detail, Activity, and Selection.

The Summary Report provides a condensed listing of the number, duration, and cost of calls. This report provides a general overview of voice terminal activity. The following information can be taken from the report:

- Large departmental voice terminal costs
- Large costs attributed to specific extension numbers
- Improper use of WATS lines
- Voice terminal usage for specific account codes
- Lengthy voice terminal conversations.

An example of a Summary Report is shown in Figure 3-54.

The Account Code Detail Report lists call record details for each call record that contains an account code. This report is helpful in tracking calls from specific users. in addition, this report can be used for user billback or cost allocation by account code. The Account Code

Report provides duration of calls, number dialed, type of calls made, account codes, and cost for each call. An example of an Account Code Detail Report is shown in Figure 3-55.

An Activity Report lists call record details for each extension number assigned to the system. The Activity Report provides time, date, type of call, account codes, and cost of each call. An example of a Unit Activity Report is shown in Figure 3-56.

A Selection Report allows the System Manager to specify the type of information to be printed in a report. All call record details stored in the TELESEER SMDR unit that pertain to parameters selected are printed. Any or all of the following data can be specified:

- Time of day
- Date for each
- Cost for each call
- Duration of each call measured in hours, minutes, and seconds
- Extension number that originated the call
- Trunk Number/Access Code
- Account code number used for the call
- Dialed number
- Type trunk used for the call
- Department.

An example of a Unit Selection Report is shown in Figure 3-57.

Printer

An 80- or 132-column (character) printer can be connected as an SMDR output device. The printer prints SMDR records in a 1-line format (V1, V2, or V3) or a 2-line format (G1). No data processing or reports are provided. The 18-word SMDR records sent to the printer are 84 bytes or 672 bits long. The 24-word SMDR records sent to the printer are 135 bytes or 1080 bits long.

94A Local Storage Unit (LSU)

The 94A LSU collects and stores Message Detail Records (MDRs) data from the system. The 94A LSU stores MDRs for Electronic Tandem Network (ETN) customers or multi-location customers served by other System 75s or DEFINITY Generic 1s. SMDR records sent to the 94A LSU are 59 bytes or 470 bits long. The 94A LSU can handle up to 14,600 call records per hour, store up to 16,000 records, and transmit up to 7200 calls per hour over a 1200 baud link to a 93B Centralized Message Detail Recorder (CMDR).

3B2 Call Detail Recording Utility (CDRU)

The 3B2 CDRU is an accounting program which collects and stores statistics about calls on a system. The CDRU runs on model 300, 310, or 400 of the AT&T 3B2 microcomputer. The CDRU system can collect up to 24,000 call records per hour. In addition, the CDRU system can store up to:

- 400,000 records if the 3B2 has a 30-megabyte disk
- 1.2 million records if the 3B2 has a 72-megabyte disk
- 1.7 million records if the 3B2 has a 72-megabyte disk and an optional 3B2 expansion module.

After call detail records are stored, the CDRU can forward them to the following systems or devices for subsequent processing:

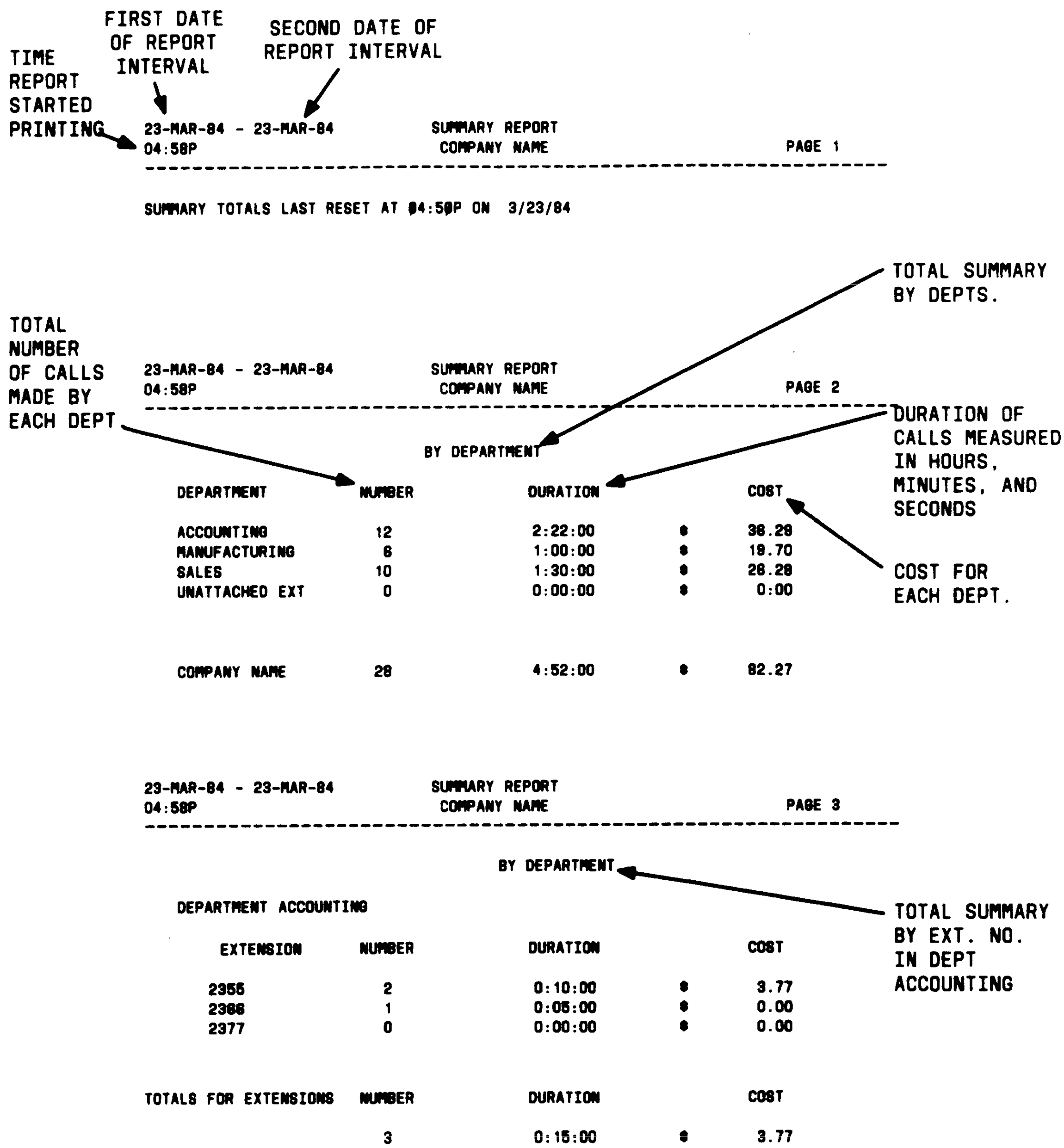
- AT&T CSM (Centralized System Management) software
- AT&T 93B CMDR remotely located polling device
- AT&T NCOSS (Network Control Operations Support System)
- Host Computer
- Tape Drive
- ASCII printer.

Applications Processor (AP) (V1, V2, V3, and XEV3)

The AP Call Detail Recording and Reporting (CDRR) collects and formats switch generated SMDR. CDRR station message detail records are generated to enable the customer to assess individual voice terminals in the system for trunk usage. A station message detail **record is created for the following types of calls:**

- Outgoing call—a call originating at a voice terminal in the system and going out on a trunk
- Incoming call—a call incoming on a trunk and terminating at a voice terminal in the system.

The customer has the option of turning off station message detail record generation for incoming calls, specific trunk group(s), and ineffective call attempts via an administration procedure.



23-MAR-84 - 23-MAR-84   SUMMARY REPORT
04:58P   COMPANY NAME   PAGE 1

SUMMARY TOTALS LAST RESET AT 04:50P ON 3/23/84

23-MAR-84 - 23-MAR-84   SUMMARY REPORT
04:58P   COMPANY NAME   PAGE 2

BY DEPARTMENT

DEPARTMENT	NUMBER	DURATION		COST
ACCOUNTING	12	2:22:00	$	36.29
MANUFACTURING	6	1:00:00	$	19.70
SALES	10	1:30:00	$	26.28
UNATTACHED EXT	0	0:00:00	$	0:00
COMPANY NAME	28	4:52:00	$	82.27

23-MAR-84 - 23-MAR-84   SUMMARY REPORT
04:58P   COMPANY NAME   PAGE 3

BY DEPARTMENT

DEPARTMENT ACCOUNTING

EXTENSION	NUMBER	DURATION		COST
2355	2	0:10:00	$	3.77
2366	1	0:05:00	$	0.00
2377	0	0:00:00	$	0.00

TOTALS FOR EXTENSIONS	NUMBER	DURATION		COST
	3	0:15:00	$	3.77

Figure 3-54. Example of a TELESEER SMDR Unit Summary Report

THIS IS THE TRUNK ACCESS CODE, TRUNK NUMBER, OR TRUNK GROUP NUMBER

THIS INDICATES THE TYPE OF CALL
IDD=INTERNATIONAL
OST=INTERSTATE LONG DISTANCE
OCC=OTHER COMMON CARRIERS

```
23-MAR-84 - 23-MAR-84                  ACCOUNT CODE REPORT
0:12P                                      COMPANY NAME                         PAGE 1

03/23 08:43AM    3:00   101   7         454-1362     FX 558432                   1.05
03/23 01:01PM    4:00   530   8         473-1502    LOC 558432                   0.08

      NUMBER             DURATION              COST
        2                 0:07:00            $   1.13

ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 88644
 DATE  TIME  DURATION EXT  ACC DIALED DIGITS      TYP ACCOUNT CODE   COST $
 ----  ----  -------- ---  --- ------ ------      --- ------- ----   -------
03/23 11:31AM    8:00 1768   9 1-416-324-5012     IDD 88644             2.55
03/23 01:16PM   30:00 2388   9 1-206-888-4587     OST 88644            13.20

      NUMBER             DURATION              COST
        2                 0:38:00            $  15.75

ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 95643
 DATE  TIME  DURATION EXT  ACC DIALED DIGITS      TYP ACCOUNT CODE   COST $
 ----  ----  -------- ---  --- ------ ------      --- ------- ----   -------
03/23 11:28AM   11:00 2400   9 9872011222333413128312458
                                                 OCC 95643              3.74

      NUMBER             DURATION              COST
        1                 0:11:00            $   3.74

ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 12367
 DATE  TIME  DURATION EXT  ACC DIALED DIGITS      TYP ACCOUNT CODE   COST $
 ----  ----  -------- ---  --- ------ ------      --- ------- ----   -------
03/23 10:10AM    5:00 2388  67                    --- 12367             0.00

      NUMBER             DURATION              COST
        1                 0:05:00            $   0.00

ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 41383
 DATE  TIME  DURATION EXT  ACC DIALED DIGITS      TYP ACCOUNT CODE   COST $
 ----  ----  -------- ---  --- ------ ------      --- ------- ----   -------
03/23 11:01AM   12:00 3011   4 683-2828            FX 41384             5.01
03/23 01:15PM    5:00 3011   7 288-5454            FX 41384             1.77
03/23 02:13PM    3:00 2400   8 1-417-887-3488     WTS 41383             0.70

      NUMBER             DURATION              COST
        3                 0:20:00            $   7.48
```

Figure 3-55. Example of a TELESEER SMDR Unit Account Code Detail Report

FIRST DATE OF REPORT INTERVAL

TIME REPORT STARTED TO PRINT

```
23-MAR-84 - 23-MAR-84                    ACTIVITY REPORT
Ø5:Ø7P                                    COMPANY NAME                    PAGE 1
DEPARTMENT SALES

ACTIVITY REPORT FOR EXTENSION 4150
  DATE  TIME  DURATION  EXT  ACC DIALED DIGITS     TYP ACCOUNT CODE   COST $
  ----  ----  --------  ---  --- ------ ------     --- ------- ----  -------
03/23 10:10AM    5:00  4155  66                    FX                   1.77
03/23 02:53PM   12:00  4155   8 1-208-324-5151     WST                  2.86
03/23 03:12PM    5:00  4155   8 1-312-654-7829     WST                  1.18

TOTALS          22:00                                                   5.81

TOTAL CALLS 3

ACTIVITY REPORT FOR EXTENSION 4366
NO RECORDS STORED

ACTIVITY REPORT FOR EXTENSION 4444
NO RECORDS STORED

23-MAR-84 - 23-MAR-84                    ACTIVITY REPORT
05:07P                                    COMPANY NAME                    PAGE 2
COST CENTER 516

ACTIVITY REPORT FOR EXTENSION 4355
  DATE  TIME  DURATION  EXT  ACC DIALED DIGITS     TYP ACCOUNT CODE   COST $
  ----  ----  --------  ---  --- ------ ------     --- ------- ----  -------
03/23 10:12AM    2:00  4355   8 1-408-454-1362     WST 88267            0.46
03/23 01:16PM   30:00  4355   9 1-206-999-5478     OST 54321           13.20

TOTALS          32:00                                                  13.66

TOTAL CALLS 2

ACTIVITY REPORT FOR EXTENSION 4455
NO RECORDS STORED

ACTIVITY REPORT FOR EXTENSION 45ØØ
  DATE  TIME  DURATION  EXT  ACC DIALED DIGITS     TYP ACCOUNT CODE   COST $
  ----  ----  --------  ---  --- ------ ------     --- ------- ----  -------
03/23 06:05AM   11:00  4500   9 1-419-643-7474     OST 34671            3.02

TOTALS          11:00                                                   3.02

TOTAL CALLS 1

ACTIVITY REPORT FOR EXTENSION 4622
  DATE  TIME  DURATION  EXT  ACC DIALED DIGITS     TYP ACCOUNT CODE   COST $
  ----  ----  --------  ---  --- ------ ------     --- ------- ----  -------
03/23 00:05AM   10:00  4622   9 1-419-643-7474     OST 54321            1.49
03/23 02:02PM    1:00/ 4622   7 344-7542            FX 88267            0.33

TOTALS          11:00                                                   1.82

TOTAL CALLS 2
```

Figure 3-56. Example of a TELESEER SMDR Unit Activity Report

SELECTION REPORT
COMPANY NAME

DURATION(10:00 - 18:00:00)

DATE	TIME	DURATION	EXT	ACC	DIALED DIGITS	TYP	ACCOUNT CODE	COST $
03/23	10:42AM	10:00	5300	64		WST		1.78
03/23	11:01AM	12:00	3011	4	663-2828	FX	41363	5.01
03/23	11:15AM	10:00	1122	9	1-315-681-0846	IST	335678	0.93
03/23	11:28AM	11:00	2400	9	9872011222333413128312458			
						OCC	85843	3.74
03/23	11:35AM	22:00	1011	9	1-213-324-5012	OST	12345	9.76
03/23	00:05AM	10:00	4622	9	1-418-843-7474	OST	54321	1.48
03/23	06:05PM	11:00	4500	9	1-818-843-7474	OST	34671	3.02
03/23	01:16PM	30:00	4355	9	1-206-999-5478	OST	54321	13.20
03/23	01:16PM	30:00	2388	9	1-206-888-4587	OST	88844	13.20
03/23	02:35PM	15:00	1122	9	9871011278565618188883200			
						OST		4.60
03/23	02:53PM	12:00	4155	6	1-206-324-5151	WST		2.86
03/23	03:02PM	35:00	2388	8		---		6.30
03/23	03:22PM	25:00	2400	9	9872011263457813153857218			
						OCC		1.72
TOTALS		3:53:00						67.62

TOTAL CALLS 13

TOTAL TIME SPENT ON 13 CALLS: 3 HOURS AND 15 MINUTES

TOTAL COST FOR 13 CALLS WHICH LASTED FOR 3 HOURS AND 15 MINUTES

Figure 3-57. Example of a TELESEER SMDR Unit Selection Report

Considerations

SMDR provides detailed call information on incoming and outgoing calls. This information can be used to facilitate cost allocation, traffic analysis, and detection of unauthorized calls.

The system can store up to 200 SMDR records for the primary port, which are sent to the output devices when the link comes back up.

The TELESEER SMDR unit can store up to 28,000 call records, 500 extension numbers, and 2000 account codes.

The 94A LSU can handle up to 14,600 call records per hour, store up to 16,000 records, and transmit up to 7200 calls per hour to a 936 CMDR.

When a voice terminal user wants an SMDR record generated for a particular account number, the SMDR access code (for example, *6) and the account number must be dialed before the Automatic Route Selection, Automatic Alternate Routing, or trunk access code and called number are dialed.

The originally dialed extension number on an incoming call, or the originator's extension number on an outgoing call, is always recorded for SMDR even if the call is transferred to another voice terminal.

On an attendant-assisted call, whether the attendant dials the outside number or allows Through Dialing, the extension. number of the requesting user will be recorded for SMDR. However, the attendant must dial an account code, if provided, before dialing the trunk access code.

If the attendant is extending a call to a voice terminal, an account code can be dialed before the extension number is dialed.

Voice terminal users cannot dial an account code when extending a call to another voice terminal. However, a voice terminal user extending a call to a trunk can dial an account code before dialing the Automatic Route Selection or trunk access code.

With G1, if the system line size is greater than 1000, the SMDR device must support a baud rate of at least 2400 bps when using 18-word SMDR records. If the system line size is greater than 600, the SMDR device must support a baud rate of at least 2400 bps when using 24-word SMDR records.

With V3 and G1, SMDR records of DS-1 calls are only generated if the answer supervision time-out is exceeded or if the call is answered at the far end. Therefore, more accurate SMDR records for DS-1 facilities can be obtained by setting the Answer Supervision Timeout field on the DS-1 tie trunk form to the highest possible value (300 seconds).

Interactions

The following interaction descriptions assume SMDR is activated.

- Abbreviated Dialing

  When Abbreviated Dialing or a Facility Busy Indication button is used to make or complete a call, all digits outpulsed (up to a maximum of 15) will appear on the SMDR record.

- Attendant Console

  If an attendant-assisted call involves an outgoing trunk, the primary extension of the voice terminal user which requested attendant service is recorded as the calling number, even if the attendant dialed the outside number. Condition Code 1 indicates the call was assisted by the attendant.

  If the attendant allows through dialing, the primary extension of the voice terminal user who dialed the number is recorded as the calling party. Condition Code 1 indicates that a trunk access code was extended by the attendant. Condition Code 7 indicates that a feature access code was extended by the attendant.

  On attendant-assisted calls that require an account code, the account code must be entered before the trunk access code.

  If the attendant is redirecting an incoming call to a voice terminal, the attendant may dial an account code before dialing the extension number.

- Authorization Codes (V3 or G1)

  Authorization codes will be recorded on 94A LSU and 3B2 CDRU SMDR records if account codes do not exceed five digits. With G1, the authorization code is always recorded on the printer, TELESEER, and 24-word SMDR records. On the 59-character SMDR records, the authorization code is never recorded.

- Automatic Alternate Routing and Automatic Route Selection

  SMDR records the following information for Automatic Route Selection (ARS):

  — Fact an ARS call was made

  — Calling extension number

  — Facility Restriction Level of the calling extension

  — Called number

  — Type of trunk group used for the ARS call

  — Time of call completion

— Call duration (how long the parties talked)

— Inter-exchange carrier code, if any, in Version 2, Version 3, or DEFINITY Generic 1.

If SMDR is suppressed for the trunk group actually used on an ARS call, an SMDR record is not generated; otherwise, Condition Code 7 applies. The ARS access code is recorded in the Access Code Dialed field and the trunk access code for the trunk group actually used is recorded in the Access Code Used field:

If an AAR call is placed to a busy trunk group and SMDR is suppressed for that trunk group, the user hears reorder tone and the SMDR output shows an ineffective call attempt.

If an ARS call is an attendant assisted call (that is, a voice terminal user calls the attendant, the attendant dials the ARS access. code, and then releases the call), the SMDR record will show the call with a condition code of 7 (ARS call) instead of a condition code of 1 (attendant assisted call). This occurs because SMDR is not notified until after the trunk is seized and, in this case, the trunk is not seized until the voice terminal user dials the number.

- Bridged Call Appearance

  SMDR does not record any information on the party who bridges onto a call. Instead, the number that was called appears in the dialed number field of the SMDR record. The duration of the call is recorded when the last party drops off the call.

- Call By Call Service Selection

  When a successful call is made on a Call By Call Service Selection trunk, the network specific facility used on the call is translated into an INS number and recorded in the INS field of the SMDR record. If a Call By Call Semite Selection call is unsuccessful because of an administered trunk usage allocation plan, the INS number is recorded in the INS field of the report with a condition code of “E”.

- Call Coverage

  When an incoming call is answered by a covering voice terminal, the extension number dialed by the originating party is recorded as the dialed number.

- Call Forwarding All Calls

  When a call is forwarded to another voice terminal, the extension number dialed by the calling party is recorded as the dialed number.

- Call Park

  When a voice terminal user parks an incoming call, that user’s extension is recorded as the dialed number in the SMDR record.

- Call Pickup

  When the call is answered by another voice terminal user in the pickup group, the extension number dialed by the calling party is recorded as the dialed number.

- Call Waiting Termination

  Call duration timing starts when the voice terminal answers an incoming call.

- Centralized Attendant Service (CAS)

  If a CAS attendant extends a call for a user, and SMDR is not assigned to the RLT group, the user's extension is recorded as the originator of the call. If the RLT trunk group does have SMDR administered, the RLT trunk is recorded. If a CAS attendant answers a call but does not extend the call, no SMDR records are made.

- Central Office Trunks

  All incoming and outgoing calls on a CO trunk group will be recorded.

- Conference

  A call is considered a conference call if it contains at least one trunk which is eligible for SMDR recording plus two or more nonattendant parties. Condition Code C applies to each SMDR record made for a conference call.

  For a conference call, a separate SMDR record is produced for each outgoing/incoming trunk serving the conference call.

  For the outgoing portion of a conference call involving multiple voice terminals, the voice terminal which requested outside dial tone to bring an outside party into the conference is recorded as the calling party.

  For the outgoing/incoming portion of a conference call, the call duration in SMDR reflects the entire time the trunk was used on the call.

  Trunk-to-trunk transfer calls are treated like conference calls for SMDR purposes. A separate SMDR record is produced for each trunk used in a trunk-to-trunk transfer.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  Either the hunt group extension number or individual hunt group member extension number (depending on administration) is recorded as the called number.

- Direct Inward Dialing (DID)

  All incoming calls on the DID trunk group will be recorded.

- Foreign Exchange (FX) Trunks

  All calls made on an FX trunk group will be recorded.

- Hot Line Service

  The stored number used on an outgoing Hot Line call is recorded by SMDR the same as if it was manually dialed.

- Intercept Treatment

  If an outgoing or tandem call is routed to Intercept Treatment, the number dialed by the calling party is recorded as the dialed number, and condition code "F" is recorded.

- Intercom—Automatic

  No SMDR record is made of Automatic Intercom calls.

- Intercom—Dial

  No SMDR record is made of Dial Intercom calls.

- Inter-PBX Attendant Calls (V2, V3, or G1)

  If a user calls an Inter-PBX attendant and the trunk group used has SMDR assigned, SMDR records the following information:

  — Condition Code — A

  — Access Code Dialed — blank

  — Access Code Used — trunk access code of trunk used

  — Dialed Digits — Inter-PBX attendant access code.

- Integrated Services Digital Network (ISDN) (G1)

  When specific answer supervision is received from the network, an indication is sent to the SMDR device to this effect. If an ISDN call has been interworked, the SMDR record will not record the call as having answer supervision.

- Manual Originating Line Service

  If an attendant establishes an outgoing call for a voice terminal, designated as a Manual Originating Line, the SMDR record for the call will be the same as for any attendant-assisted outgoing call. The calling voice terminal extension number is recorded as the calling number, and Condition Code 1 applies.

- Multiple Listed Directory Numbers (LDNs)

  If incoming call information is recorded, the called number recorded for LDN calls is the extension number or trunk group access code to which the attendant completes the call. If the call terminates at the attendant console only, the called number recorded is 0, which is used to identify the attendants.

- Night Service—Night Station

  The extension number assigned to the attendants (0) is recorded as the dialed number.

- Night Service—Trunk Answer From Any Station

  The extension number assigned to the attendants (0) is recorded as the dialed number.

- Off-Premises Station

  SMDR data is recorded if the voice terminal is involved in an outgoing/incoming trunk call.

- Personal Central Office Line Group (PCOLG)

  An outgoing PCOLG call will be recorded as a call from the originating extension number via the trunk group associated with the PCOLG. The answering voice terminal's primary extension is recorded as the called number if incoming calls are recorded.

- Private Network Access

  Private Network Access calls will be recorded.

- Remote Access

  Remote Access calls will be recorded if Remote Access is provided on a per trunk group basis.

- Ringback Queuing

  Condition Code 8 is recorded for an outgoing call which is queued for a trunk before completion. The length of time the call is queued will not be recorded.

  When an outgoing call is queued for a trunk and is unsuccessful (the queue times out or the calling party does not answer the callback) an SMDR record is not generated for the call.

- Tandem Tie Trunk Switching

  The calling party on an incoming trunk can dial the SMDR account code. The calling number field in SMDR is the trunk access code for the incoming trunk group, the called number is the number dialed.

- Temporary Bridged Appearance

  An SMDR record is not affected by any second or subsequent voice terminal bridging a call.

- Tie Trunk Access

  Tie trunk calls will be recorded.

- Transfer

  If a user originates a call on an outgoing trunk and then transfers the call to another voice terminal, the originating voice terminal will be recorded as the calling party.

  If a voice terminal user receives a call on an incoming trunk and then transfers the call to another extension, the extension that originally received the call is recorded as the dialed number,

- Trunk-to-Trunk Transfer

  Although they are not really conference calls, Trunk-to-Trunk Transfer connections are treated as such for SMDR purposes. A separate SMDR record is generated for each trunk in the connection.

  Unanswered Trunk Calls may or may not be recorded depending on administration. Each trunk group can be administered so that unanswered calls will be recorded if they remain unanswered for a specified period of time.

  Uniform Dial Plan

  If one user calls another user via a Uniform Dial Plan extension number, and the trunk group used has SMDR assigned, SMDR records the following information:

  — Condition Code — 7

  — Access Code Dialed — blank

  — Access Code Used — trunk access code of trunk used

  — Dialed Digits — Uniform Dial Plan extension.

- Wide Area Telecommunications Service (WATS) and 800 Service

  Calls made on a WATS or 800 Service trunk group are recorded if SMDR is assigned to the trunk group.

Administration

SMDR is administered by the System Manager. The following items can be administered

System Parameters

- Type of SMDR output device to be used. With G1, the type of output device must be assigned for both the primary and secondary output device, if both the primary and secondary ports are used.

- Extension number assigned to the output device. With G1, the extension number must be assigned for both the primary and secondary output device. Before the extension number is assigned, the System Manager should check to make sure that it is not already assigned as a Property Management System (PMS) (V3 or G1) extension or a Permanent Switched Call (PCS) extension.
- Whether standard or ISDN formats are used (G1).
- Printer paper width (80 or 132 columns) if a printer is the output device (V1, V2, or V3).
- SMDR account code length (from 1 to 15), the system defaults to 2 digits.
- The speed at which the SMDR device connected to the direct RS-232C interface on the processor circuit pack will operate (300, 1200, 2400, 4800, or 9600 baud rate) (XEV2, XEV3, V3, and G1).
- Whether the reason for disconnect is recorded instead of the FRL.
- Whether an account code is required on a toll call.
- Whether the hunt group extension or the hunt group member extension is recorded by SMDR.
- SMDR can be suppressed for Ineffective Call Attempts or for All Calls Excluding Outgoing Calls; system defaults to "no". Ineffective call attempts are call originated by a voice terminal user that are blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy. With G1, ineffective call attempts include calls to incoming or outgoing trunks that are unavailable due to trunk usage allocation for ISDN Call By Call Service Selection trunks and incoming calls rejected by the switch due to Network Specific Facility (NSF) mismatch.

Date and Time

The date and time should always be updated for events such as a leap year, daylight savings time, or a system restart after a power failure. If a time of day is not administered, SMDR records will not be generated.

Trunks, Loudspeaker Paging, and Code Calling Access

SMDR can be assigned to all trunk groups, Loudspeaker Paging Access trunks, and Personal Central Office Line trunks. The system defaults to yes for SMDR. The System Manager must determine which types of trunks will be assigned SMDR.

Class of Restriction

With Version 2, Version 3, or DEFINITY Generic 1, specify if SMDR account code entry is forced.

Feature Access Codes

Assign SMDR account code access code. The system defaults to *6.

IXC Codes

IXC access numbers

Name of IXC (optional).

Modules and Modems

With V1, V2, and V3, the SMDR output device can be connected to a Processor Data Module (PDM), Trunk Data Module, or a Modem. With G1, one or both of the SMDR output devices can be connected to a Processor Data Module (PDM), Trunk Data Module, or a Modem. The following items must be administered if SMDR is not connected to an AP.

- A data-channel numbered 01 to 04 (V1) or netcon channel (V2, V3, or G1) must be assigned using a data module form and entering data-channel or netcom channel for the type. This channel provides a path for SMDR data from the Switch Processing Element to the time-division bus.
- If the SMDR output device is connected to a PDM, administer a PDM form.
- If the SMDR output device is connected to a Trunk Data Module, administer a Trunk Data Module form.
- If the SMDR output device is connected to a 212A-type modem, a 2500 Voice Terminal form and Pooled Modem form must be completed. This allows circuit switched data connections between digital data communications equipment (data modules) and analog data communications equipment (modems).

If the SMDR output device is connected to an AP, the following forms must be administered (V2 or V3).

- Data Module form administered as an Interface 3 module. This is used with the TN719 Interface 3 circuit pack.
- Processor Data Module
- Interface Link
- Processor Channel.

If the EIA port on the Processor Interface circuit pack is used by the output device, the SMDR output device extension should be administered as "eia".

Hardware and Software Requirements

Hardware requirements depend on the type of output device used for SMDR. With XEV2, XEV3, V3, and G1, the SMDR output device can be connected directly to the processor circuit pack which provides a standard RS-232C interface. This eliminates the need for an MPDM or MTDM as described for the output devices as follows:

- If the output device is a printer, personal computer, tape unit, or the TELESEER SMDR unit (Data Terminal Equipment), the interface equipment consists of either a PDM to a port on a TN754 or TN784 (G1) Digital Line circuit pack or a 212A-type modem to a port on a TN742 Analog Line circuit pack. In the latter case, a standard modem pool facility is required for the data path.

- If the output device is the 94A LSU (Digital Communications Equipment), the interface equipment consists of either a Trunk Data Module to a port on a TN754 or TN784 (G1) Digital Line circuit pack or a 212A-type modem to a port on a TN742 Analog Line circuit pack. In the latter case, a modem pool facility is also required.

- If SMDR is connected to a host computer, the computer must be connected over a private line terminated at the switch with a Trunk Data Module.

- If SMDR is used with the AP (not available with XEV2, or G1), a TN716 Interface 1, a TN720 (V1) or TN738 (V2 or V3) Interface 2, and a TN719 Interface 3 circuit pack must be installed and connected. XEV3 requires a TN765 Processor Interface circuit pack.

- With V2, V3, or G1, a TN726 Data Line circuit pack can be used in conjunction with an Asynchronous Data Unit (ADU) to connect a 94A LSU, TELESEER SMDR unit, or printer.

For V2, V3, or G1, forced entry of account codes software is required.

Straightforward Outward Completion

Description

Allows an attendant to complete an outgoing trunk call for a voice terminal user, without requiring the voice terminal user to hang up.

Considerations

With Straightforward Outward Completion the attendant determines which calls should be allowed and can select the trunk group used for the call.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Subnet Trunking (V2, V3, or G1)

Description

Provides modification of the dialed number so an Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) call can route over trunk groups that terminate in switches with different dial plans.

Subnet Trunking provides digit insertion, deletion, pauses, and/or wait for dial tone in digit outpulsing, as required, to permit calls to route:

- To or through a remote switch
- Over tie trunks to private network switch
- Over central office (CO) trunks to the serving CO

All AAR and ARS calls ultimately reach a point where they can no longer route on a private network. That is, the call reaches a point where another on-network switch is not available for the call. (In an ARS standalone configuration, this is the originating switch.) Assuming the call is not denied at this point, then the call must route to one of the following:

- Directly to a party at the local switch.
- To a party at a remote switch (without accessing the public network).
- Through a remote switch to a party at a subtending location (without accessing the public network).
- Directly to a Wide Area Telecommunications Service (WATS) serving office.
- Directly to a local CO or a foreign exchange (FX) CO, both of which may or may not provide dial access to a long-distance carrier. (The alternative to dial access is for the CO to automatically provide access to a single long-distance carrier of the subscriber's choice.)
- Through a remote switch to the local or FX CO serving the remote switch.
- Through a remote switch to the WATS office serving the remote switch.
- To an Enhanced Private Switched Communications Service (EPSCS), Common Control Switching Arrangement (CCSA) office, or Electronic Tandem Network (ETN) office.

Subnet Trunking is not required on calls terminating directly to a party at the local switch. AAR handles these calls.

Subnet Trunking is required on calls routing to or through a remote switch, regardless of the call's destination.

With direct access to a WATS, EPSCS, CCSA, or ETN office, Subnet Trunking is not normally used. The called number on these types of calls is not normally modified. Subnet Trunking is needed only if the number is modified or if the call passes through some intermediate switch, such as a main.

Calls accessing a local or FX CO directly from the terminating switch normally require Subnet Trunking only if access to a long-distance carrier is other than the carrier automatically provided by the CO. In this case, the appropriate dial access code is inserted into the digit string by the system.

Aside from the normal cases, Subnet Trunking can be used to provide added functionality to the system, for example, to convert an AAR number into an international number. Also, Subnet Trunking can modify a digit string so that a Remote Access trunk group can be used on calls. This capability is called equivalent Direct inward Dialing (DID) and may be useful when a location has Remote Access but does not have DID or Network Inward Dialing (NID). (NID is the private network equivalent of DID.)

Addition or deletion of an Area Code on an ARS call does not require Subnet Trunking. ARS handles it via code conversion, as required.

With AAR, an on-network number can be converted into a public network number. In this case, the conversion may include an Area Code insertion via Subnet Trunking.

Any of three special characters may be used with Subnet Trunking:

- Pause—Delays outpulsing of subsequent digits for 1.5 seconds.
- Wait—Delays outpulsing of subsequent digits for a preprogrammed interval (from 5 to 25 seconds) or, if TN748B Tone Detectors are provided, until dial tone is received from the distant switch or the interval expires, whichever occurs first.
- Convert-to-tone—Causes all remaining digits to be outpulsed using tone signaling.

Use of these special characters is described in the following paragraphs.

During outpulsing of a digit string, it may be necessary to pause or wait for the distant switch to act upon the digits already sent. A programmed pause (a "," symbol) is used when the required action by the distant switch occurs within 1.5 seconds. Multiple pauses can be used. A programmed wait (a "+" symbol) is used to specify a longer interval. Receipt of dial tone automatically cancels the remainder of an interval when TN748B Tone Detectors are provided. If a dial tone detector is not available on a given call, the system uses the wait interval to determine when to resume outpulsing. Multiple waits can be used.

The type of outpulsing, either dial pulse (rotary) or tone, used on a call is specified by the trunk group selected for the call. In some cases, it may be necessary to assure that a portion of the digits are sent using tone signaling. The convert-to-tone character (a "%" symbol) is used to indicate that all digits remaining in the string to be outpulsed will use tone signaling.

Digit deletion always begins with the first digit. Subnet Trunking can delete up to 11 digits and can insert up to 36 digits. The last four digits dialed are normally retained. Thus, the new digit string can be up to 40 digits long. Typical uses of digit insertion are the conversion of an AAR call to an international call and the insertion of a long-distance carrier code, 10xxx, on a domestic call.

The insertion of a long-distance carrier access code in the string of digits to be outpulsed does not usually require a pause or wait symbol. Interconnecting offices, other than crossbar offices, can handle the code and the called number as a single string. However, a crossbar office returns dial tone after receiving the long-distance carrier code. Thus, a pause or wait is required between the long-distance carrier code and the called number.

Considerations

Subnet Trunking allows AAR and ARS calls to access the public network. With AAR, the major advantage is that the call continues although no on-network routes are available to handle the call. With ARS, the major advantage is that calls destined for the public network can route partially over the private network, if there is one. This saves toll charges for a portion of the call.

It is not necessary to include the trunk access code for the trunk group connecting to the distant switch in the string of digits to be outpulsed. In fact, such inclusion must be avoided. Access to the interconnecting trunk group is automatic. Outpulsing the access code, therefore, serves no purpose, and will cause mishandling of the call at the distant end.

The wait interval is a System Parameter. This interval can be from 5 to 25 seconds (in increments of 1 second).

Up to four special characters can be included in a string of digits to be outpulsed. Each special character counts as two digits.

Interactions

Subnet Trunking is a function associated with the AAR and ARS features. Interactions are the same as those given for AAR and ARS.

Administration

Subnet Trunking is set by the System Manager as a part of AAR and/or ARS administration. The following items require administration:

- Wait—Specify the wait interval used with Subnet Trunking.
- Routing Pattern—Specify the number of digits to delete (beginning with the first digit) and the specific string of digits to insert. Special characters, if any, are included in the inserted string.

Hardware and Software Requirements

No additional hardware is required.

Private Network Access or ARS software is required for Subnet Trunking.

System Measurements

Description

Provides reports on trunk group usage, hunt group usage and efficiency, attendant group activity and efficiency, and security violations.

Group reports are ail on a clock-hour basis. Reports available are yesterday's peak usage, today’s peak usage, and last hour’s usage. The peak usage hour is simply the clock-hour the group received the most usage. This hour can be different for different groups, such as hunt group 15 and hunt group 11. Conversely, today’s peak hour usage may be the same as the last hour’s usage.

Individual reports are available for each of the following:

- Trunk group—Yesterday’s peak usage
- Trunk group—Today’s peak usage
- Trunk group—Last hour’s usage
- Hunt group—Yesterday’s peak usage
- Hunt group—Today’s peak usage
- Hunt group—Last hour’s usage
- Attendant group—Yesterday’s peak, today’s peak, and last hour’s usage are all one report.

Security violations are attempts to access the system via an invalid login or Remote Access Barrier code. These violations are accumulated from the time at which the count was reset. (The System Manager performs this function.)

In addition to the trunk group, hunt group, attendant group, and security violation reports already described, Version 2, Version 3, and DEFINITY Generic 1 systems offer the following reports:

- Automatic Route Selection (ARS) Pattern Measurement

  A maximum of 20 ARS patterns can be measured, as specified by the customer.

- Trunk Outage Measurements

  Provides measurements on the four trunk groups which were out of service the most during the measurement period. Separate reports are available for yesterday, today, and the last hour.

- Trunk Lightly Used Measurements

  Provides measurements on the five trunks in each trunk group that have carried the least calls. Separate reports are available for yesterday, today, and the last hour.

- Modem Pool Measurements

  Provides traffic data for Modem Pool groups. Separate reports are available for yesterday's peak usage hour, today's peak usage hour, and the last hour.

- Attendant Group Performance

  Provides measurements on time taken by attendants to answer calls. Separate reports are available for any 8-hour period, yesterday or today.

- Hunt Group Performance

  Provides data on the longest time taken by a group member to answer a call for each hunt group. Separate reports are available for yesterday and today.

- Trunk Group Performance

  Provides data on calls that attempt to access trunk groups but cannot because all trunks are busy or the queue is full. Separate reports are available for yesterday and today.

- Summary Performance Report

  Provides a summary of data found in the ARS Pattern Measurements, Trunk Outage Measurements, Trunk Lightly Used Measurements, and Trunk Group Performance reports. Separate reports are available for yesterday and today.

- Automatic Circuit Assurance Report

  Provide data on referral calls that are generated by the system to notify the attendant or designated user of trunk members that exceed a long or short holding time threshold. The report shows an ongoing record.

All reports are on-demand reports. None are given automatically. Reports are available on the System Access Terminal (SAT) (V1, V2, and V3), the Manager I terminal (G1) or a remote administration terminal. The reports can be printed if a printer is associated with the terminal.

Considerations

Reports provided by System Measurements contain data that is useful to determine group efficiency. Details of specific items on the reports, as well as guidelines to use the data provided, are given in DEFINITY *Communications* System *Generic 1 and System 75— Administration and Measurement Reports 555-200-500.*

Traffic measurements are automatically accumulated by the system and are available on demand. However, reports are not archived. If needed, reports must be requested periodically. Obtaining a printed copy can aid in maintaining a history of the system traffic.

Detailed information of each call handled by a specific trunk group, if required, must be provided by the Station Message Detail Recording (SMDR) feature. Processed SMDR data can also provide detailed information on trunk group usage. However, if individual call details are not required for bill-back or cost-allocation, System Measurements should be considered as the means to determine and maintain trunk group efficiency.

Interactions

The Call by Call Service Selection feature, described previously in this chapter, enhances System Measurements by allowing them to be displayed on a per-service basis.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

System Status Report (V2, V3, or G1)

Description

Allows the user to view data associated with attendants, major and minor alarms, and traffic measurements. The information is displayed on the System Access Terminal (SAT), or Manager I terminal (G1), and presents a basic picture of the System 75 condition. The report can only be displayed by the System Manager and maintenance personnel.

The Status Report is displayed by entering one of the commands listed below. Once the command is entered, the system continually displays the report until it is canceled. Once the report is canceled, the user is automatically logged off the system.

- **monitor system view1**

  This command displays the following information:

  — Activation status of all attendants (updated every minute)

  — Maintenance status which includes major and minor alarms for trunk ports, terminal ports, and all maintained objects in the system except terminals and trunks (updated every minute)

  — Traffic measurements for trunk groups, hunt groups, and attendant groups (updated every hour)

- **monitor system view2**

  This report is a subset of the view1 report and displays the same information listed for the view1 report except the last hour's measurement for the hunt groups.

- **monitor traffic trunk-groups (V3 and G1 only) (updated every minute)**

  This command displays the following information:

  — Trunk group number

  — Number of members in each trunk group

  — Number of members in each trunk group that are active on a call

  — Length of group queue

  — Number of calls waiting in the group queue

- **monitor traffic hunt-group (V3 and G1 only) (updated every minute)**

  This command displays the following information:

  — Hunt group number

— Number of members in each hunt group

— Number of members in each hunt group that are active on a call

— Length of group queue

— Number of calls waiting in the group queue

— Length of time the oldest call in queue has been waiting to be serviced.

When a central office (CO) call enters a full Automatic Call Distribution (ACD) split queue, the system and the CMS may show different measurements. The system measurements indicate the maximum number of calls allowed in the queue, whereas the CMS measurements indicate all calls in the queue plus any call on the CO trunk waiting to enter the split queue.

Considerations

In addition to providing status reports, this feature also provides an indication that the administration terminal is functioning. Any attempt to stop the "monitor system view1/view2 reports logs the administration terminal off the system. Therefore, no unauthorized administration can be performed.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Temporary Bridged Appearance

Description

Allows multi-appearance voice terminal users in a Terminating Extension Group or Personal Central Office Line Group to bridge onto an existing group call. If a call has been answered using the Call Pickup feature, the originally called patty can bridge onto the call. Also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it.

A call incoming to a Terminating Extension Group or Personal Central Office Line Group is not a call to an individual, although one particular member of the group may be the most qualified person to handle the given call. If this individual did not anser the call originally, then he or she can simply bridge onto the call. The call does not have to be transferred.

A call to an individual can be answered by a Call Pickup group member. If the called person returns while the call is still connected, he or she simply bridges onto the call and the answering party hangs up.

Call Coverage provides redirection of calls to alternate answering positions (covering users). A Temporary Bridged Appearance is maintained at the called voice terminal.

The called party can answer the call at any time, even if already answered by a covering user. If the called party does not bridge onto the call, the covering user can use the Consult function of Call Coverage to determine if the called party wants to accept the call. The Consult function uses the Temporary Bridged Appearance maintained on the call. When the consult call is finished, the Temporary Bridged Appearance is removed.

Stations that normally would have a temporary bridged appearance with their coverage point will not, if the coverage point is Audio Information Exchange (AUDIX).

Considerations

Temporary Bridged Appearance permits the desired party to bridge onto a call without manually transferring the call. This provides convenience of operation and also saves time.

Temporary Bridged Appearance does not provide any call originating capability or the capability to answer another party’s calls. These capabilities are provided by the Bridged Call Appearance feature.

Interactions

Privacy—Manual Exclusion, when activated, prevents other users from bridging onto a call A user who attempts to bridge onto a call with the Privacy—Manual Exclusion feature active will be dropped.

The Bridged Call Appearance feature enhances Temporary Bridged Appearance by allowing more than one call to an extension to be bridged and by allowing calls to be originated from bridged appearances.

Calls redirected to Call Coverage maintain a Temporary Bridged Appearance on the called voice terminal if a call appearance is available to handle the call. The called party can bridge onto the call at any time. With V3 and G1, the system can be administered to allow a temporary bridged appearance of the call to either remain at or be removed from the covering voice terminal after the principal bridges onto the call. With V1 and V2, a temporary bridged appearance automatically remains at the covering voice terminal after the principal bridges onto the call.

Consult calls use the Temporary Bridged Appearance maintained on the call. At the conclusion of a consult call, the bridged appearance is no longer maintained. If the principal chooses not to talk with the calling party, the principal cannot bridge onto the call later.

If a call has, or has had, a Temporary Bridged Appearance, is conference or transferred, and redirects to coverage again, a Temporary Bridged Appearance is not maintained at the conferenced-to or transferred-to extension.

Administration

The only required administration is to administer whether or not a temporary bridged appearance is maintained by the covering user after the principal bridges onto the call. (Keep Held SBA at Coverage Point field on Feature-Related System Parameters screen form) (V3 and G1 only)

Hardware and Software Requirements

No additional hardware or software is required.

Ten-Digit to Seven-Digit Conversion (G1)

Description

The Ten-Digit to Seven-Digit Conversion feature allows customers to dial certain 10-digit and 7-digit public network telephone numbers and have the numbers converted to 7-digit private network telephone numbers. This allows the calls to route partially or completely over private network trunks. Using the private network trunks can significantly reduce long-distance telephone bills.

This feature can be used by the following users:

- Single Line Stations
- Multi-appearance Stations
- Attendants
- Remote Access
- Incoming Tie Trunks
- Terminal Dialing Data Endpoints.

The feature gives the system the ability to recognize certain 7-digit and 10-digit Direct Distance Dialing numbers as the numbers of 7-digit Electronic Tandem Network (ETN) location (RNX) numbers. The 7-digit or 10-digit numbers will be replaced by 7-digit ETN numbers. Then, these calls can be routed over tie trunks or terminated at an internal station.

Ten-Digit to Seven-Digit Conversion is an optional capability that works with the Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) features.

Automatic Alternate Routing (AAR)

AAR provides alternate routing choices for private on-network calls. It also allows on-network calls to route through the public network when on-network routes are not available. The routing order is specified by the customer by translating a group of trunks into a routing pattern, which is used for calls to a particular network location. For a detailed description of this feature, see Automatic Alternate Routing elsewhere in this chapter.

Automatic Route Selection (ARS)

The ARS and AAR features are closely related. ARS routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed. AAR routes calls to private networks, and can also route calls through the public network when private network routes are not available. For a detailed description of this feature, see Automatic Route Selection elsewhere in this chapter.

Direct Distance Dialing (DDD) Numbers

A DOD number is a public network number with three parts:

- A 3-digit area code
- A 3-digit Central Office (Exchange) code
- A 4-digit extension (endpoint) number.

Electronic Tandem Network (ETN)

An ETN is a group of PBXs, connected by tie trunks, that share a common Dial Plan and Uniform Numbering Plan. Automatic Alternate Routing controls call routing over an ETN. For a detailed description of an ETN, see Chapter 2 of this manual.

Feature Operation

If the AAR or ARS access code is dialed, the type of call is examined. The following call types are handled by Ten-Digit to Seven-Digit Conversion:

- Local Calls

  Seven-digit ARS calls to a Central Office (Exchange) in the Home Area Code preceded by the digit 1, if required.

- Ten-Digit Direct Distance Dialing Calls

  The digit 1 (if required), an Area code, and Exchange code, and a station number.

- Calls to Inter-exchange Carriers

  A Carrier Access Code, followed by one of the two call types just described. Carrier Access Codes are of the form 10XXX, where XXXs are digits to identify a particular long-distance carrier.

Ten-digit DOD numbers will be replaced by ETN numbers if the first six or more digits of the DDD number match a number in the Ten-Digit to Seven-Digit Conversion Table (see Table 3-I). If a conversion takes place, then the call is routed through AAR using the converted string; otherwise, the call is routed through ARS.

For purposes of Ten-Digit to Seven-Digit Conversion, if a 7-digit number is dialed after the ARS access code, the call is treated as if the home area code of the switch was dialed. However, if a 7-digit number is dialed after the AAR access code, the call is treated as if it was a 7-digit private network call. Ten-digit calls are subject to conversion if either the AAR or ARS access code is dialed. Seven-digit AAR calls are not subject to conversion. Seven-digit ARS calls are subject to conversion with the Home Area Code of the system inserted.

Table 3-1. Ten-Digit to Seven-Digit Conversion Table Example

TEN-DIGIT TO SEVEN-DIGIT CONVERSION TABLE	
Match	Replace
...	...
201-957-5XXX	222-5XXX
...	...
212-976-1313	0
212-976-2525	222-0111
...	...
303-526-XXXX	362-XXXX
303-538-2XXX	374-3XXX
303-538-3XXX	374-0XXX
...	...
...	...
303-538-56XX	374-59XX
...	...
...	...
303-538-975X	374-970X
...	...
...	...
303-539-4901	374-4901
303-539-6212	374-0111
...	...
...	...

Note: The table has a maximum of 180 entries, and the columns are defined:

- Match—The digits or string to be checked against the dialed number. This digit string will be six to ten digits in length. The number of digits in this pattern is equal to the number of digits to be deleted from the dialed number. All entries in this field must begin with an Area Code for a match to occur, including entries where the Area Code is the Home Area Code of the switch. This string should always be three digits greater in length than the corresponding "Replace" string to avoid receiving intercept treatment.

- Replace—The digit string that will replace the "Match" string if a match occurs.

Conversion Examples

The following assumptions are made for the examples:

- The ARS access code is the digit "9".
- The Ten-Digit to Seven-Digit Conversion Table is administered as in Table 3-1.
- The Home Area Code of the switch is "201".
- The home RNX of the switch is "222".
- A prefix of "1" is required for all DDD calls.

Examples:

1. Routing over the ETN to another switch—

   For instance, the number 9-1-303-538-2100 is the Listed Directory Number of a branch office in Denver. If a user dials the number, the system checks the Conversion Table, and in this case finds a match and replaces the number with 374-3100. Then the routing pattern associated with the location code (RNX) 374 is used to route the call.

2. Routing to the same switch—

   Usually when a user dials a number that belongs to a station within the same system, only the extension number need be dialed. However, the user may not know that the number terminates on the same system. Also, the system could serve more than one company with different Listed Directory Numbers or a company could have several systems.

   In any case, if a user dials a number that belongs to a station with the same system, for instance 9-957-5300 or 9-1-201-957-5300, the system checks the Conversion Table, finds a match, and replaces the number with 222-5300. Then the call is routed within the same system.

3. Routing to Intercept, an Attendant, or an Announcement—

   There may be DDD numbers that you want to block or keep unauthorized people from using, such as a number to access your computer system. These calls may be routed to intercept, to an attendant, or to an announcement.

   - Routing to Intercept—For instance, if a user dials 9-1-212-976-1313, the system checks the Conversion Table, finds a match, and replaces the number with the digit 0. Since this is an invalid ETN number, the call will receive intercept treatment. In general, the "Match" string must be at least six digits long and have exactly three more digits than the corresponding "Replace" string. A "Replace" string of one or two digits will always result in an invalid ETN number.

- Routing to an Attendant—If a user dials 9-1-212-976-2525, the system checks the Conversion Table, finds a match, and replaces the number with the switch's attendant number 222-0111.
- Routing to an Announcement—As in the other examples, a dialed number may be routed to an announcement if the "Replace" string corresponds to an announcement extension number.

Considerations

The Conversion Table can have up to 180 entries.

All ARS/AAR feature limitations also apply to the Conversion feature since this feature is accessed when control is passed to ARS directly or by AAR.

The AAR access code is normally the digit 8, and the ARS code is normally the digit 9. Two different ARS access codes can be assigned.

If a dialed number is converted, then it becomes "trapped" by AAR. That is, if for any reason, AAR cannot complete the call, then the call will not be returned to ARS. It is just this feature which allows unauthorized calls to be blocked by Ten-Digit to Seven-Digit Conversion. However, it is possible that a call cannot complete because of congested tie trunks while there is an ample supply of idle CO trunks. This situation can be avoided by including an outgoing trunk in the routing pattern and by using the digit deletion/insertion feature of AAR. This allows AAR to recover the original number before outpulsing.

Interactions

The following features interact with the Ten-Digit to Seven-Digit Conversion feature.

- Automatic Alternate Routing (AAR)

  If an AAR access code is dialed, followed by the prefix digit "1" (if required) and a 10-digit DDD or IDDD number, AAR recognizes this as an off-net call. Control of the call is then transferred to ARS as if the calling party had dialed the ARS access code.

- Automatic Route Selection (ARS)

  Conversion begins by attempting to match the dialed number with the administered "Match" number, and if a match is found, proceeds with conversion. If a match is not found, the call is routed through ARS.

- AAR/ARS Partitioning

  If AAR/ARS Partitioning is enabled and Time of Day Routing is not, then the Partition Group Number of the originating party determines which Routing Plan is in effect for a given user.

- Time of Day Routing

  If Time of Day Routing is enabled, then the Time of Day Routing Tables determine which routing plan is in effect for a given user.

- Station Message Detail Recording (SMDR)

  Even if a call is altered by Ten-Digit to Seven-Digit Conversion, the actual digits dialed are recorded by SMDR.

  SMDR generates a record only if a trunk is seized If a call is converted by Ten-Digit to Seven-Digit Conversion and routed within the local switch, then a trunk is not seized and an SMDR record is not generated.

Administration

The only administrable item unique to this Conversion feature is the Conversion Table (see Table 3-I). The table has a maximum of 180 entries.

The administrator does not need to make entries to the Conversion Table in numerical order. The system automatically sorts the "Match" field.

The Conversion Table is only available to an administrator if ARS is activated and either the Private Network or Uniform Dial Plan is activated.

Hardware and Software Requirements

No additional hardware is required.

ARS software and either Private Network or Uniform Dial Plan software is required.

Terminating Extension Group

Description

Allows an incoming call to ring (either audible or silent alerting) as many as four voice terminals at one time. Any user in the group can answer the call.

Any voice terminal can be administered as a Terminating Extension Group (TEG) member; however, only a multi-appearance voice terminal can be assigned a TEG button with associated status lamp. The TEG button allows the user to select a TEG call appearance for answering or for bridging onto an existing call but not for call origination.

When an incoming call is answered by a TEG member, a Temporary Bridged Appearance is maintained at the multi-appearance voice terminals in the group. However, the Temporary Bridged Appearance is not visible on a call appearance. Any of the TEG members can bridge onto the call by pressing the TEG button, if assigned. For example, suppose an incoming call has been answered by a certain TEG member, and this TEG member does not have the needed information. If another member has the needed information, that member needs only to bridge onto the call to provide the information.

The Privacy—Manual Exclusion feature can be assigned to any or all of the multi-appearance voice terminals in a TEG. This allows the answering TEG member, by pressing the Exclusion button, to prohibit bridging by other group members. Pressing the button again reestablishes the bridging capability.

A single-line voice terminal administered as a TEG member is rung for a TEG call if it is idle.

A TEG is established by associating the individual member’s extension number with a TEG extension number. The members have call placing and receiving privileges for their individual extension numbers, as defined by the assigned Class of Restriction (COR). Each TEG is also assigned a COR. The group COR overrides an individual member’s COR on calls to the group. Thus, the members could be Termination Restricted, but still receive TEG calls.

Considerations

TEGs are useful when it is desirable to have incoming calls to a specific extension number ring more than one voice terminal simultaneously. For example, the appliance department of a large retailer might have three voice terminals. Anyone in the department can anser the call. The salesperson most qualified to handle the call can bridge onto the call from either of the other two voice terminals.

The system allows for as many as 32 TEGs with up to 4 members each. A voice terminal user can be a member of more than one TEG, but can have only one TEG button for each group.

A TEG can only handle one TEG call at a time. If any member of a TEG is active on a call to the TEG, a second call to the TEG waits until the first call is terminated before it rings the TEG. The TEG members have no way to know when a TEG call is waiting. If a coverage path is assigned to the TEG, the waiting call routes accordingly.

Interactions

The following features interact with the Terminating Extension Group feature.

- Automatic Callback

  This feature cannot be activated for a TEG.

- Bridged Call Appearance

  Calls to a TEG cannot be bridged, except via a Temporary Bridged Appearance.

- Call Forwarding All Calls (V1)

  A TEG call cannot be forwarded on V1 systems.

- Call Park

  A TEG call cannot be parked on the group extension number; however, a group member answering a call can park such a call on his or her own extension number.

- Direct Department Calling and Uniform Call Distribution

  A TEG cannot be a member of a Direct Department Calling or Uniform Call Distribution group.

- Call Coverage

  Calls to a TEG can be redirected to alternate answering positions whenever the Call Coverage feature is assigned and no group member is available to answer the tail. If any member of a TEG is active on a TEG call, all subsequent TEG calls redirect to coverage. However, a TEG cannot serve as an alternate answering position. In other words, a TEG can have a Call Coverage path assigned, but cannot be a point in a Call Coverage path.

  A Send Term button for the TEG can be assigned to any or all group members who have multi-appearance voice terminals. When the Send Term button is pressed, all calls to that TEG redirect to coverage. The associated status lamp lights on the activating voice terminal and all other voice terminals with a Send Term button. Any member with a Send Term button can deactivate Send Term by pressing the button. The Send Term status lamp then goes dark on all voice terminals. Incoming calls are again directed to the group.

- Leave Word Calling

  Leave Word Calling messages can be stored for a TEG and can be retrieved by a member of the group, a covering user of the group, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization must be assigned to the message retriever. Also, a remote Automatic Message Waiting lamp an be assigned to a group member to provide a visual indication that a message has been stored for the group. One indicator is allowed per TEG.

- Temporary Bridged Appearance

  At multi-appearance voice terminals in the TEG, a Temporary Bridged Appearance is maintained after a call is answered. This allows other members of the group to bridge onto the call.

The Privacy—Manual Exclusion feature, when activated, prevents other TEG members from bridging onto a call. A TEG member who attempts to bridge onto a call with Privacy—Manual Exclusion activated will be dropped.

Administration

Terminating Extension Groups are administered by the System Manager. The following items require administration for each group:

- Group number (from 1 to 32)
- Extension number for the group
- Group name (for display purposes)
- Call Coverage path number
- Group COR
- Up to four group member extension numbers.

The following items can be administered to multi-appearance voice terminal TEG members:

- TEG button with associated status lamp.
- Exclusion button associated with the TEG extension number. (Keeps other group members from bridging onto an existing call.)
- Send Term button for the TEG extension number.
- Remote Automatic Message Waiting lamp (one per TEG extension number).
- Audible or silent alerting.

Hardware and Software Requirements

No additional hardware or software is required.

Time of Day

Description

Provides the most economical routing of Automatic Route Selection (ARS) and Automatic Alternate Routing (AAR) calls, based on the time of day and day of the week that each call is made.

With Time of Day Routing, a company can take advantage of lower calling rates during specific times of the day and week. In addition, companies with locations in different time zones may be able to maximize the use of facilities by utilizing those in a location that has a lower rate at different times of the day or week. This feature can also be used to change the patterns during the times an office is closed in order to reduce or eliminate unauthorized calls.

Time of Day Routing uses the Partition Group Numbers (PGNs) associated with the AAR/ARS Partitioning feature. A Time of Day Routing Plan (see Figure 3-59) can be administered for each of the eight PGNs. When a user makes an AAR/ARS call, the call is routed according to the Time of Day Routing Plan associated with that user's PGN.

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                                                              Page 1 of 1

                              TIME OF DAY ROUTING PLAN 1

         Act    Pln   Act    Pln   Act    Pln   Act    Pln   Act    Pln   Act    Pln
         Time   #     Time   #     Time   #     Time   #     Time   #     Time   #

  Sun    00:01  1     __:__  _     __:__  _     __:__  _     __:__  _     __:__  _
  Mon    00:01  1     __:__  _     __:__  _     __:__  _     __:__  _     __:__  _
  Tue    00:01  1     __:__  _     __:__  _     __:__  _     __:__  _     __:__  _
  Wed    00:01  1     __:__  _     __:__  _     __:__  _     __:__  _     __:__  _
  Thu    00:01  1     __:__  _     __:__  _     __:__  _     __:__  _     __:__  _
  Fri    00:01  1     __:__  _     __:__  _     __:__  _     __:__  _     __:__  _
  Sat    00:01  1     __:__  _     __:__  _     __:__  _     __:__  _     __:__  _
```

Figure 3-58. Screen Form Used for implementing Time of Day Routing

Time of Day Routing provides the flexibility to change the routing of outgoing calls as many as six times a day, each day of the week. Each of the six possible routing plans (from a pool or eight) is assigned an activation time and a Routing Plan Number (RPN) (shown as "Pin #" on the Manager I terminal screen form.) Each RPN has a complete and distinct group of

Home Numbering Plan Area (HNPA), Foreign Numbering Plan Area (FNPA), and location code (RNX) tables to be used to route AAR/ARS calls. When a particular RPN is activated, it remains in effect until the activation time of the next RPN or until it is overridden. Activation of a new RPN does not affect calls in progress.

When a user dials the AAR or ARS feature access code followed by the desired number, and the system has collected enough digits to search for the routing pattern, the system will then select one of the eight Time of Day Routing Plan tables based on the PGN assigned to the user's Class of Restriction (COR). Then, depending on the day of the week and the time of the day the call is being made, the system selects the RPN ("Pin #" on the Manager I screen form). The system then uses the HNPA, FNPA, or RNX table associated with this RPN to select the Routing Pattern to be used on the call.

When Time of Day Routing is activated, it applies to all outgoing calls from voice terminals, attendants, data terminals, remote access users, incoming tie trunks, Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI) trunks, and trunks used for call forwarding to external numbers.

For additional information on AAR, ARS, and AAR/ARS Partitioning, see the respective feature descriptions elsewhere in this chapter.

Time of Day Routing Example

Assume the following:

- Jim is the user at extension 1234
- Extension 1234 is assigned a COR of 2
- COR 2 is assigned a PGN of 3
- The Time of Day Routing Plan table for PGN 3 is administered as shown in Figure 3-59.

When Jim comes into work on Monday morning at 8:30 and at that time makes an ARS call (dials the ARS access code followed by the number of the person he is calling), the system will look at the PGN assigned to Jim's Class of Restriction to determine which Time of Day Routing Plan table is to be used.

Since Jim has a COR of 2 and COR 2 has a PGN of 3, the system will use Time of Day Routing Plan 3 to route the call.

According to Time of Day Routing Plan 3, all calls made between 8:00 a.m. and 12:00 p.m. route according to the HNPA, FNPA, and RNX tables associated with RPN 2. Therefore, these HNPA, FNPA, and RNX tables will be used to find a Routing Pattern for the call.

If Jim makes a call between 12:00 p.m. and 1:00 p.m. on Monday, the same Time of Day Routing Plan table (number 3) is used and the call is routed according to the HNPA, FNPA, and RNX associated with RPN 1.

Page 1 of 1

TIME OF DAY ROUTING PLAN 3

	Act Time	Pln #	Act Time	Pln #	Act Time	Pln #	Act Time	Pln #	Act Time	Pln #	Act Time	Pln #
Sun	00:01	1	__:__	_	__:__	_	__:__	_	__:__	_	__:__	_
Mon	00:01	1	08:00	2	12:00	1	13:00	2	17:00	1	__:__	_
Tue	00:01	1	08:00	2	12:00	1	13:00	2	17:00	1	__:__	_
Wed	00:01	1	08:00	2	12:00	1	13:00	2	17:00	1	__:__	_
Thu	00:01	1	08:00	2	12:00	1	13:00	2	17:00	1	__:__	_
Fri	00:01	1	08:00	2	12:00	1	13:00	2	17:00	1	__:__	_
Sat	00:01	1	__:__	_	__:__	_	__:__	_	__:__	_	__:__	_

Figure 3-59. Screen Form Used for Time of Day Routing Example

Overriding the Time of Day Routing Plan

An attendant or a voice terminal user with console permission and a display can temporarily override the routing plan that is currently in effect for the activating user's PGN. This can be accomplished by either of two methods:

- Immediate Manual Override
- Clocked Manual Override.

Both types of override are described in detail in the following paragraphs. It should be noted that both types of override cannot be activated simultaneously. If either type is activated while the other is still in effect, the newly activated override goes into effect and the other override is automatically deactivated.

There is no indication via the Manager I terminal that either type of override has been activated. Also, since these overrides are temporary, they are not saved to translations during a "save translations". This way, the overrides are not reactivated at a later time when the system reboots. Therefore, in the event of a system reset, these overrides are deactivated.

Immediate Manual Override

This type of override is button activated, takes place immedicately upon activation, and remains in effect for the activating user's PGN (and all who share this PGN) until it is manually deactivated or the next scheduled change in the Time of Day Routing Plan takes place.

When a user presses an idle immediate Manual Override button, the associated lamp flashes (unless the user is an attendant, in which case, the lamp will not flash) and the display shows:

OLD ROUTE PLAN: x ENTER NEW PLAN:

Note: x is the number of the routing plan currently in effect.

The user then enters an RPN (1 to 8) using the dial pad and the display updates to:

OLD ROUTE PLAN: x NEW PLAN: y

The user then presses the flashing Immediate Manual Override button or the Normal button. The Immediate Manual Override button lamp then lights steadily and the display is returned to the NORMAL mode. At this time, the old RPN (x) is deactivated and the new RPN (y) is in effect.

The override attempt is denied if any of the following occurs:

- The activating user is not an attendant or a voice terminal user with console permission and a display.
- The activating user enters anything other than 1 through 8 when prompted by the display for the new RPN.
- The activating user presses the flashing Immediate Manual Override button before entering a new RPN.
- The activating user enters another display mode (that is "Normal") before completing the attempt.

When Immediate Manual Override is activated by a user, the override is also in effect for all other users with the same COR PGN.

A user can deactivate the override by pressing the steadily lighted Immediate Manual Override button. When the override is deactivated, the scheduled routing plan goes into effect. The on/off status of the button lamp is tracked by all other users with the same COR PGN who have Immediate Manual Override buttons. Therefore, a user other than the activating user can deactivate the override.

Clocked Manual Override

This type of override requires the user to manually enter a specific day and time for activation and deactivation of the override. The override occurs at the specified day and time of activation and remains in effect until the specified day and time of. deactivation. When the override is deactivated, the normally scheduled routing plan goes into effect.

When a user presses an idle Clocked Manual Override button, the associated lamp flashes (unless the user is an attendant, in which case, the lamp will not flash) and the display shows:

ENTER ACTIVATION ROUTE PLAN, DAY & TIME

The user then uses the dial pad to enter an RPN (1 to 8), followed by the day (1 to 7, where "1" is for Sunday and "7" is for Saturday) and the activation time (0000 to 2359, military time). The display then updates to:

ROUTE PLAN: x FOR: yyy ACT-TIME: zz:zz

In the above display, x is the new RPN, yyy is a 3-letter abbreviation for the day of the week, and zz:zz is the activation time for the override.

The user then presses the flashing Clocked Manual Override button again, the lamp continues to flash, and the display shows:

ENTER DEACTIVATION DAY AND TIME

The user then uses the dial pad to enter the day (1 to 7, where "1" is for Sunday and "7" is for Saturday) and the activation time (0001 to 2400, military time). The display then updates to:

ROUTE PLAN: x FOR: yyy DEACT-TIME: zz:zz

In the above display, x is the new RPN, yyy is a 3-letter abbreviation for the day of the week, and zz:zz is the deactivation time for the override.

The user then presses the flashing Clocked Manual Override button or the Normal button. The Clocked Manual Override button lamp then lights steadily and the display is returned to the NORMAL mode. At the entered times and days, the new RPN (x) is activated and then deactivated.

The override attempt is denied if any of the following occurs:

- The activating user is not an attendant or a voice terminal user with console permission and a display.
- The activating user enters anything other than valid information when prompted by the display.
- The activating user enters another display mode (that is, normal) before completing the attempt.

When Clocked Manual Override is activated by a user, the override is also in effect for all other users with the same COR PGN.

A user can deactivate the override by pressing the steadily lighted Clocked Manual Override button. The on/off status of the button lamp is tracked by all other users with the same COR PGN who have Clocked Manual Override buttons. Therefore, a user other than the activating user can deactivate the override.

Considerations

Time of Day Routing enhances AAR, ARS, and AAR/ARS Partitioning by allowing companies to choose more economical call routing based on the day of the week and the time of the day.

Time of Day Routing provides up to eight different routing plans. The routing plan can be changed as many as six times a day, each day of the week. At least one time period and RPN must be assigned to each day of the week.

A maximum of ten Immediate Manual Override buttons and ten Clocked Manual Override buttons is allowed per PGN. These buttons can only be assigned to and used by attendants and voice terminal users with both console permission and a display. Each attendant console or voice terminal can be assigned a maximum of one Immediate Manual Override button and one Clocked Manual Override button.

Time of Day Routing can only be used if AAR/ARS Partitioning and AAR and/or ARS are provided.

Interactions

The following features interact with the Time of Day Routing feature.

- Abbreviated Dialing

  For Time of Day Routing purposes, a user's own COR PGN is used when accessing an Abbreviated Dialing privileged list. The call is processed the same as if the call had been dialed directly using AAR or ARS.

- Attendant Extended Calls

  When an attendant extends (places) a call for a station user or trunk and that call uses AAR or ARS to process the call, the call is routed according to the PGN of the attendant's COR.

- Authorization Codes

  If a user's Facility Restriction Level (FRL) has been changed through the use of an Authorization Code, the COR FRL associated with the entered Authorization Code will be used in routing pattern selection.

- Automatic Alternate Routing (AAR)

  When Time of Day Routing is assigned, all AAR calls use the Time of Day Routing Plans for routing calls.

- Automatic Route Selection (ARS)

  When Time of Day Routing is assigned, all ARS calls use the Time of Day Routing Plans for routing calls.

- Bridged Call Appearance

  The COR PGN of the primary extension applies to calls originated from a bridged call appearance of the primary extension.

- Call Forwarding All Calls

  If a user has activated Call Forwarding All Calls, and AAR or ARS is used to route an incoming call to the forwarded-to number, the COR PGN of the calling party is used to route the call.

- Distributed Communications System (DCS)

  Care should be taken when making Time of Day Routing assignments in a DCS environment. Depending on a user's PGN, a user may or may not be routed to a DCS trunk group. If a user is not routed to a DCS trunk group, feature transparency may be lost.

  When a call routes over a DCS trunk, the switch at the far end will route the call according to the COR PGN of the incoming trunk.

- Individual Attendant Access

  When an AAR/ARS call is made from an individual attendant (that is, not extending a call), the individual attendant's COR PGN is used for routing the call.

- Recent Change History

  Changes made to Time of Day Routing Plan charts, routing plans, routing patterns, trunk groups, and CORs are recorded by the Recent Change History feature.

- Remote Access

  When an AAR or ARS call is made via Remote Access, the COR PGN of the Barrier Code and/or Authorization Code that was entered is used for routing the call.

- Station Message Detail Recording (SMDR)

  Normal SMDR records are generated for AAR/ARS calls on trunks administered for SMDR. However, information about the PGN used to route the call is not provided.

- Uniform Dial Plan (UDP)

  Since the dialed digits of UDP calls are expanded into an RNX digit string, the AAR feature creates a potential for the use of different routing patterns. Once the call begins to be routed by AAR, the originating user's COR PGN is used to route the call.

Administration

Time of Day Routing must be activated on the System Parameter-Customer Options form. It can then be administered by the System Manager. The following items require administration:

- AAR/ARS Partitioning must be administered as well as AAR and/or ARS.
- Different FNPA, HNPA, and RNX tables must be administered for each PGN.
- A Time of Day Routing Plan must be administered for each PGN.
- A PGN must be assigned to each COR table. Up to eight PGNs can be used.
- Immediate Manual Override and Clocked Manual Override buttons must be administered in order to manually override the Time of Day Routing Plan. These buttons can only be assigned to and used by attendants and voice terminal users with both console permission and a display.

Hardware and Software Requirements

No additional hardware or software is required.

Through Dialing

Description

Allows the attendant to select an outgoing trunk for a voice terminal user. The attendant then releases from the connection, and the user completes the call.

The attendant can select a trunk by dialing an access code or by pressing a Trunk Group Select button. Also, the attendant can dial the Automatic Route Selection feature access code prior to releasing from the call.

Considerations

Through Dialing saves the attendant time by allowing the calling party to dial the called number.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Timed Reminder

Description

Automatically alerts the attendant after a predetermined time for the following types of calls:

- Extended calls waiting to be answered or waiting to be connected to a busy single-line voice terminal
- One-party incoming calls placed on hold on the console
- Incoming calls answered by a voice terminal user, but which are unanswered after being transferred.

The attendant can reenter the call and decide whether to terminate the call or permit the waiting to continue.

Considerations

Timed Reminder informs the attendant that a call requires additional attention. After the attendant reconnects to the call, the user can either choose to try another extension number, hang up, or continue to wait. This personal attention can help establish rapport with clients and customers.

The Timed Reminder intervals for calls waiting for connection and for calls placed on hold are assigned separately. Each interval can be from 10 seconds to 17 minutes.

Interactions

The following features interact with the Timed Reminder feature.

- Attendant Call Waiting

  An attendant-extended call to a busy single-line voice terminal will return to an attendant console if the Timed Reminder Interval expires before the call is answered, or redirects to coverage.

- Call Coverage

  After a voice terminal user transfers a call to an on-premises voice terminal, the call, if unanswered at the expiration of the Timed Reminder Interval, redirects to an attendant console. Redirection to an attendant occurs even if the call has redirected via Call Coverage or Call Forwarding from the transferred-to voice terminal.

  An attendant-extended call redirects to coverage instead of returning to an attendant console, if the coverage criteria are met before the Timed Reminder Interval expires. However, unanswered calls return to a console at the expiration of the Timed Reminder Interval.

- Centralized Attendant Service (CAS)

  If an attendant at the main location transfers a call from a branch location to an extension at the main location, the timed reminder does not apply and the call will not return to the attendant if unanswered.

Administration

Timed Reminder is administered on a per-system basis by the System Manager. The following items require administration:

- Time a call remains on hold before the attendant is rung
- Time a call remains unanswered before the attendant is rung.

Hardware and Software Requirements

No additional hardware or software is required.

Touch-Tone Dialing

Description

Provides quick and easy pushbutton dialing. Touch-Tone Dialing is always provided with the system. In addition to the 0 through 9 buttons, the * and # buttons have special functions, such as forming a part of a feature access code. A distinctive tone is generated when each button is pressed.

If a distant switching system can accept only dial pulse signals, the system converts the touch-tone signals to the required dial pulses for transmission to the distant end.

Considerations

With Touch-Tone Dialing, users are more efficient when placing and handling calls.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

Transfer

Description

Allows voice terminal users to transfer trunk or internal calls to other voice terminals within the system without attendant assistance.

Single-line voice terminal users momentarily flash the switchhook or press the Recall button, dial the desired extension number, and hang up.

Multi-appearance voice terminal users press the Transfer button, dial the desired extension number, and press the Transfer button again.

Considerations

The Transfer feature provides a convenient way to connect a party with someone better qualified to handle the call. Attendant assistance is not required and the call does not have to be redialed.

With V2, V3, or G1, transferred trunk calls can be administered to receive either music or silence.

Multi-appearance voice terminals must have an idle call appearance in order to transfer a call.

Interactions

None.

Administration

The Transfer feature is administered on a per-system basis by the System Manager. The only administration required is whether music or silence is heard on transferred trunk calls.

Hardware and Software Requirements

No additional hardware or software is required.

Trunk Group Busy/Warning Indicators to Attendant

Description

Provides the attendant with a visual indication that the number of busy trunks in a group has reached an administered level. A visual indication is also provided when all trunks in a group are busy.

The two lamps which provide the visual indications are as follows:

- Warn Lamp

  Located on Trunk Group Select buttons that have three lamps. The Warn lamp lights when a preset number (warning threshold) of trunks are busy in the associated trunk group.

- Busy Lamp

  Located at each of the 12 Fixed Trunk Group Select buttons and, with G1, on each feature button administered as a Trunk Group Select button. The Busy lamp lights when all trunks in the associated trunk group are busy.

Considerations

The Trunk Group Busy and the Trunk Group Warning Indicators are particularly useful when the Attendant Control of Trunk Group Access feature is provided. The indicators show the attendant that control of access to trunk groups is necessary.

Each attendant console has 12 buttons designated for Trunk Group Select buttons. With G1, 12 additional Trunk Group Select buttons may be administered on feature buttons for a total of 24 Trunk Group Select buttons. All Trunk Group Select buttons have a busy indicator. Warning indicators appear on 6 of the 12 Trunk Group Select buttons on a basic console, and on all 12 Trunk Group Select buttons on an enhanced console. Feature buttons used for Trunk Group Select buttons (G1) never have Warning lamps. With V1, V2, and V3, only the first 6 Trunk Group Select buttons will activate the Warning lamps.

Interactions

If Trunk Group Select buttons are assigned for Loudspeaker Paging Acces zones, Trunk Group Busy Indicators will provide a visual indication of the busy or idle status of the zones.

Administration

This feature is administered by the System Maager. The following items require administration:

- Trunk Group Select buttons (per attendant console)
- Warning threshold (per trunk group).

Hardware and Software Requirements

No additional hardware or software is required.

Trunk Identification By Attendant (V2, V3, or G1)

Description

Allows an attendant or display-equipped voice terminal user to identify a specific trunk being used on a call. This capability is provided by assigning a Trunk ID button to the attendant console or voice terminal.

The Trunk Identification By Attendant feature can be used when a user is on an established call of one of the following types:

- An incoming trunk call
- An outgoing trunk call
- A transferred or conference call involving a trunk
- A trunk-to-trunk call.

In addition to its use during an established call, the Trunk ID button can be used while a trunk is being seized, while digits are being outpulsed on a trunk, or during intervals between digit outpulsing.

When a user is connected to a trunk, as described above, and presses the Trunk ID button, the identification of the trunk is displayed on the 40-character alphanumeric display. The trunk identification consists of the trunk access code (2-digit) for that trunk group and the trunk group member number (2-digit).

The trunk identification displayed depends on the type of call in process. If the call is incoming, the incoming trunk identification is displayed. If the call is outgoing, the outgoing trunk identification is displayed. If the call is trunk-to-trunk, the identification displayed is of the last trunk added to the call.

Considerations

Trunk Identification By Attendant is useful whenever it is necessary to identify a particular trunk being used. The feature is particularly useful for identification of a faulty trunk. That trunk can then be removed from service and the problem quickly corrected.

A maximum of one Trunk ID button is allowed per each attendant console and voice terminal with a display.

The Trunk Identification By Attendant feature is denied if there are more than two trunks on the call.

Interactions

The following features interact with the Trunk Identification By Attendant feature.

- Busy Verification

  A trunk being busy-verified can be identified.

- Attendant Display and Voice Terminal Display

  Any action by the user or the system which changes the display removes the trunk identification currently displayed. The lamp associated with the Trunk ID button remains lighted as long as the call on which the button was used remains active. While the lamp is lighted, the user can use the associated button to re-display the trunk identification.

  If the Trunk ID button is pressed during a call origination (before all digits have been dialed), the trunk identification appears. On a voice terminal display, any subsequently dialed digits are not displayed. On an attendant display, the subsequently dialed digits overwrite other digits on the dispay.

- Hold

  A trunk held by a user cannot be identified.

Administration

Trunk Identification By Attendant is assigned by the System Manager on a per-voice terminal and per-attendant console basis. The only administration required is the assignment of a Trunk ID button.

Hardware and Software Requirements

No additional hardware or software is required.

Trunk-to-Trunk Transfer

Description

Allows the attendant or voice terminal user to connect an incoming trunk call to an outgoing trunk.

Considerations

Trunk-to-Trunk Transfer is particularly useful when a caller outside the system calls a user or attendant and requests a transfer to another outside number. For example, a worker, away on business, can call in and have the call transferred elsewhere.

With V2, V3, or G1, transferred trunk calls can be administered to receive either music or silence.

An attendant-assisted call connecting an outgoing trunk to an outgoing trunk must be held on the console. The system does not allow the attendant to release such a call. The attendant can, however, use the Forced Release button and disconnect all parties associated with the call.

If a voice terminal user has connected two outgoing trunks, the user must remain on the call. Otherwise, the call will be dropped. An incoming trunk can be connected to an outgoing trunk without the user remaining on the call. An incoming trunk can also be connected to another incoming trunk without the user remaining on the call.

Trunk-to-Trunk Transfer does not apply to tie trunks.

Interactions

The Attendant Lockout feature does not function on Trunk-to-Trunk Transfer.

Administration

Trunk-to-Trunk Transfer is administered on a per-system basis by the System Manager. The only administration required is whether or not Trunk-to-Trunk Transfer is permitted and whether music or silence is heard on transferred trunk calls.

Hardware and Software Requirements

No additional hardware or software is required.

Uniform Dial Plan (V2, V3, or G1)

Description

Provides a common 4- or 5-digit dial plan that can be shared among a group of switches. Interswitch dialing and intraswitch dialing both require 4- or 5-digit dialing. The Uniform Dial Plan (UDP) is used with Electronic Tandem Network (ETN), Main/Satellite/Tributary, and Distributed Communications System (DCS) configurations. Additionally, UDP can be used alone to provide uniform 4- or 5-digit dialing between two or more private switching systems without ETN, Main/Satellite/Tributary, or DCS configurations.

In a UDP, the first 1, 2, 3, or 4 digits of the 4- or 5-digit extension number make up a PBX code which determines the switch to which a call is directed. When a UDP is administered, a list of PBX codes is assigned to each switch. A UDP can have as many as 240 PBX codes.

Each PBX code is assigned a private network office code (RNX). The RNX of a PBX in a UDP is the equivalent of an office code of a central office in a public network. It is this RNX that is actually used to determine how a UDP call is routed. Each PBX code is also administered as either local or remote to the switch

Whenever a UDP is used to route a call, the number it outputs is in the form of RNX plus XXXX. This always needs to be taken into account so that the correct digit deletion and/or insertion can be specified within the routing pattern so that the receiving switch gets digits in the format it expects.

To understand the function of a UDP, look at Figure 3-60. In this figure, a 5-digit UDP is used in an ETN. Three switches are included in the UDP. Each switch has an assigned RNX and a prefix code (described later). Each switch has also been assigned a list of PBX codes with an RNX assigned to each PBX code. Assume that the following PBX codes and associated RNXs have been assigned:

PBX CODE	RNX
41	224
51	223
52	223
60	222
61	222

If the user at extension 41000 wants to call extension 61234, he or she has two choices of how to do this. The user at extension 41000 can either dial "61234" or, if AAR is provided, the user can dial the AAR access code followed by "222-1234". If 61234 is dialed, the system recognizes 61 as a PBX code, determines the associated RNX (222), and uses AAR to route the call to extension 61234. If the AAR access code and 222-1234 are dialed, the system finds the routing pattern for RNX 222 and routes the call to the PBX associated with that RNX. The routing pattern must be administered to insert the prefix "6" at the beginning of the extension number so that the call will continue to route correctly.

If the user at extension 51234 on Switch C dials extension 61234, the call must first go through Switch A before proceeding to Switch B. When 61234 is dialed, the system recognizes 61 as a PBX code, determines the associated RNX (222), and uses AAR to route the call. The call will first be routed to Switch A, where Switch A will then recognize the RNX 222 as a remote switch and route the call to Switch B and extension 61234. This same type of call routing occurs when an extension at Switch B calls an extension at Switch C.

If extension 61234 on Switch B calls extension 61235, the system recognizes 61 as a PBX code with an RNX that is local to the switch, and the call is routed directly to extension 61235.

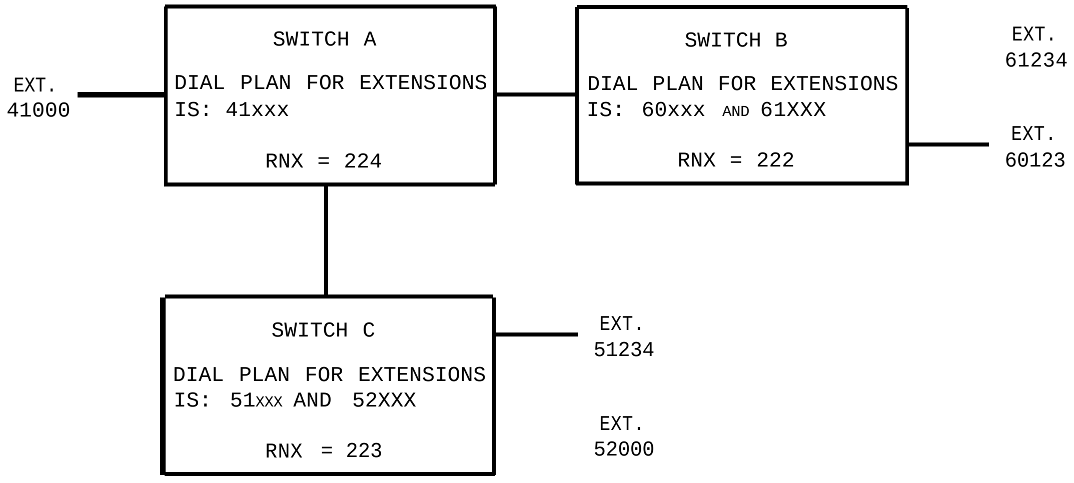

Figure 3-60. Uniform Dial Plan Example

Once a certain PBX code is assigned to a switch, no other switch within the UDP can use that same PBX code.

When a user at a switch that is included in a UDP dials an extension, the system checks to see if the first digit(s) of the extension is an assigned PBX code. If the first digit(s) is not an assigned PBX code, the call is routed via the regular, non-UDP, dial plan. If the first digit(s) is an assigned PBX code, the system translates the PBX code into the administered RNX. If the PBX code indicates that the called extension is on the same switch, the call is routed to the local extension. If the PBX code indicates that the called extension is at another switch within the UDP, Automatic Alternate Routing (AAR) uses the associated RNX to route the call to the correct switch within the private network. (The necessary subset of AAR is provided with the UDP software.) If the PBX code is not assigned a corresponding RNX, the user receives intercept treatment.

The UDP allows a user to call other extensions within a private network by dialing a 4- or 5-digit number. However, if AAR is provided, a user can also call other extensions by dialing the AAR access code, the RNX of the switch to be called, and then the desired extension number. For example, if a user on switch A wants to call extension 3797 on switch B, the user can either dial 3797 or dial the AAR access code followed by RNX 3797. When the user dials RNX 3797, AAR will route the call to the correct switch and extension.

If a 5-digit UDP is used, the routing pattern of each RNX must be administered so that it inserts a prefix digit at the beginning of the extension. For example, as shown in Figure 3-61, if a user on switch A wants to call extension 61234 on switch B, the user could dial 222-3797. Then, the routing pattern assigned to the dialed RNX would insert the prefix 6 at the beginning of the extension and route the call to the desired extension.

Considerations

The UDP feature enables a terminal user at any switch to call any other terminal on any switch in the UDP complex, using only the 4- or 5-digit extension number.

When calling an extension on another switch, there is a slight delay before call progress tones are applied. This delay is due to the trunk signaling necessary to complete the call to the remote switch.

It is possible that the first one, two, three, or four digits (PBX code) of the 4- or 5-digit extension number could be the same as a local extension number. In this case, the UDP PBX code overrides the extension number at the local switch. Problems can be avoided by assuring that the PBX code does not match an extension number.

The list of PBX codes for a switch can contain PBX codes of varying lengths. For example, a switch may be assigned both 2-digit and 3-digit PBX codes. It is also possible that one PBX code may be included in another. For example, a switch may be assigned both 61 and 612 as PBX codes. In this case, all calls beginning with 61, except those beginning with 612, are routed according to the RNX assigned to PBX code 61. Calls beginning with 612 are routed according to the RNX for PBX code 612. (The system always looks at the first four digits before routing the call.)

Interactions

The following features interact with the Uniform Dial Plan feature.

- Automatic Alternate Routing (AAR)

  After the system determines the RNX of the switch being called, AAR routes the call to the correct switch. The required subset of AAR is provided with the UDP software. If the AAR feature is provided in addition to the UDP, then the 7-digit AAR number will provide the exact same routing as the UDP.

- Direct inward Dialing (DID)

  DID calls to 5-digit UDP extension numbers require that the DID trunk group insert enough digits to make a 5-digit extension number.

- Dial Plan

  All of the extension numbers on a switch are not necessarily part of the UDP. Any that do not belong to the UDP are handled by a regular, non-UDP dial plan associated with the local switch. It is possible that the PBX code of the 4- or 5-digit extension number could be the same as a local extension number. In this case, the UDP PBX code overrides the extension number at the local switch.

- Distributed Communications System (DCS)

  UDP is required when DCS is provided. The necessary UDP software is provided with the DCS software.

Administration

The UDP is administered by the System Manager. The following items require administration:

- Whether UDP has 4- or 5-digit extension numbers
- PBX Codes (expands first 1, 2, 3, or 4 digits of dialed extension to an RNX)
- RNX Table (used by AAR to route calls to the correct switch)
- Routing Patterns.

Hardware and Software Requirements

AP/DCS interface hardware (V2, V3, or G1) or a Processor Interface circuit pack (XEV2 or XEV3) is required for DCS applications. DCS or UDP software is required.

Voice Message Retrieval (V2, V3, or G1)

Description

Allows attendants, voice terminal users, and remote access users to retrieve Leave Word Calling (LWC) and Call Coverage messages in the form of a voice output.

Voice Message Retrieval is used only for the retrieval of messages. It can be used to retrieve a user's own messages or to retrieve messages for another user. However, a different user's messages can only be retrieved by a user at a voice terminal or attendant console in the associated coverage path, by an administered systemwide message retriever, or by a Remote Access user when the extension number and associated security code are known.

Messages are protected by restricting unauthorized users from retrieving messages. A Lock function restricts a voice terminal and an Unlock function releases the restriction. The Lock function is activated by dialing a systemwide access code. The Lock function is canceled by dialing a systemwide access code and then an Unlock security code unique to the voice terminal. These functions apply only to the voice terminal where the operation is performed. The systemwide access codes and security code used for the Lock and Unlock functions are the same as those used for LWC message retrieval by display. A status lamp can be assigned to show the locked or unlocked status of a terminal.

Voice Message Retrieval is activated by dial access code. Separate access codes are used for Message Retrieval and Coverage Message Retrieval (someone else's messages). Voice Message Retrieval is activated as follows:

- To retrieve one's own messages:

  Dial the access code for Voice Message Retrieval of LWC messages. Then, dial a # sign to indicate that the extension whose messages are to be retrieved is the dialing extension, or enter a specific extension and corresponding password (same as the security code used for the Lock and Unlock functions).

- To retrieve someone else's messages:

  Dial the access code for Voice Message Retrieval of Call Coverage messages, followed by the extension of the user (within the same coverage path) whose messages are to be retrieved.

After Voice Message Retrieval has been activated, message retrieval may not be allowed for the following reasons:

- Speech Synthesizer circuit pack fails or all voice channels are busy—Reorder tone is provided.

- If the system has an Applications Processor (AP) and the AP link is down—"Messages are not available, please try later" is heard and the user is disconnected.

- An attempt is made to retrieve a message for a user not in the same coverage path or an invalid password has been entered—"Message Retrieval Denied" is heard and the user is disconnected.
- Message retrieval for the terminal whose messages are to be retrieved is locked—"Message Retrieval is locked" is heard and the user is disconnected.
- No message is heard within 10 seconds—"Press 4 for help" is heard and if no messages are heard within another 10 seconds, the user is disconnected.

When Voice Message Retrieval has successfully been activated, voice messages are heard as follows:

- If no messages are left, one of the following is heard:
  - "No messages for [abc]"—where [abc] are the initials associated with the user whose messages are to be retrieved.
  - "No messages for extension [xxxxx]"—where [xxxxx] is the extension number when no name is associated with the user whose messages are to be retrieved.
- If one message, or more, is left, one of the following is heard:
  - "[n] messages(s) for [abc]"—where [n] is the number of messages left for the user whose messages are being retrieved.
  - "[n] messages(s) for [xxxxx]"
  - "Messages for [abc]"—when an AP is provided
  - "Messages for [xxxxx]"—when an AP is provided

When a user's initials are given in a voice message, as previously described the initials are computed in the "first name(s) followed by last name" order. If a single name (Brown, for example) is administered, the entire name is spelled out.

When a user has activated Voice Message Retrieval and has heard the number of messages that has been left, one of the following functions, as described below, can be performed:

- NEXT—dial #
- REPEAT—dial 5 or *5
- DELETE—dial 3 or *3
- CALL—dial 8 or *8
- HELP—dial 4 or *4

If the NEXT function is selected, the next message, if there is one, is played. The following messages may be heard:

- "No more messages for [abc]" or "No more messages for extension [xxxxx]"—when there are no more messages.
- "No more messages, press pound to repeat messages"—when already at the last message and no AP-based messages are left.
- "Please call Message Center for more messages"—when already at the last message and there are still AP-based messages left.
- "[abc] called [n] times, last message at [time] [date] extension [xxxxx]"—when the user has received more than one message from the same caller; [abc] are the caller's initials; [n] is the number of times called; [time] is expressed an hour followed by minute (for example, "Nine Thirty-Five PM"); [date] is expressed as month followed by day (for example, "July third") or "today," if applicable; [xxxxx] is the calling party's extension number.
- "extension [xxxxx] called [n] times, last message at [time] [date]"—when the user has received more than one message from the same caller and no name is associated with the extension.
- "[abc] called at [time] [date] extension [xxxxx]" or "extension [xxxxx] called at [time] [date]"—when only one message has been left by a particular extension number.

If the REPEAT function is selected, the synthesized voice repeats the previously retrieved message with the calling party's name spelled out (instead of initials). The name is spelled out as it is administered in the system (with pauses between first and last names and also between first names if there is more than one). If no name is associated with the extension, the current message is repeated. If a message has not been retrieved, "press pound for the next message" is heard.

If the DELETE function is selected, the previously retrieved message is deleted and the user hears "Message is deleted." If no message was previously retrieved, "Press pound for next message" is heard. After a message is deleted, the user can still place a call to calling party of the deleted message, via the CALL function, as long as no other function has been entered between DELETE and CALL.

If the CALL function is selected, the extension of the calling party from the previously retrieved message is called. If no message was previously retrieved, "Press pound for next message" is heard. Otherwise, the call is initiated and the user leaves the message retrieval mode.

If the HELP function is selected, the following speech synthesized message is heard: "Press pound for the next message, press 3 to delete the message, press 4 for help, press 5 to repeat the message, press 8 to place the call."

The system expects the user to enter a function after each voice message. If a function is not entered before a specified time or if an invalid digit (digit other than #, * , 3, 4, 5, or 8) is dialed, the voice message "Press 4 for help" is heard. If no other input is entered within 10 seconds after this message, the user is automatically disconnected.

Voice Message Retrieval can be deactivated to get out of the voice message retrieval mode by doing any of the following:

- Hang up.
- Press the Drop or Disconnect button.
- Activate CALL function.

Considerations

With Voice Message Retrieval, a display-equipped voice terminal is not required to retrieve messages. Authorized users on any touch-tone terminal can retrieve messages. This results in significantly reduced traffic to the Message Centers and systemwide message retrievers.

The number of simultaneous Voice Message Retrieval users possible is dependent on the number of speech synthesizer circuit packs used in the system.

Voice Message Retrieval cannot be accessed by rotary dialing voice terminals.

Certain voice terminals and attendants can be designated for systemwide message retrieval. These systemwide retrievers are the same as those used for display message retrieval and have the same privileges.

When a terminal is in the Voice Message Retrieval mode, it cannot be used to make calls or access other features.

Interactions

The following features interact with the Voice Message Retrieval feature.

- Audio Information Exchange (AUDIX) Interface

  Retrieval of LWC messages via Voice Message Retrieval is separate and distinct from AUDIX voice message retrieval. LWC messages left for a Principal on AUDIX may not be accessed via Voice Message Retrieval; however, the invoker of Voice Message Retrieval will be told if there are any new messages for the principal on AUDIX; it will **voice** that there are message center messages (dialing 8-callout will call AUDIX), and the display retrieval will display "Message Center AUDIX Call". The LWC Messages accessible to Voice Message Retrieval are inaccessible to AUDIX; but AUDIX will inform the invoker that the messages exist.

- Bridged Call Appearance

  Activation of Voice Message Retrieval on a Bridged Call Appearance functions the same as if it was activated by the primary extension associated with the Bridged Call Appearance.

- Call Forwarding

  A forwarded-to user cannot retrieve messages for a forwarding user unless both users are in the same coverage path.

- Call Pickup

  A user cannot retrieve messages for a member of his or her Call Pickup group, unless both users are in the same coverage path.

- Leave Word Calling

  Voice Message Retrieval enhances Leave Word Calling (LWC) by allowing any authorized touch-tone terminal user to retrieve messages.

Administration

Voice Message Retrieval is administered by the System Manager. The following items require administration:

- Voice Message Retrieval Access Code for LWC message retrieval (per system)
- Voice Message Retrieval Access Code for Call Coverage message retrieval (per system)
- Lock and Unlock Access Codes (per system)
- Unlock Security Code (per voice terminal)
- Identities of authorized systemwide message retrievers.

Hardware and Software Requirements

Requires a TN725 Speech Synthesizer circuit pack. Each circuit pack has four ports to provide Voice Message Retrieval. No additional software is required.

Voice Terminal Display

Description

Provides multi-appearance voice terminal users with updated call and message information. This information is displayed on a display-equipped terminal. The information displayed depends upon the display mode selected by the user.

Several modes can be assigned to buttons and then selected by pressing the assigned button. All buttons are located on the display module or voice terminal. All buttons are administrable.

- Normal Mode

  Displays call-related information for the active call appearance. This display includes information identifying the call appearance, calling or called party, and calling or called number. The display must be in the Normal mode to answer incoming calls and to display information associated with the V2, V3, or G1 Automatic Incoming Call Display feature.

- Inspect Mode

  Displays call-related information for an incoming call when the user is active on a different call appearance. This button is pressed when the user is active on one call appearance and receives a call on another appearance.

- Stored Number Mode

  Displays the last number the user dialed (Last Number Dialed feature), the number stored in an Abbreviated Dialing button administered to the voice terminal, a number stored in an Abbreviated Dialing list, or a number assigned to a button administered through the Facility Busy Indication feature.

- Date and Time Mode

  Displays the current date and time of day.

- Elapsed Time Mode

  Displays elapsed time in hours, minutes, and seconds. The timing starts or stops when the button is pressed. This button can be pressed at any time.

- Integrated Directory Mode

  Turns off the touch-tone signals and allows the touch-tone buttons to be used to key in the name of a system user. After a name is keyed in, the display shows that name and associated extension number. (Refer to the Integrated Directory feature for complete details.)

- Message Retrieval Mode

  Retrieves messages for voice terminal users. If no messages are stored, display shows NO MESSAGES.

- Coverage Message Retrieval Mode

  Retrieves messages for voice terminal users who do not have a display module assigned to their voice terminal. Retrieval permission must be administered for a user to retrieve another user's messages. Messages can be retrieved at any time. The retriever does not need to lift the handset to retrieve messages. Also, messages can be retrieved even if the retriever is active on a call.

The Message Retrieval, Coverage Message Retrieval, or Integrated Directory buttons have three other associated buttons:

- Next Message

  Retrieves the next message or displays END OF FILE, PUSH Next TO REPEAT when in the retrieval mode. Displays the next name in the alphabetical listing when in the Integrated Directory mode. This button must be assigned when a Retrieval button is assigned.

- Delete

  Deletes the currently displayed message. This button must be assigned when a Retrieval button is assigned.

- Return Call

  Automatically returns the call requested by the currently displayed message or the currently displayed name and extension number.

The system provides the following call-related information:

- Call Appearance Identification

  The call appearance buttons are designated on the display by a lower-case letter; for example, a, b, and c. The display shows a= for a call incoming on the first call appearance button, b= for a call incoming on the second call appearance button, and so on.

- Calling Party Identification

  — Version 1

    When the call is from a system user, the display shows the caller's extension number, the caller's name, or a unique identification administered for the voice terminal being used. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, assigned to the trunk group used for the call.

— V2, V3, or G1

When the call is from a system user, the display shows the caller's name or a unique identification administered for the voice terminal being used, along with the calling party's extension number. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, and the trunk access code, assigned to the trunk group used for the call. If a user is active on a call, and receives a subsequent call, the display automatically shows the identification of the subsequent caller.

- Called Party Identification

— Version 1

On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name. If no name is assigned, the called party's extension number is displayed.

On outgoing calls, the display shows the digits as they are dialed, followed by the name assigned to the trunk group being used. The System Manager can suppress the name of any trunk group. If such a trunk group is accessed, the called party portion of the display is blank.

— V2, V3, or G1

On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name and extension number. If no name is assigned, only the called party's extension number is displayed.

On outgoing calls, the display shows the digits as they are dialed, followed by the name and trunk access code assigned to the trunk group being used. The System Manager can suppress the name of any trunk group. If such a trunk group is accessed, the name portion of the display is blank.

- Call Purpose

This identifies the reason for an incoming call or a redirected call. (A normal incoming call is not identified by a call purpose.) The following call purpose identifiers can be displayed:

f—Call Forwarding—Indicates that another user has forwarded calls to this voice terminal.

s—Send All Calls—Indicates that the called user is temporarily sending all calls to coverage, and that the call has been redirected to this voice terminal.

d—Don't Answer or Cover—Indicates that the called voice terminal was not answered or that the calling system user has sent the call to coverage, or the called voice terminal user is not available. This identifier also indicates that the called voice terminal user has a temporary bridged appearance of the call.

b—Busy—Indicates that the called voice terminal user is active on a call, and the called voice terminal user has a temporary bridged appearance of the call.

B—Busy—Indicates that the called voice terminal user is active on a call, and the called voice terminal user does not have a temporary bridged appearance of the call.

callback—Indicates that the call is an Automatic Callback call from the system.

icom—Indicates that the incoming call is an Intercom call.

park—indicates that the user parked a call.

pickup—Indicates that the user answered a Call Pickup group member’s call.

priority—Indicates that the incoming call has priority status.

Some typical displays are as follows:

- Internal call (V1):

a=3602

then

a =	TOM BROWN

a=	EXT 3602

- Internal call (V2, V3, or G1):

a=3602

then

a=	TOM BROWN	3062

or

a=	EXT 3602	3062

- Outgoing trunk call (V1):

b=87843541

Where 8 is the trunk access code and 784-3541 is the number dialed.

then

b= OUTSIDE CALL

or

b= WATS

- Outgoing trunk call (V2, V3, or G1):

b=87843541

Where 8 is the trunk access code and 784-3541 is the number dialed.

then

b=	OUTSIDE CALL	8

or

b=	WATS	101

- Incoming trunk call (V1):

a= OUTSIDE CALL

- Incoming trunk call (V2, V3, or G1):

```
a =   OUTSIDE CALL            102
```

Where 102 is the trunk access code of the incoming trunk group.

- Conference call originated by the attendant:

```
b=  CONFERENCE  4
```

Where 4 is the number of conferees. The number does not include the conference call originator.

- Internal call redirected to coverage:

```
b=     EXT  3174  to  EXT  3077    d
```

or

```
b=     BOB  SMITH  to  JOYCE  THOMAS  d
```

Where d indicates that Go To Cover was activated by the calling voice terminal user.

- Incoming trunk call redirected to coverage:

```
b =   OUTSIDE  CALL  to  DON  SMITH   s
```

Where s indicates that Send All Calls was activated by the called voice terminal user.

- Message Retrieval

```
IN PROGRESS
```

then

```
MESSAGES FOR BETTY R. SIMS
```

then

```
JOE JONES 10/16 11:40a 2 CALL 3124
```

This message means that Joe Jones called Betty Sims the morning of October 16. The second message was stored at 11:40 a.m. Joe wants Betty to call his extension number, 3124.

- Integrated Directory mode:

```
CARTER, ANN        3408           3
```

This display shows the name and extension number as administered in the system. The 3 indicates that three buttons were pressed to reach this particular display.

Considerations

The Voice Terminal Display feature provides an instant display of information associated with certain system features, functions, and services. Information that allows personalized call answering is available on many calls. Retrieval of stored information, such as messages received and directory information, is easy as well as convenient.

Up to 62 (V1), 225 (V2 or V3), or 500 (G1) display modules can be provided per system.

Certain voice terminals and the attendant group can be designated for systemwide message retrieval. Users of these voice terminals or consoles can retrieve Leave Word Calling and Call Coverage messages for other voice terminal users including Direct Department Calling groups, Uniform Call Distribution groups, Personal Central Office Line groups, and Terminating Extension Groups. Selected users cannot retrieve messages for other selected users. Up to ten voice terminals, or up to nine voice terminals and the attendant group, can be designated for systemwide message retrieval. Systemwide retrieving voice terminals or consoles are assigned when the system is implemented.

If the following conditions are met, messages for a voice terminal user can be retrieved at selected terminals or any attendant console:

- The retriever must be in the user's Call Coverage path.
- Permission to retrieve messages must be assigned for the user's voice terminal.

If permission is granted, any voice terminal with a display module or the attendant group in the user's Call Coverage path can retrieve messages for that user.

When all messages have been displayed and deleted for an extension number, the Message lamp on the voice terminal and any associated Remote Message Waiting Indicator, if assigned, go dark.

The display module used with voice terminals is similar to the attendant console display. However, the display module has an On-Off button, and can be turned off when not in use. The display module can be used only with various voice terminals.

Interactions

The following features interact with the Voice Terminal Display feature.

- Bridged Call Appearance

  A call from the primary extension number or a bridged call appearance of the primary extension number is displayed as a call from the primary extension number.

- Last Number Dialed

  If the Last Number Dialed feature access code is dialed after the stored number button has been pressed, the last number dialed is no longer displayed. However, if the Last Number Dialed button is pressed after the stored number button has been pressed, the last number dialed is displayed.

- Single-Digit Dialing and Mixed Station Numbering (V3 or G1)

  If prefixed extensions are used in the system's dial plan, the prefix is not displayed when the extension is displayed. With V3, the Return Call button cannot be used to dial prefixed extensions, because this button causes the system to dial the displayed number, which does not contain the entire extension. With G1, the Return Call button can be used to dial prefixed extensions, because the G1 system will dial the prefix, even though it is not displayed.

Administration

Voice Terminal Display is administered on a per-voice terminal basis by the System Manager. The following items require administration.

- Whether or not a display module is provided (per display capable voice terminal)
- Whether or not to restrict other users from reading or canceling the voice terminal's message (per display module)
- The following buttons (per display module):
  - Normal
  - Inspect
  - Stored Number
  - Date and Time
  - Elapsed Time
  - Integrated Directory
  - Message Retrieval
  - Coverage Message Retrieval
  - Next Message (must be assigned with either Retrieval button)
  - Delete (must be assigned with either Retrieval button)
  - Return Call (optional with either Retrieval button or the Integrated Directory button).

Hardware and Software Requirements

Requires a display-equipped voice terminal and one port on a TN754 or TN784 (G1) Digital Line circuit pack. No additional software is required.

CHAPTER 4. SYSTEM PARAMETERS

CHAPTER 4. SYSTEM PARAMETERS

Overview

This chapter provides information on the overall characteristics and capacities of the system.

The items presented in this chapter are grouped here for easy reference. However, most items are described under each applicable feature.

Feature Administration

Administration Not Required

Administration is not required to activate the following features.

Attendant Auto-Manual Splitting
Attendant Call Waiting
Attendant Recall
Attendant Release Loop Operation
Automatic Incoming Call Display (V2, V3, or G1)
Conference—Attendant
Conference—Terminal
Hold
Line Lockout
Move Agents from CMS (V3 or G1)
Recall Signaling
Recent Change History (G1)
Senderized Operation
Straightforward Outward Completion
Temporary Bridged Appearance
Through Dialing
Touch-Tone Dialing (for Terminals)
Transfer

Administration Required

Administration is required to activate the following features.

AAR/ARS Partitioning (V3 or G1)
Abandoned Call Search (V3 or G1)
Abbreviated Dialing
Agent Call Handling (V3 or G1)
Attendant Control of Trunk Group Access
Attendant Direct Extension Selection With Busy Lamp Field
Attendant Direct Trunk Group Selection

Attendant Display (Buttons only)
Audio Information Exchange (AUDIX) Interface (V3 or G1)
Authorization Codes (V3 or G1)
Automatic Alternate Routing (V2, V3, or G1)
Automatic Callback
Automatic Call Distribution (V3 or G1)
Automatic Circuit Assurance (V2, V3, or G1)
Automatic Route Selection
Automatic Wakeup (V3 or G1)
Basic Call Management System (G1)
Bridged Call Appearance—Multi-Appearance Voice Terminal
Bridged Call Appearance—Single-Line Voice Terminal (G1)
Busy Verification of Terminals and Trunks (V2, V3, or G1)
Call by Call Service Selection (G1)
Call Coverage
Call Forwarding All Calls
Call Park
Call Pickup
Call Waiting Termination
Centralized Attendant Service (V2, V3, or G1)
Code Calling Access
Customer-Provided Equipment (CPE) Alarm (XEV2, XEV3, or G1)
Data Call Setup
Data-Only Off-Premises Extensions
Data Privacy
Data Restriction
Dial Access to Attendant
Digital Multiplexed Interface (V2, V3, or G1)
Direct Department Calling
Direct Inward Dialing
Direct Outward Dialing
Distinctive Ringing
Do Not Disturb (V3 or G1)
DS1 Tie Trunk Service (V2, V3, or G1)
EIA Interface (V2, V3, or G1)
Emergency Access to the Attendant (V3 or G1)
Facility Busy Indication
Facility Test Calls
Forced Entry of Account Codes (V3 or G1)
Generalized Route Selection (G1)
Hot Line Service
Hunting
Individual Attendant Access (V2, V3, or G1)
Information Systems Network (ISN) Interface
Integrated Directory
Integrated Services Digital Network—Primary Rate Interface (G1)
Intercept Treatment
Intercom—Dial
Inter-PBX Attendant Calls (V2, V3, or G1)
Intraflow and Interflow (V3 or G1)

Last Number Dialed
Leave Word Calling
Loudspeaker Paging Access
Loudspeaker Paging Access—Deluxe
Manual Message Waiting
Manual Originating Line Service
Manual Signaling
Modem Pooling
Multi-Appearance Preselection and Preference
Multiple Listed Directory Numbers
Music-on-Hold Access
Names Registration (G1)
Night Service
Off-Premises Station
Outbound Call Management (G1)
PC/PBX Connection
Permanent Switched Calls (V2, V3, or G1)
Personal Central Office Line
Personalized Ringing (V2, V3, or G1)
Power Failure Transfer
Priority Calling
Privacy—Attendant Lockout
Privacy—Manual Exclusion
Property Management System Interface (V3 or G1)
Queue Status Indications (V3 or G1)
Recorded Announcement
Recorded Telephone Dictation Access
Remote Access
Report Scheduler and System Printer (G1)
Restrictions
Ringback Queuing
Ringer Cutoff
Rotary Dialing (V2, V3, or G1)
Service Observing (V3 or G1)
Single-Digit Dialing and Mixed Station Numbering (V3 or G1)
SMDR Account Code Dialing
Station Message Detail Recording
Ten-Digit to Seven-Digit Conversion (G1)
Terminating Extension Group
Timed Reminder
Time of Day Routing (G1)
Touch-Tone Dialing (for Trunks)
Trunk Group Busy/Warning Indicators to Attendant
Trunk-to-Trunk Transfer
Uniform Call Distribution (see Direct Department Calling)
Uniform Dial Plan (V2, V3, or G1)
Voice Message Retrieval (V2, V3, or G1)
Voice Terminal Display

Feature Access

Dial Access

The following features or feature options can be activated and/or deactivated by dialing the assigned Feature Access Code or Trunk Access Code.

- Abbreviated Dialing:

  List 1
  List 2
  List 3
  Program

- Agent Call Handling (V3 or G1)

  Agent Log-In
  Agent Log-Out
  Manual-In
  Auto-In
  After Call Work
  Auxiliary Work
  Assist

- AP Demand Print
- Automatic Route Selection (V2, V3, or G1)
- Automatic Callback (activate and deactivate) (applies to single-line voice terminals only)
- Automatic Wakeup (V3 or G1)

  Wakeup Call
  Verify Wakeup Announcement

- Call By Call Service Selection (G1)
- Call Forwarding All Calls (activate and deactivate)
- Call Park and Call Park Answer Back
- Call Pickup
- Code Calling Access
- Controlled Restriction:

  Single Voice Terminal (activate and deactivate)
  Group of Voice Terminals (activate and deactivate)

- Data Origination (associated with Data Call Setup and Pooled Modem)
- Data Privacy (associated with Data Call Setup and Pooled Modem)
- Do Not Disturb (V3 or G1)
- Emergency Access to the Attendant (V3 or G1)
- Facility Test Calls (V2, V3, or G1)
- Generalized Route Selection (G1)
- Hunt Group Make Busy (activate and deactivate) (associated with Direct Department Calling and Uniform Call Distribution)
- Integrated Services Digital Network—Primary Rate Interface
- Last Number Dialed
- Leave Word Calling:

  Cancel a Message
  Display Module Lock
  Display Module Unlock
  Store a Message
- Loudspeaker Paging Access
- Loudspeaker Paging Access—Deluxe (G1)
- Private Network Access
- Priority Calling
- Property Management System Interface (V3 or G1)
- Public Network Access
- Recorded Telephone Dictation Access
- Send All Calls (associated with Call Coverage)
- SMDR Account Code Dialing
- Trunk Answer From Any Station (associated with Night Service)
- Voice Message Retrieval (V2, V3, or G1)

  Message Retrieval Mode
  Coverage Message Retrieval Mode
  Delete Message
  Repeat Message
  Next Message

Help
Call

Button Access

The following features or feature options must be assigned to a button. Feature Access Codes cannot be provided.

- Automatic Callback (applies to multi-appearance voice terminals only)
- Automatic Circuit Assurance
- Bridged Call Appearance—Multi-Appearance Voice Terminal
- Bridged Call Appearance—Single-Line Voice Terminal (G1)
- Busy Verification of Terminals and Trunks (V2, V3, or G1)
- Call Coverage:

  Consult
  Coverage Callback
  Coverage Message Retrieval
  Go To Cover

- Data Extension (associated with Data Call Setup)
- Display—Attendant or Voice Terminal:

  Date and Time
  Timer (Elapsed Time)
  Inspect
  Integrated Directory
  Normal
  Stored Number (associated with Abbreviated Dialing)

- Facility Busy Indication
- Intercom:

  Automatic
  Dial

- Leave Word Calling:

  Delete Message
  Message Retrieval
  Next Message (also used with Integrated Directory)
  Return Call (also used with Integrated Directory)

- Manual Message Waiting
- Manual Signaling
- Personal Central Office Line
- Privacy—Manual Exclusion
- Property Management System Interface (V3 or G1)

  Message Waiting Notification (Activate)
  Message Waiting Notification (Deactivate)
  Checkout

- Queue Status Indications (V3 or G1)

  NQC (number of queued calls)
  OQT (oldest queued time)
  AQC (attendant queued calls)
  AQT (attendant queued time)

- Ringer Cutoff
- Service Observing (V3 or G1)
- Special Characters (associated with Abbreviated Dialing)

  Pause, Wait, Mark, and Suppress can each be assigned to a button or a Function Entry button can be assigned. Pressing Function Entry and then dialing 1, 2, 3, or 4 depicts Pause, Wait, Mark, or Suppress, respectively.

- Terminating Extension Group
- Time of Day Routing (G1)

  Immediate Manual Override
  Clocked Manual Override

- Trunk Identification by Attendant.

Dial and Button Access

The following features or feature options can be activated and/or deactivated by dialing the assigned Feature Access Code; they can also be assigned to a button for button access.

- Abbreviated Dialing:

  List 1
  List 2
  List 3
  Program

- Agent Call Handling (V3 or G1)

  Manual-In
  Auto-In
  Auxiliary Work
  After Call Work
  Assist
  Release

- Automatic Wakeup (V3 or G1):

  Auto Wakeup Entry
  Failed Messages Wakeup

- Call Park and Call Park Answer Back

- Call Pickup

- Do Not Disturb (V3 or G1)

- Emergency Access to the Attendant (V3 or G1)

- Hunt Group Make Busy (activate and deactivate) (associated with Direct Department Calling and Uniform Call Distribution)

- Last Number Dialed

- Leave Word Calling:

  Cancel a Message
  Display Module Lock
  Store a Message

- Priority Calling

  The Priority Calling access code and extension number to be called, or the Priority Calling access code only, can be assigned to an Abbreviated Dialing (AD) button.

- Send All Calls (associated with Call Coverage).

Feature Status Button Indicators

The following buttons are not operational, but can be assigned to indicate the status of a feature or feature option. The lamp associated with the button lights when the assigned feature or option is active or is in use.

- Group Call (Lights to indicate that an incoming call is associated with a Call Coverage Answer group, a Direct Department Calling group, or a Uniform Call Distribution group.)

- Lock (Associated with the Voice Terminal Display; lights when activated and means that Leave Word Calling message retrieval will be denied from that terminal. Other display modes still work, including Coverage Message Retrieval.)

System Capacities

A synopsis of significant hardware, feature, and function capacities for System 75, System 75 XE, and DEFINITY Generic 1 is shown on the next few pages. In the following tables, these items should be considered:

- The headings V1, V2, V3 refer to System 75 Release 1 Version 1, Release 1 Version 2, and Release 1 Version 3, respectively.
- The heading G1 refers to the G1 multi-carrier cabinet.
- The quantities shown under the V2 and V3 headings are the same for XEV2 and XEV3 **unless otherwise noted.**
- The quantities shown under the G1 heading are the same for the single-carrier cabinet **unless otherwise noted.**

ITEM	V1	V2	V3	G1
Abbreviated Dialing				
Lists Per System	502	802	802	1600
List Entry Size	16	24	24	24
Entries Per System	2500	4010	4010	8000
Personal Lists	400	800	800	1600
Max. Entries	10	10	10	10
Per Extension	1	3	3	3
Group Lists	100	100	100	100
Max. Entries	15	90	90	90
Per Extension	3	3	3	3
System Lists	1	1	1	1
Max. Entries	50	90	90	90
Enhanced Lists	–	–	1	1
Max. Entries	–	–	1000	1000
Adjuncts/Applications				
Applications Processors (AP16)	1	1*	–	–
BX.25 Links (SCC/MCC)	1	4	4/4	4/8
Message Server Adjuncts	–	–	1	1
AUDIX Adjuncts	–	–	1	1
CMS Adjuncts	–	–	1	1
OCM Adjuncts†	–	–	–	1
ISDN Gateway†	–	–	–	8
Asynchronous Links (RS-232)	–	1	4	5
SMDR Output Devices	–	1	1	2
Journal Printer	–	–	2	2
System Printer	–	–	–	1
PMS	–	–	1	1
BX.25 Processor Channels	–	64	64	64
Hop Channels	–	64	64	64

* Quantity shown is for V2 and V3. XEV2 and XEV3 do not support the AP.

† The system can have either an OCM adjunct or an ISDN Gateway. Both cannot co-reside on the same switch.

ITEM	V1	V2	V3	G1
AAR/ARS				
Patterns	16	254	254	254
ARS Patterns for Measurement	–	20	20	20
Trunks in an ARS Pattern	6	6	6	6
Entries in RNX Table	–	640	640	640
Entries in FNPA Table	200	200	200	200
RHNPA Tables	4	32	32	32
Toll Tables	4	32	32	32
HNPA Tables	1	1	4	8
FNPA Tables	1	1	4	8
RNX Tables	–	1	4	8
UDP (Entries)	–	240	240	240
Choices per RHNPA Table	–	12	12	12
Entries in Toll Table	800	800	800	800
Entries in HNPA/RHNPA Tables	800	800	800	800
FRLs	8	8	8	8
Inserted Digit Strings*	–	1200	1200	1200
Digits Inserted for ARS/AAR	–	36	36	36
Digits Deleted for ARS/AAR	–	11	11	11
Routing Plans (PGNs)	–	–	4	8
TOD Charts	–	–	–	8
Digit Conversion Entries (ARS/AAR)	–	–	–	180
Attendant Service				
Attendant Positions (day/night)	6/1	6/1	6/1	6/1
100's Groups/Attendant Console	8	8	8	20
Queue Length	30	30	30	30
Emergency Access Queue Length	–	–	50	50
Centralized Attendant Service				
Release Link Trunks at Branch	–	16	16	16
Release Link Trunks at Main	–	–	200	400
Release Link Trunk Groups	–	–	99	99
Authorization				
Classes of Restriction	64	64	64	64
Classes of Service	16	16	16	16
Authorization Codes	–	–	5000	5000
Length of Authorization Code	–	–	4-7	4-7
Remote Access Barrier Codes	10	10	10	10
Length of Barrier Code	4-7	4-7	4-7	4-7
Unrestricted/Allowed Call Lists	1	1	1	1
Total Call List Entries	10	10	10	10

* The number of 12-character inserted digit strings available for AAR/ARS preferences.

ITEM	V1	V2	V3	G1
Automatic Callback Calls	40	80	80	160
Automatic Wakeup				
Wakeup Requests per System (Shared With Do Not Disturb)	–	–	800	1600
Wakeup Request per Extension	–	–	1	1
Wakeup Requests per 15 min. Time Interval	–	–	200	300
Advance Wakeup Request Time				
(Hours)	–	–	23	23
(Minutes)	–	–	55	55
Simultaneous Display Requests	–	–	10	10
Bridged Call Appearances	400	500	800	1600
Cabinets				
PPN (Multi-Carrier Cabinet)	1	1	1	1
PPN (Single-Carrier Cabinet)	–	4	4	4
EPN* (Multi-Carrier Cabinet)	–	–	–	1
EPN* (Single-Carrier Cabinet)	–	–	–	4
EPN* (Small)	–	–	–	1
Call Coverage				
Coverage Paths	200	400	400	600
With Hospitality Parameter Reduction	–	–	5	5
Coverage Points in a Path	3	3	3	3
Coverage Paths Linked Together	–	–	4	4
Coverage Answer Group	100	200	200	200
Members per Coverage Answer Group	8	8	8	8
CMS Reports				
BCMS				
Measured Agents	–	–	–	30
Measured Splits	–	–	–	30
Measured Trunk Groups	–	–	–	30
Measured Trunk Group Members	–	–	–	400
Reporting Periods (30 or 60 Minutes)	–	–	–	25
Daily Summary Reports	–	–	–	7
External CMS†				
Measured Agents	–	–	200	400
Measured Splits	–	–	32	32
Measured Trunk Groups	–	–	99	99

* The numbers given are total cabinets.

† Limited by the switch's capacity. The external CMS adjunct limits may differ.

ITEM	V1	V2	V3	G1
Call Park				
Attendant Group Common Shared Extension Numbers*	10	10	10	10
Number of Parked Calls	236	241	241	482
Call Pickup Groups	200	400	400	800
With Hospitality Parameter Reduction	–	–	5	5
Call Pickup Members per System	400	800	800	1600
Call Pickup Members per Group	25	50	50	50
Conference Parties	6	6	6	6
Data Parameters				
Administered Connections*	–	18	18	18
Digital Data Endpoints	200	400	400	800
DS1 Circuit Packs	–	20	20	30
Dial Plan (Name/Number Database)†				
Extensions	600	1200	1200	2500
Dial Access Codes	50	50	70	70
Trunk Access Codes	100	118	157	197
Name Size in Characters	15	15	15	15
Integrated Directory Entries	400	800	800	1600
Maximum Extension Size	4	5	5‡	5‡
Multiple Listed Directory Numbers	50	50	50	50
DID numbers	8	8	8	8
Do Not Disturb (DND)				
DND Requests per System (Shared with Automatic Wakeup)	–	–	800	1600
Simultaneous Display Requests	–	–	10	10
Facility Busy Indicators	1000	1600	1600	2400
Buttons per Tracked Resource	100	100	100	100

* Subject to Dial Plan limitation.

† There is an internal limitation of 860 total PCOL groups, common shared extensions, access endpoints, code calling ids, LDNs, hunt groups, announcements, and TEGs.

‡ A prefixed extension can be 6 digits.

ITEM	V1	V2	V3	G1
Hunt Groups or Splits				
Groups/Splits*	32	32	32	99
With Hospitality Parameter Reduction	–	–	5	5
Group Members per System	448	448	448	500
Group Members per Group/Split	32	32	100	200
Measured Groups/Splits	32	32	32	99
Queue Slots per Group/Split	35	35	100	200
Queue Slots per System	1120	1120	1000	1000
Announcements per Group/Split	1	1	2	2
ACD Supervisors per System	–	–	32	99
Intercom Groups (Automatic and Dial Combined)	32	32	32	32
Members per Group	32	32	32	32
Members per System	128	128	1024	1024
Leave Word Calling (Switch-Based, No AP)				
Messages Stored	1000	2000	2000	2000
Messages per User	10	10	10	10
Individual Message Retrievers	60	60	60	60
System-Wide Message Retrievers	10	10	10	10
Remote Message Waiting Indicator				
Per Extension	1	1	80	80
Per System	50	80	80	80
Modem Pool Groups				
Mode 2/Analog	1	5	5	5
Members per Group	32	32	32	32
Group Member per System	32	160	160	160
Personal CO Line Group (PCOL)				
PCOL Groups*	25	40	40	40
PCOL Members in a Group	4	4	4	4
Paging				
Loudspeaker Zones	9	9	9	9
Code Calling Ids*	125	125	125	125

* Subject to Dial Plan limitation.

ITEM	V1	V2	V3	G1
Port Circuit Pack Slots*				
PPN				
Multi-Carrier Cabinet Without Duplication	85	85	85	89
Multi-Carrier Cabinet With Duplication	–	–	–	78
Single-Carrier Cabinet Without Duplication	–	–	–	64
Single-Carrier Cabinet With Duplication	–	–	–	56
EPN†				
Multi-Carrier Cabinet	–	–	–	99
Single-Carrier Cabinet	–	–	–	71
Small (Upgrades Only for Generic 1)	–	–	–	39
Power Failure Transfer Extensions				
Model 574-5 Panel	35	35	35	35
Z1A Panel	42	42	42	42
Recorded Announcements				
Recorded Announcements‡	10	10	64	64
Analog Queue Slots per System	50	50	150	150
Analog Queue Slots per Announcement	5	5	150	150
Integrated Queue Slots per System	–	–	50	50
Calls Connected per Announcement	–	–	5	5
Integrated Circuit Packs	–	–	1	1
Channels per Integrated Announcement Pack	–	–	16	16
Integrated Announcement Recording Time				
32 Kb Recording (min:sec)	–	–	4:16	4:16
Station Message Detail Recording (SMDR)				
SMDR Output Devices	1	1	1	2§
Tracked Trunks	200	200	400	400
Buffered Records	–	–	–	200
Speech Synthesizer Circuit Packs	–	6	6	6
Channels per Speech Circuit Pack	–	4	4	4
System Administration				
Async Links (RS-232)	6	6	6	7
Simultaneous SAT/Manager Sessions	3	3	3	3
Simultaneous Administration Commands	1	1	1	1
Simultaneous Maintenance Commands	1	1	1	1
Scheduled Report Entries	–	–	–	50
History File Entries	–	–	–	250

* Numbers indicate total port slots available. Maintenance circuit packs, Expansion Interfaces, Tone/Clock circuit packs, TTRs, etc., all use these port slots.

† Numbers are per EPN. The Single-Carrier Cabinet number assumes 4 carriers.

‡ Subject to Dial Plan limitation.

§ All SMDR records are sent to both ports.

ITEM	V1	V2	V3	G1
Terminating Extension Groups (TEGs)*	32	32	32	32
Users that may share a TEG	4	4	4	4
Time Slots	512	512	512	1024
Slots Available for Call Switching	473	483	483	966
Simultaneous Conversations	236	241	241	482
Tone Classifiers				
Tone Detector Circuit Packs	5	5	5	20
General Purpose Tone Detectors	10	10	10	40
Touch-Tone Receivers	20	20	20	80
TTR Queue Size	4	4	4	4
Traffic Handling Capability (CCS)	8500	8676	8676	17352
Busy Hour Call Completions†				
Basic System	3600	3600	3600	7200
ISDN System	–	–	–	5000
ACD System	–	–	–	5500
OCM	–	–	–	4300
Trunks	200	200	200	400
With Hospitality Parameter Reduction	-	-	5	5
Trunk Members in a Trunk Group	60	60	60	99
Trunk Groups in the System	50	60	99	99
Queue Slots for Trunks	100	120	198	198
Ringback Queue Slots	100	120	120	120
DS1 Circuit Packs	–	20	20	30
PRI Interfaces (D-channels)				
Multi-Carrier Cabinet	–	–	–	8
Single-Carrier Cabinet	–	–	–	4
PRI Interfaces (B-channels)				
Multi-Carrier Cabinet	–	–	–	184
Single-Carrier Cabinet	–	–	–	92
Voice Terminals	400	800	800	1600
Digital Terminals	400	680	680	1472
Digital Display Modules‡	62	225	225	500
Button Modules (Terminal Modules [adjuncts], Terminals with more than 10 assignable buttons)	125	450	450	1000
Phantom Users	125	125	125	150

* Subject to Dial Plan limitation.

† Based on a Distribution of 36% incoming, 36% outgoing, and 28% intercom calls Traffic capabilities for individual configurations will vary.

‡ Subject to button module limitations.

CHAPTER 5. REFERENCES

The following is an abbreviated listing of Generic 1 and System 75 documents. Included is a brief description of each document in the list. User instructions are also available for all terminals used with the systems. For a complete listing of documents, refer to the *DEFINITY Communications System Generic 1 and System 75—Documentation Guide,* 555-200-010.

To order copies of any of these documents, refer to the address on the back of the title page.

Business Communications Systems Publications Catalog **555-000-010**

Provides a list of publications that support AT&T business communications systems. Also provides a brief description of each publication listed.

DEFINITY® Communications System and System 75 and System 85—Terminals and Adjuncts—Installation and Test **555-015-104**

Provides procedures for installing and removing voice terminals (including Business Communications Terminal built-in voice terminals) and adding modules and adjuncts to voice terminals. Also shows how to provide auxiliary power for voice terminals and associated modules and adjuncts. Provides references to other documents that contain step-by-step instructions for making cross-connections.

DEFINITY® Communications System and System 75 and System 85—Terminals and Adjuncts—Reference **555-015-201**

Provides concise physical and functional descriptions of the peripheral equipment that can be used with DEFINITY Generic 1, DEFINITY Generic 2, System 75, and System 85. It is intended as an aid for both AT&T and customer personnel in selecting appropriate components for these systems and in training and management.

DEFINITY® Communications System and System 75 and System 85—DS1/DMI/ISDN-PRI—Reference **555-025-101**

Provides a broad but detailed description of the DS1 Tie Trunk Service, Digital Multiplexed Interface (DMI), and Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI) features. Introduces and defines concepts and terminology unique to DS1, DMI, and ISDN-PRI. Also includes applications, engineering procedures and considerations, cabling and connection arrangements, administration requirements, restrictions and limitations, etc.

An Introduction to DEFINITY® Communications System Generic 1 **555-200-024**

Provides an overview of DEFINITY Generic 1. Major hardware components, such as the switch, terminals, and software applications, are described to provide an understanding of the system's functional areas. Also provides an overview of System Management and data features and functions available with DEFINITY Generic 1.

DEFINITY® Communications System Generic 1 Hospitality Services **555-200-026**

Provides an overview of DEFINITY Generic 1 Hospitality Services. Major hardware components, such as the system cabinet, and terminals are described to provide an understanding of the system’s functional areas. Also provides an overview of the hospitality features available with the system, as well as other voice, data, and System Management features.

AT&T System 75—Installation and Test **555-200-104**

Provides the information necessary to perform the tasks of installing and testing the system’s common equipment. Includes a description of the necessary tools and equipment. Information in this document applies to both System 75 and System 75 XE.

AT&T System 75 and System 75 XE—Maintenance **555-200-105**

Provides the information necessary for monitoring, testing, and maintaining the System 75 and System 75 XE. It is intended to cover many of the faults and troubles that can occur in the system.

AT&T System 75—Upgrades and Additions **555-200-106**

Provides procedures and information required to upgrade an R1V1 system to an R1V2 system, upgrade an R1V1 or R1V2 system to an R1V3 system, and make additions to an operational System 75 or System 75 XE, after the initial switch installation.

AT&T System 75—Wiring **555-200-111**

Provides an overview of the System 75 wiring plan. Included are:

- General guidelines on hardware selection
- Descriptions of cross-connect hardware and installation procedures.
- Equipment room hardware and cabling instructions including the cross-connect field
- Instructions for station wiring and associated hardware installation, adjunct powering, and patch cord installation and administration.
- Miscellaneous wiring installation procedures.

AT&T System 75—System Description **555-200-200**

Provides a technical description of the system, hardware, environmental and space requirements, and parameters. Also provides a brief description of features and services.

DEFINITY Communications System Generic 1 and System 75—Feature Description **555-200-201**

Provides a technical description of the system features and parameters. For each feature, the following information is provided:

- Limitations/considerations
- Feature interactions
- Administration requirements
- Hardware and software requirements.

DEFINITY® Communications System Generic 1 and System 75—Pocket Reference **555-200-202**

Provides the reader with a quick, pocket-sized reference to the benefits, requirements, limitations, parameters, features, and circuit packs associated with the system.

DEFINITY® Communications System Generic 1 and System 75—Administration and Measurement Reports **555-200-500**

Describes the management of the system's administration and operation. Includes the guidelines for initialization, reconfiguration, backup procedures, monitoring system performance, and maintaining system security. Includes a description of the tasks that can be performed via the administration terminal and prerequisites for completion. Also included is a description of the Traffic Measurement Reports for the system.

DEFINITY® Communications System Generic 1 and System 75—Planning/Configuration **555-200-600**

Provides information be used by the Account Team to determine the customer's requirements and to collect the information needed to estimate system hardware requirements.

AT&T System 75—Implementation Manual—R1V1 **555-200-650**

Provides the procedures and associated forms for collecting system and terminal software information. This information is used to initialize the system using the System Access Terminal.

AT&T System 75—Implementation—R1V2 **555-200-651**

Provides the procedures and associated forms for collecting system and terminal software information. This information is used to initialize the system using the System Access Terminal.

AT&T System 75—Implementation—R1V3 **555-200-652**

Provides the procedures and associated forms for collecting system and terminal software information. This information is used to initialize the system using the System Access Terminal.

DEFINITY® Communications System Generic 1 and System 75—Console Operations **555-200-700**

Provides "how-to-operate" instructions for the attendant console. Serves as a reference when defining the console control keys and Incoming Call Identification requirements.

DEFINITY® Communications System Generic 1 and System 75—Voice Terminal Operations **555-200-701**

Describes all the voice features and provides the "how-to-operate" instructions for each voice terminal. Serves as a training guide for system users.

AT&T System 75—Automatic Call Distribution—Agent Instructions **555-200-722**

Provides information for use by agents after training is completed. The various ACD features are described and the procedures for using them are provided in this document. While directly supporting System 75 R1V3, these instructions apply to DEFINITY Communications System Generic 1 also.

DEFINITY® Communications System Generic 1 and System 75—User's Guide—Hospitality Operations **555-200-723**

Contains procedures for using the Hospitality Services of DEFINITY Generic 1 and System 75 R1V3. These services include a group of system-based features that support the lodging and health industries.

AT&T System 75—Automatic Call Distribution—Supervisor Instructions **555-200-724**

Provides information for use by supervisors after training is completed. The various ACD features are described and the procedures for using them are provided in this document. While directly supporting System 75 R1V3, these instructions apply to DEFINITY Communications System Generic 1 also.

AT&T System 75 XE—System Description **555-201-200**

Provides a technical description of the system, hardware, environmental and space requirements, and parameters for System 75 XE, R1V2, and R1V3. Also provides a brief description of features and services.

DEFINITY® Communications System Generic 1—Installation and Test **555-204-104**

Provides the information necessary to perform the tasks of installing and testing the system's common equipment. Includes a description of the necessary tools and equipment.

DEFINITY® Communications System Generic 1—Maintenance **555-204-105**

Provides the information necessary for monitoring, testing, and maintaining DEFINITY Generic 1. It is intended to cover many of the faults and troubles that can occur in the system.

DEFINITY® Communications System Generic 1—Upgrades and Additions **555-204-106**

Provides procedures and information required to upgrade a System 75 V1, V2, or V3 to a DEFINITY Generic 1 system and make additions to an operational system, after the initial switch installation.

DEFINITY® Communications System Generic 1—Wiring **555-204-111**

Provides an overview of the DEFINITY Generic 1 wiring plan. It contains the same type of information as 555-200-111 (described previously).

DEFINITY® Communications System Generic 1—System Description **555-204-200**

Provides a technical description of the system and its hardware, environmental and space requirements, and parameters. Also provides a brief description of features and services.

DEFINITY® Communications System Generic 1—Enhancements **555-204-498**

Provides a description of the enhancements available on DEFINITY Generic 1.

DEFINITY® Communications System Generic 1—Capabilities **555-204-499**

Provides a description of the new capabilities, features, and enhancements available on DEFINITY Generic 1 systems that were not available on System 75 R1V3.

DEFINITY® Communications System Generic 1—Implementation **555-204-654**

Provides the procedures and associated forms for collecting system and terminal software information. This information is used to initialize the system using the DEFINITY Manager I Terminal.

DEFINITY® Communications System Generic 1—Basic Call Management System Operations **555-204-703**

Describes all the features and provides the "how-to-operate" instructions for the Basic Call Management System (BCMS) feature.

DEFINITY® Communications System Generic 1 and System 75—Application Notes—Automatic Call Distribution **555-209-013**

Describes in detail the Automatic Call Distribution (ACD) feature of System 75 R1V3 and DEFINITY Generic 1 systems. Also described are the associated "embedded" features (such as Intraflow/Interflow, Queue Status Indications, Agent Call Handling, etc.) required for efficient operation and use of the ACD feature.

DEFINITY® Communications System Generic 1—Application Notes—Outbound Call Management **555-209-014**

Describes in detail the Outbound Call Management feature of DEFINITY Generic 1 and provides information about system capabilities and requirements for implementing and using the feature.

DEFINITY® Communications System Generic 1 and System 75—Application Notes—7400B Data Module **555-209-017**

Provides guidelines for administering the 7400B Data Module in System 75 R1V1 through R1V3 and DEFINITY Communications System Generic 1.

AT&T ISDN Gateway Release 1 Version 2 Planning and Application Development **585-245-201**

Provides a description of the AT&T ISDN Gateway and information on how to plan for it. It also contains information on ISDN Gateway interfaces that can be used for software application development.

CHAPTER 6. GLOSSARY

A

Access Code

A 1-, 2-, or 3-digit dial code used to activate or cancel a feature or access an outgoing trunk. The star (*) and pound (#) can be used as the first digit of an access code.

Access Tie Trunks

Tie trunks used to handle normal ETN calls between Main and Tandem switches.

Administer

To access and change the parameters associated with the services or features of the system.

Answer-Back Code

A code dialed to retrieve a parked call.

Appearance

See Call Appearance.

Asynchronous Data Transmission

A scheme for transmitting data where each character is preceded by a start bit and followed by a stop bit, thus permitting data elements to occur at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard).

Asynchronous Data Unit (ADU)

A data communications equipment (DCE) type device that allows direct connection between RS-232C equipment and the system digital switch.

Attendant

The operator of the console.

Applications Processor

A minicomputer used to support several user-controlled applications such as traffic analysis and electronic documentation.

Attendant Console

An electronic call-handling position with pushbutton control. Used by attendants to answer and place calls and to manage and monitor some of the system operations.

Audio Information Exchange (AUDIX)

A unit that provides voice mail service to users.

Automatic Trunk

A trunk that does not require the sending or receiving of digits. The destination is predetermined. A request for service on the trunk (called a seizure) is sufficient to route the call. The normal destination of an automatic trunk is the system attendant group.

B

Barrier Code

A security code used with the Remote Access feature to prevent unauthorized access to the system.

Bit (Binary Digit)

One unit of information in binary notation (having two possible states or values, zero or one).

Bridge (Bridging)

The appearance of a voice terminal's extension at one or more other voice terminals.

Bridged Appearance

A call appearance on a voice terminal that matches a call appearance on another voice terminal for the duration of a call.

Buffer

A circuit or component that isolates one electrical circuit from another. Typically, a buffer holds data from one circuit or process until another circuit or process is ready to accept the data.

Bus

A multi-conductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

Bus, Time Division Multiplex

See Time Division Multiplex Bus.

Business Communications Terminal

An advanced series of semi-intelligent terminals.

Bypass Tie Trunks

One-way, outgoing tie trunks from a Tandem switch to a Main switch in an ETN. These trunks, provided in limited quantities, are used as a "last-choice" route when all trunks to another Tandem switch are busy. Bypass tie trunks are used only if all applicable intertandem trunks are busy.

Byte

A sequence of bits, 8 bits long, that is usually shorter than a word. A word is 16 bits long.

C

Call Appearance, Attendant Console

Six buttons, labeled a through f, used to originate, receive, and hold calls. Each button has two associated lamps to show the status of the call appearance.

Call Appearance, Voice Terminal

A button labeled with an extension number used to place outgoing calls, receive incoming calls, or hold calls. Two lamps next to the button show the status of the call appearance or status of the call.

Call Management System (CMS)

An adjunct processor that collects data from an ACD and generates reports to be stored or displayed concerning status of agents, splits, and trunks.

Callback Call

A call that is automatically returned to a voice terminal user who activated the Automatic Callback or Ringback Queuing feature.

Call Waiting Ringback Tone

A low-pitched tone identical to the ringback tone except the tone decreases the last 0.2 second. This tone notifies the attendant that the Attendant Call Waiting feature has been activated and that the called user is aware of the waiting call.

Central Office

The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

Central Office Codes

The first three digits of a 7-digit public network telephone number. These codes are numbered from 200 through 999.

Central Office Trunk

A telecommunications channel that provides access from the system to the public network through the local central office.

Channel

A communications path for transmitting voice and data.

Class of Restriction (COR)

A number (0 through 63) that specifies the restrictions assigned to voice terminals, voice terminal groups, data modules, and trunk groups.

Class of Service (COS)

A number (0 through 15) that specifies if voice terminal users can activate the Automatic Callback, Call Forwarding-All Calls, Data Privacy, or Priority Calling features.

Common Control Switching Arrangement (CCSA)

A private telecommunications network using dedicated trunks and a shared switching center for interconnecting company locations.

Confirmation Tone

Three short bursts of tone followed by silence; indicates that the feature activated, deactivated, or canceled has been accepted.

Console

See Attendant Console.

Coverage Answer Group

A group of up to eight voice terminals that ring simultaneously when a call is redirected to it by Call Coverage. Any one of the group can answer the call.

Coverage Call

A call that is automatically redirected from the called party's extension number to an alternate answering position when certain coverage criteria are met.

Coverage Path

The order in which calls are redirected to alternate answering positions.

Coverage Point

The attendant positions (as a group), Direct Department Calling group, Uniform Call Distribution group, Coverage Answer Group, a voice terminal extension, or Message Center Hunt Group designated as an alternate answering position in a coverage path.

Covering User

The person at an alternate answering position who answers a coverage call.

D

Data Channel

A communications path between two points used to transmit digital signals.

Data Communications Equipment (DCE)

The equipment on the network side of a communication link that provides all the functions required to make the binary serial data from the source or transmitter compatible with the communications channel.

Data Terminal Equipment (DTE)

Equipment comprising the source or link of data, or both, that also provides communication control functions (protocol). DTE is any piece of equipment at which a communications path begins or ends.

Delay-Dial Trunk

After a request for service (called a seizure) is detected on an incoming trunk, the system sends a momentary signal followed by a steady tone over the trunk. This informs the calling party that dialing can start. This type of trunk allows dialing directly into the system. That is, the digits are received as they are dialed.

Designated Voice Terminal

The specific voice terminal to which calls, originally directed to a certain extension number, are redirected. Commonly used to mean the "forwarded-to" terminal when Call Forwarding All Calls is active.

Dial Repeating Tie Trunk

A telecommunications channel between two private switching systems. The number dialed is repeated or dialed-in at the distant end.

Digital Communications Protocol (DCP)

Defines the capability for providing simultaneous voice and data transmission over the same channel.

Digital Data Endpoints

Digital data endpoints include the following:

- 510D Personal Terminal or 515-Type Business Communications Terminal
- 7404D Terminals
- 7406D or 7407D Equipped With Optional Data Module Base
- Asynchronous Data Units
- Digital Terminal Data Modules
- (Modular) Processor Data Modules
- (Modular) Trunk Data Modules
- 3270 Data Modules
- Internal Data Channels.

Digital Multiplexed Interface (DMI)

Specifies the remote interface requirements for multiplexed data communications between a host computer and a private switching system.

Digital Terminal Data Module (DTDM)

An adjunct to Model 7403D or 7405D voice terminals. Provides the required interface between the system and a data terminal such as a 513 Business Communications Terminal.

Digital Trunk

A circuit in a telecommunications channel designed to handle digital voice and data.

Direct Extension Selection (DXS)

An option at the attendant console that allows an attendant direct access to voice terminals by pressing a Group Select button and a DXS button.

Distributed Communications System (DCS)

A network of two or more switches, each with its terminals and trunks, configured to function as a single large system.

E

Electronic Tandem Network (ETN)

A special tandem tie trunk network that has automatic call routing capabilities based on the number dialed and most preferred route available at the time the call is placed. Each switch in the network is assigned a unique private network office code (RNX) and each voice terminal is assigned a unique extension number.

End-to-End Signaling

The transmission of touch-tone signals generated by dialing from a voice terminal user to remote computer equipment. A connection must first be established over an outgoing trunk from the calling party to the computer equipment. Then additional digits can be dialed to transmit information to be processed by the computer equipment.

Enhanced Private Switched Communications Service (EPSCS)

A private telecommunications network that provides advanced voice and data telecommunications services to companies with many locations.

Extension Number

A 1- to 5-digit number assigned to each voice terminal, certain system groups, data modules, 510D Personal Terminal, or 515 Business Communications Terminal within the system. A 1- or a 5-digit extension number is available for Version 2 and Version 3.

External Call

A connection between a system user and a party on the public telephone network or on a tie trunk.

F

Facility

A general term used for the telecommunications transmission pathway and associated equipment.

Feature

A specifically defined function or service provided by the system.

Feature Button

A labeled button on a voice terminal or attendant console designating a specific feature.

Foreign Exchange (FX)

A central office other than the one providing local access to the public telephone network.

Foreign Exchange Trunk

A telecommunications channel that directly connects the system to a central office other than its local central office.

Foreign Numbering Plan Area Code

An area code other than the local area code. The foreign area code must be dialed to call outside the local geographical area.

G

Ground-Start Trunk

On outgoing calls, the system transmits a request for services to the distant switching system by grounding the trunk ring lead. When the distant system is ready to receive the digits of the called number, that system grounds the trunk tip lead. When the system detects this ground, the digits are sent. (Tip and ring are common nomenclature to differentiate between ground-start trunk leads.) On incoming calls, detection of ground on the ring lead is sufficient to cause the call to route to a predetermined destination, normally the system attendant group. No digits are received.

H

Handshaking Logic

A format used to initiate a data connection between two data module devices.

Home Numbering Plan Area Code

The local area code. The area code does not nave to be dialed to call numbers within the local geographical area.

Immediate-Start Tie Trunk

After establishing a connection with the distant switching system for an outgoing call, the system waits a nominal 65 milliseconds before sending the digits of the called number. This allows time for the distant system to prepare to receive the digits. Similarly, on an incoming call, the system has less than 65 milliseconds to prepare to receive the digits.

Information Exchange

The exchange of data between users of two different systems (the system and host computer) over a local area network.

In-Use Lamp

A red lamp on a multi-appearance voice terminal that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

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.

Internal Call

A connection between two users within the system.

L

Link

A transmitter-receiver channel or system that connects two locations.

Loop-Start Trunk

After establishing a connection with the distant switching system for an outgoing call, the system waits for a signal on the loop formed by the trunk leads before sending the digits of the called number. On incoming calls, the received request for service is sufficient to cause the call to route to a predetermined destination, normally the system attendant group. No digits are received.

M

Main/Satellite/Tributary

A Main switch provides: interconnection, via tie trunks, with one or more subtending switches, called Satellites; all attendant positions for the Main/Satellite configuration; and, access to and from the public network. To a user outside the complex, a Main/Satellite configuration appears as a single switch, with a single Listed Directory Number (LDN). A Tributary is a switch, connected to the Main via tie trunks, but which has its own attendant position(s) and its own LDN.

Message Center

An answering service for calls that might otherwise go unanswered; an agent accepts and stores messages for later retrieval. (Requires an Applications Processor.)

Message Center Agent

A member of the Message Center Hunt Group who takes and retrieves messages for voice terminal users.

Modular Processor Data Module

See Processor Data Module.

Modular Trunk Data Module

See Trunk Data Module.

Modem Pooling

Provides shared-use conversion resources that eliminate the need for a dedicated modem when a data module accesses, or is accessed by, an analog line or trunk.

Multi-Appearance Voice Terminal

A terminal equipped with several call appearance buttons for the same extension number. Allows the user to handle more than one call, on that same extension number, at the same time.

Multiplexer

A device for simultaneous transmission of two or more signals over a common transmission medium.

N

Network

An arrangement of inter and/or intra location circuits designed to perform specific functions.

P

Paging Trunk

A telecommunications channel used to access an amplifier for loudspeaker paging.

Pickup Group

A group of individuals authorized to answer any call directed to an extension number within the group.

Port

A designation of the location of a circuit that provides an interface between the system and lines and/or trunks.

Principal (User)

In terms of Call Coverage, a person for whom a call was originally intended.

Private Network

A network used exclusively for handling the telecommunications needs of a particular customer.

Private Network Office Code (RNX)

The first three digits of a 7-digit private network number. These codes are numbered 220 through 999, excluding any codes that have a 0 or 1 as the second digit.

Processor Data Module (PDM)

Provides the required interface between the system and an EIA computer or data terminal.

Property Management System (PMS)

A stand-alone computer which Lodging and Health Services organizations use for services such as reservations, housekeeping, billing, etc.

Protocol

A set of conventions or rules governing the format and timing of message exchanges to control data movement and correction of errors.

Public Network

The network that can be openly accessed by all customers for local or long-distance calling.

Q

Queue

An ordered sequence of calls waiting to be processed.

Queuing

The process of holding calls in order of their arrival to await connection to an attendant, to an answering group, or to an idle trunk. Calls are automatically connected in first-in, first-out sequence.

R

Random Access Memory (RAM)

A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

Read Only Memory (ROM)

A storage arrangement primarily for information retrieval applications.

Recall Dial Tone

Three short bursts of tone followed by steady dial tone: indicates the system has completed some action (such as holding a call) and is ready to accept dialing.

Redirection Criteria

The information administered for each voice terminal's coverage path that determines when an incoming call is redirected to coverage.

Remote Home Numbering Plan Area Code (RHNPA)

A foreign numbering plan area code that is treated as a home area code by the Automatic Route Selection feature. Calls can be allowed or denied based on the area code and the dialed central office code rather than just the area code. If the call is allowed, the Automatic Route Selection pattern used for the call is determined by these six digits.

Removable Mass Storage Subsystem (RMSS)

A tape storage device that stores the software information for the system.

Reorder Tone

A fast-busy tone repeated 120 times a minute; indicates that at least one of the facilities, such as a trunk or a digit transmitter, required for the call was not available at the time the call was placed.

S

Single-Line Voice Terminals

Voice terminals served by a single-line tip and ring circuit (Models 500, 2500, 7101A, 7103A).

Software

A set of computer programs that accomplish one or more tasks.

Split

A condition whereby a caller is temporarily separated from a connection with the attendant. This split condition automatically occurs when the attendant, active on a call, presses the Start button.

Standard Serial Interface (SSI)

A communications protocol developed by AT&T Teletype Corporation for use with the 500 Business Communications Terminals and the 400-series printers.

Status Lamp

A green lamp that shows the status of a call appearance or a feature button by the state of the lamp (lighted, flashing, fluttering, broken flutter, or dark).

Switchhook

The button(s) on a voice terminal located under the receiver.

Synchronous Data Transmission

A scheme for sending and receiving data, where data elements may occur only at regular specified times. Sending and receiving devices must operate in step with each other.

System Manager

A person responsible for specifying and administering features and services for the system.

System Reload

A process that allows stored data to be written from a tape into the system memory (normally after a power outage).

T

Tandem Switch

A switch within an ETN that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.

Tandem Through

The switched connection of an incoming trunk to an outgoing trunk without human intervention.

Tandem Tie Trunk Network

A private network that interconnects several customer switching systems by dial repeating tie trunks. Access to the various systems is dictated by codes that must be individually dialed for each system.

Tie Trunk

A telecommunications channel that directly connects two private switching systems.

Time Division Multiplex Bus

A special bus that is time shared by preallocating short time slots to each transmitter on a regular basis. In a PBX, all port circuits are connected to the time division multiplex bus permitting any port to send a signal to any other port.

Tone Ringer

A device with a speaker, used in electronic voice terminals to alert the user.

Trunk

A telecommunications channel between two switching systems.

Trunk Data Module

Provides the required interface between the system and a data set (modem) or data service unit connected to a private or switched data line.

Trunk Group

Telecommunications channels assigned as a group for certain functions.

U

Uniform Dial Plan

A feature that allows a unique 4- or 5-digit number assignment for each terminal in a multi-switch configuration, such as a Distributed Communications System (DCS) or Main/Satellite/Tributary configuration.

V

Voice Terminal

A single-line or multi-appearance voice instrument (telephone).

W

Wide Area Telecommunications Service (WATS)

A service that allows calls to a certain area or areas for a flat-rate charge based on expected usage.

Wink-Start Tie Trunk

After establishing a connection with a distant switching system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

Write Operation

The process of putting information onto a storage medium such as magnetic tape.

8

800 Service

A service that allows incoming calls from a certain area or areas to an assigned number for a flat-rate charge based on usage.

CHAPTER 7. ABBREVIATIONS AND ACRONYMS

AAR	Automatic Alternate Routing
AC	Alternating Current
ACA	Automatic Circuit Assurance
ACD	Automatic Call Distribution
ACU	Automatic Call Unit
ACW	After Call Work
AD	Abbreviated Dialing
ADU	Asynchronous Data Unit
AIM	Asynchronous Interface Module
ALM-ACK	Alarm Acknowledge
AMW	Automatic Message Waiting
ANI	Automatic Number Identification
AP	Applications Processor
APLT	Advanced Private Line Termination
ARS	Automatic Route Selection
ASCII	American Standard Code for Information Interchange
ATB	All Trunks Busy
AUDIX	Audio Information Exchange
AVD	Alternate Voice Data
AWT	Average Work Time
BCC	Bearer Capability Class
BCMS	Basic Call Management **System**
BCT	Business Communications Terminal
BHCC	Busy Hour Calls Completions
BLF	Busy Lamp Field
BOS	Bit Oriented Signaling
BTU	British Thermal Unit
CAMA	Centralized Automatic Message Accounting
CACR	Cancellation of Authorization Code Request
CAS	Centralized Attendant Service
CBC	Call-by-Call
Cell-r	Consultative Committee for International Telephone and Telegraph
CCMS	Common Channel Message Set
CCS	Hundred Call Seconds
CCSA	Common Control Switching Arrangement
CDM	Channel Division Multiplexing
CDOS	Customer-Dialed and Operator Serviced
CDRR	Call Detail Recording and Reporting
CDRU	Call Detail Recording Utility
CEM	Channel Expansion Multiplex
CMDR	Centralized Message Detail Recorder

CMS	Call Management System
CO	Central Office
COR	Class of Restriction
COS	Class of Service
CP	Circuit Pack
CPE	Customer Premises Equipment
CPTR	Call Progress Tone Receiver
CRC	Cyclical Redundancy Checking
CSA	Canadian Safety Association
CSM	Centralized System Management
CSSO	Customer Services Support Organization
DO	Direct Current
DCE	Data Communications Equipment
DCP	Digital Communications Protocol
DCS	Distributed Communications System
DDC	Direct Department Calling
DDD	Direct Distance Dialing
DID	Direct Inward Dialing
DLC	Data Line Circuit
DLDM	Data Line Data Module
DMI	Digital Multiplexed Interface
DND	Do Not Disturb
DOD	Direct Outward Dialing
DOSS	Delivery Operations Support System
DS1	Data Services Level 1
DSU	Data Service Unit
DTDM	Digital Terminal Data Module
DTE	Data Terminal Equipment
DTGS	Direct Trunk Group Select
DTMF	Dual Tone Multifrequency
DXS	Direct Extension Selection
E&M	Ear and Mouth (Receive and Transmit)
EBCDIC	Extended Binary Coded Decimal Inter-exchange Code
EI	Expansion Interface
EIA	Electronic Industries Association
EMI	Electro-Magnetic Interference
EPN	Expansion Port Network
EPROM	Erasable Programmable Read Only Memory
EPSCS	Enhanced Private Switched Communications Services
ESF	Extended Superframe Format
ETN	Electronic Tandem Network
FAC	Feature Access Code
FCC	Federal Communications Commission
FIC	Facility Interface Codes
FNPA	Foreign Numbering Plan Area Code

FRL	Facility Restriction Level
FX	Foreign Exchange
GPTR	General Purpose Tone Receiver
GRS	Generalized Route Selection
HNPA	Home Numbering Plan Area Code
IAS	Inter-PBX Attendant Service
ICC	Inter Carrier Cable
ICI	Incoming Call Identifier
IDDD	International Direct Distance Dialing
IE	Information Elements
INADS	Initialization and Administration System
INWATS	Inward Wide Area Telephone Service
INS	ISDN Network Service
ISDN	Integrated Services Digital Network
ISN	Information Systems Network
IXC	Inter-Exchange Carrier Code
kbps	Kilo-Bits Per Second
LAN	Local Area Network
LAPD	Link Access Procedure D
LDN	Listed Directory Number
LED	Light-Emitting Diode
LSU	Local Storage Units
LWC	Leave Word Calling
MA-UUI	Message Associated User-to-User Signaling
M-Bus	Memory Bus
Mbps	Mega-Bits Per Second
MCC	Multi-Carrier Cabinet
MCS	Message Center Service
MDM	Modular Data Module
MDR	Message Detail Record
MET	Multibutton Electronic Telephone
MIS	Management Information System
MISCID	Miscellaneous Identification
MOS	Message Oriented Signaling
MS	Message Server
MSA	Message Service Adjunct
MPDM	Modular Processor Data Module
MTDM	Modular Trunk Data Module
MTP	Maintenance Tape Processor
MWL	Message Waiting Lamp
NAU	Network Access Unit

NCOSS	Network Control Operations Support Center
NEC	National Engineering Center
NID	Network Inward Dialing
NPA	Numbering Plan Area Code
NPE	Network Processing Element
NQC	Number of Queued Calls
NSE	Night Service Extension
NSU	Network Sharing Unit
NXX	Public Network Office Code
OCM	Outbound Call Management
OPS	Off-Premises Station
OQT	Oldest Queued Time
OSHA	Occupational Safety and Health Act
PBX	Private Branch Exchange
PC	Personal Computer
PCOL	Personal Central Office Line
PCOLG	Personal Central Office Line Group
PCM	Pulse Code Modulated
PCS	Permanent Switched Calls
PDM	Processor Data Module
PDS	Premises Distribution System
PGN	Partitioned Group Number
PIB	Processor Interface Board
PL	Private Line
PMS	Property Management System
PN	Port Network
PPN	Processor Port Network
PRI	Primary Rate Interface
PSC	Premises Service Consultant
PSDN	Packet Switch Public Data Network
PT	Personal Terminal
RAM	Random Access Memory
RHNPA	Remote Home Numbering Plan Area Code
RLT	Release Link Trunk
RNX	Private Network Office Code
ROM	Random Access Memory
RPN	Routing Plan Number
RSC	Regional Support Center
SAKI	Sanity and Control Interface
SAT	System Access Terminal
SCC	Single-Carrier Cabinet
SCI	Switch Communications Interface
SCO	System Control Office
SDN	Software Defined Network

SID	Station Identification Number
SMDR	Station Message Detail Recording
SPE	Switch Processing Element
SSI	Standard Serial Interface
STARLAN	Star-based Local Area Network
TAAS	Trunk Answer From Any Station
TAC	Trunk Access Code
TCM	Traveling Class Mark
TDM	Trunk Data Module
TEG	Terminating Extension Groups
TOD	Time of Day
TOP	Task Oriented Protocol
TTTN	Tandem Tie Trunk Network
TTY	Teletypewriter
UAP	Usage Allocation Plan
UCD	Uniform Call Distribution
UDP	Uniform Dial Plan
UPS	Uninterruptible Power Supply
WATS	Wide Area Telecommunications Service

CHAPTER 8. INDEX

C

D