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DEFINITY® Communications System Generic 1 and Generic 3

Feature Description

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Part 15: Class A Statement. This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio-frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his or her own expense.

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AS593M-14695-MA-E.

Trademarks

Ordering Information

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Acknowledgment

This document was prepared by the AT&T Product Documentation Development Department, Denver, CO 80234-2703.

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About This Document

This manual provides a technical description of the system features and parameters for DEFINITY® Communications System Generic 3i (G3i) and DEFINITY® Communications System Generic 1 (G1.1).

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

This document also provides information about the International DEFINITY Communications System Generic 1. Information that applies only to the international product is clearly marked as such. Unless otherwise noted, information pertaining to G3 and G3i-U.S. also applies to G3i-Global.

The term ''DEFINITY Generic 3i'' refers to the DEFINITY Communications System Generic 3i (G3i). The term ''DEFINITY Generic 1'' refers to the DEFINITY Communications System Generic 1 (G1.1).

Purpose

This manual, along with *DEFINITY® Communications System Generic 1 and Generic* 3i — 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 *DEFINITY® Communications System Generic 3i — Implementation, 555-230-650* for G3i and *555-204-654* for G1.1, 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 G3i enhancements, as well as other miscellaneous information.

The new features and functions added for G3i include the following:

- Access Endpoints (See the Administered Connections feature)

- CallVisor™ Adjunct/Switch Application Interface (ASAI)
- Administered Connections
- Administration Without Hardware
- Alphanumeric Dialing
- Call Prompting
- Call Vectoring
- DCS Over ISDN-PRI D-Channel
- Default Dialing
- Facility and Non-Facility Associated Signaling
- Inbound Call Management
- Integrated Services Digital Network — Basic Rate Interface (ISDN-BRI)
- Look Ahead Interflow
- Restriction — Toll
- Security Violation Notification

Other features with significant G3i enhancements are as follows:

- Automatic Route Selection — New feature description for G3i
- Automatic Alternate Routing — New feature description for G3i
- Agent Call Handling — Stroke Counts, Call Work Codes, and Forced Entry of Stroke Counts and Call Work Codes have been added
- Automatic Call Distribution — Direct Agent Calling has been added
- Class of Restriction — New restrictions have been added
- Integrated Services Digital Network — Primary Rate Interface (ISDN-PRI) — Software Defined Data Network (SDDN) has been added
- Call Detail Recording — New CDR record formats have been added

Organization

This manual is organized as follows:

- **About This Document** provides an introduction to this manual.
- **Chapter 1: 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 Telemarketing. 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 2: Feature Descriptions** provides a detailed description of the

features associated with Voice Management, Data Management, Network Services, System Management, Hospitality Services, and Telemarketing. These feature descriptions are arranged in alphabetical order, regardless of function area.

- **Chapter 3: 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 4: References** provides a list of reference documentation. A brief description of each document is included.
- **Abbreviations and Acronyms** provides a list of all abbreviations and acronyms used in this manual.
- **Glossary:** provides a glossary for the entire manual.
- **Index:** provides an index for the entire manual.

Conventions Used in This Document

This manual uses the following conventions:

- The names of commands are shown in the following typeface: **change system-parameters feature**
- Information you type is shown in the following typeface: `EIA`
- Information displayed on the screen is shown in the following typeface: `login:`
- Keyboard keys are shown as follows: ***RETURN***
- Function keys are shown as follows: **CANCEL**
- Telephone dialpad buttons are shown as follows: **2** or *****, and may refer to either rotary or touch-tone telephones.

Functional Description

1

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 Telemarketing. 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 2 of this manual.

Voice Management

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

Voice Management Features

The following features are associated with Voice Management:

1. Abbreviated Dialing
2. Attendant Auto-Manual Splitting
3. Attendant Call Waiting
4. Attendant Control of Trunk Group Access
5. Attendant Direct Extension Selection With Busy Lamp Field
6. Attendant Direct Trunk Group Selection

7. Attendant Display
8. Attendant Recall
9. Attendant Release Loop Operation
10. Audio Information Exchange (AUDIX) Interface
11. Authorization Codes
12. Automatic Callback
13. Automatic Incoming Call Display
14. Bridged Call Appearance — Multi-Appearance Voice Terminal
15. Bridged Call Appearance — Single-Line Voice Terminal
16. Busy Verification of Terminals and Trunks
17. Call By Call Service Selection
18. Call Coverage
19. Call Forwarding All Calls
20. Call Park
21. Call Pickup
22. Call Waiting Termination
23. Centralized Attendant Service
24. Class of Restriction
25. Class of Service
26. Code Calling Access
27. Conference — Attendant
28. Conference — Terminal
29. Consult
30. Coverage Callback
31. Coverage Incoming Call Identification
32. Dial Access to Attendant
33. Dial Plan
34. Direct Department Calling and Uniform Call Distribution
35. Direct Inward Dialing
36. Direct Outward Dialing
37. Distinctive Ringing
38. Emergency Access to the Attendant
39. Facility Busy Indication

40. Forced Entry of Account Codes
41. Go To Cover
42. Hold
43. Hot Line Service
44. Hunting
45. Individual Attendant Access
46. Integrated Directory
47. Integrated Services Digital Network — Basic Rate Interface (ISDN-BRI)
48. Intercept Treatment
49. Intercom — Automatic
50. Intercom — Dial
51. Inter-PBX Attendant Calls
52. Last Number Dialed
53. Leave Word Calling
54. Line Lockout
55. Loudspeaker Paging Access
56. Loudspeaker Paging Access — Deluxe
57. Manual Message Waiting
58. Manual Originating Line Service
59. Manual Signaling
60. Multi-Appearance Preselection and Preference
61. Multiple Listed Directory Numbers
62. Music-on-Hold Access
63. Night Service — Hunt Group
64. Night Service — Night Console Service
65. Night Service — Night Station Service
66. Night Service — Trunk Answer From Any Station
67. Night Service — Trunk Group
68. Personal Central Office Line
69. Personalized Ringing
70. Power Failure Transfer
71. Priority Calling
72. Privacy — Attendant Lockout

73. Privacy — Manual Exclusion
74. Recall Signaling
75. Recorded Announcement
76. Recorded Telephone Dictation Access
77. Remote Access
78. Restriction — Controlled
79. Restriction — Miscellaneous Terminal
80. Restriction — Miscellaneous Trunk
81. Restriction — Toll (G3i)
82. Restriction — Toll/Code
83. Restriction — Voice Terminal — Inward
84. Restriction — Voice Terminal — Manual Terminating Line
85. Restriction — Voice Terminal — Origination
86. Restriction — Voice Terminal — Outward
87. Restriction — Voice Terminal — Termination
88. Ringback Queuing
89. Ringer Cutoff
90. Rotary Dialing
91. Send All Calls
92. Senderized Operation
93. Single-Digit Dialing and Mixed Station Numbering
94. CDR Account Code Dialing
95. Straightforward Outward Completion
96. Temporary Bridged Appearance
97. Terminating Extension Group
98. Through Dialing
99. Timed Reminder
100. Touch-Tone Dialing
101. Transfer
102. Trunk Flash
103. Trunk Group Busy/Warning Indicators to Attendant
104. Trunk Identification by Attendant
105. Trunk-to-Trunk Transfer

106. Voice Message Retrieval
107. Voice Terminal Display.

Data Management

DEFINITY Generic 3i is a private digital switching system that permits 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.

Internationally, modems that comply with the CCITT 108.1 signaling procedures are supported. Modem pools are not supported in countries requiring "a law" companding mode. Administration forms have been modified to support these modems.

The physical connection can be through a digital data module, analog modem, or access endpoint (G3i).

The system provides the customer with options for selecting data modules [or data-like devices such as a Data Line Circuit (DLC)] 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 Trunk Data Module (TDM), a Z702AL1-DSU Data Module Base, a 7400A Data Module, a 7400B Data Module, a 7500B Data Module, an ISDN Asynchronous Data Module (ADM), and a 3270 Data Module. The data modules are generally more versatile than modems, operate at faster data rates, and provide additional features.

The DTDM provides synchronous or asynchronous data communications to 7403D and 7405D digital voice terminal users who have a terminal or personal computer. The DTDM and voice terminal integrate data and voice into the Digital Communications Protocol (DCP) to the digital switch.

The Z702AL1-DSU Data Module Base provides the Data Communications Equipment (DCE) interface connection for a 7407D voice terminal to data terminals. The module provides full-duplex asynchronous operation only. The module and 7407D voice terminal integrate data and voice into the DCP to the digital switch.

The MTDM provides an EIA 232C Data Terminal Equipment (DTE) interface for connection to off-premises (out of building) private-line trunk facilities, or a switched telecommunications network, and a DCP interface for connection to the digital switch. The MTDM may also serve as part of a conversion resource for modem pooling. The MTDM is also used to interface with DCE-type multiplexers.

The MPDM provides a DCE asynchronous or synchronous interface for connection to data terminals, Call Detail Recording (CDR) output devices, Manager I terminals (G1), DEFINITY Communications System Generic 3 Management Terminals (G3-MT), on-premises (in building) administration terminals, and host computers. The MPDM can be preset in the factory to provide the following interfaces: EIA 232C, RS-449, V.35, and RS-366 to support Automatic Calling Unit (ACU) type dialing. The MPDM can be configured to support the Data Call Setup or Off-Premises Data-Only Extension feature. The MPDM also supports data rates of 56 and 64 kbps for downloading and other high-speed data transfer requirements.

The 7400A Data Module may be used instead of an MTDM when supporting the combined Modem Pooling feature. The 7400A Data Module supports asynchronous operation and provides a DCP interface to the switch and an EIA 232C interface to the associated modem. The 7400A Data Module also can be used with a data terminal and supports keyboard dialing in the same manner as the MPDM.

The 7400B Data Module supports asynchronous data communications and can operate in the stand-alone mode for data-only service or in the linked mode, which provides simultaneous voice and data service (electrically acts like a DTDM). The 7400B provides data communications to 7400D series voice terminals and the 602A1 CALLMASTER® digital voice terminal that have a connection to a data terminal or personal computer. The 7400B integrates data and voice into the DCP protocol required to interface the switch via a port on a Digital Line circuit pack. The 7400B may be used instead of an MPDM when asynchronous operation at speeds of 19.2 kbps or less is required to provide a DCP interface to the switch for data terminals, printers, and so on. The 7400B does not support synchronous operation and keyboard dialing.

The 7500B Data Module is a stand-alone unit that supports asynchronous or synchronous DCE and asynchronous DTE on the Basic Rate ISDN (BRI) switch interface (DEFINITY G3i, DEFINITY G2, and the 5ESS switch). In asynchronous mode, the 7500B supports packet or circuit-switched data communications, and can be controlled via the front panel or the keyboard of a connected terminal. In synchronous mode, the 7500B supports circuit-switched or nailed-up data communications, requires either the Multipurpose Enhancement Board or the High-Speed Synchronous Enhancement Board, and only can be controlled via the front panel.

When configured as an asynchronous DCE, the 7500B provides an EIA-232D interface and supports full-duplex data transmission at rates of 300, 1200, 2400, 4800, 9600, and 19200 bps. The following optional enhancements are available for the 7500B in an asynchronous DCE configuration: an RS-366 ACU interface and a second asynchronous EIA-232D interface. With an additional asynchronous EIA-232D interface, the 7500B can simultaneously support either two D-channel packet data calls or one D-channel packet call and one B-channel circuit call. However, the 7500B cannot simultaneously support two B-channel circuit-switched calls.

When configured as an asynchronous DTE, the 7500B provides an EIA-232D interface and supports full-duplex data transmission at rates of up to 19200 bps. This configuration is most commonly used for modem pooling applications.

In order to be configured as a synchronous DCE, the 7500B must have either the Multipurpose Enhancement Board or the High-Speed Synchronous Enhancement Board. With the Multipurpose Board, the 7500B provides an EIA-232D interface and an RS-366 ACU interface, and supports full-duplex data transmission at rates of 1200, 2400, 4800, 9600, 19200, 56000, and 64000 bps. The 7500B also supports half-duplex emulation at rates of 1200, 2400, 4800, 9600, 19200, and 56000 bps. With the High-Speed Synchronous Enhancement Board, the 7500B provides a V.35 interface and supports full-duplex data transmission at rates of 48000, 56000, and 64000 bps. The 7500B only provides half-duplex emulation at a rate of 56000 bps. Regardless of the configuration, the 7500B provides no voice functions and is not used with voice terminals.

The ISDN ADM may be used with asynchronous DTE as a data stand for 7500-series BRI voice terminals. Consisting of a board located inside the BRI voice terminal, the ISDN ADM allows the transmission of integrated voice and data through one voice terminal. The ISDN ADM supports the Hayes command set for compatibility with PC communications packages.

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) EIA 232C interface, the 515 BCT can connect to other data equipment.

The DLC, which provides eight ports to connect user's asynchronous EIA 232C interface to DTE, can be used as an alternative to DTDM or PDM.

All data modules except the MPDM and 3270 provide a modified EIA 232C interface. The MPDM provides either EIA 232C V.35 or RS-449 interface. The MPDM can also emulate an 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 2. The 3270 Data Module provides a Category A coaxial DCE interface for connection to 3270-type data terminals or a cluster controller. It also provides a 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.

Trunks or channels of a DS1 can also be used as non-signaling data endpoints with the Access Endpoints function. An access endpoint is either a non-signaling channel on a DS1 interface or a non-signaling port on an Analog Tie Trunk circuit pack that is assigned a unique extension. Since an access endpoint is non-signaling, it will neither generate nor respond to signaling. As a result, an access endpoint cannot be used as a trunking facility (it cannot receive incoming calls or route outgoing calls). An access endpoint is used primarily to support devices, switches, or services that have a trunk interface but do not support signaling for the trunk. An access endpoint may be designated as the originating (local) endpoint or destination endpoint in an Administered Connection. The status of an access endpoint can be displayed by entering the **status access-endpoint** command from the Manager I Terminal (G1) or the G3 Management Terminal.

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 DEFINITY Manager I terminals (G1), G3 Management Terminals, and Call Detail Recording (CDR)]. 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 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.

With G3i, the ISDN-BRI provides one 16 kbps signaling channel (D-channel) and two 64 kbps information channels (B-channels), with a total information rate of 144 kbps. The primary purpose of the signaling channel is to convey Q.931 message-oriented signaling for the setup and tear down of calls carried by the B-Channels on the BRI. Since all the signaling is done on the D-channel, both B-channels are "clear." As a result, the entire widths of the B-channels are used for simultaneously carrying voice and circuit-switched data. Voice and data information can be simultaneously transmitted on calls to or from a 7505, 7506, or 7507 voice terminal equipped with an optional ADM. Without the optional ADM, the 7505, 7506, and 7507 voice terminals can handle voice-only calls.

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 1-1.

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. Switched Access systems can emulate hardwired networks through use of the Administered Connections feature.

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

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.

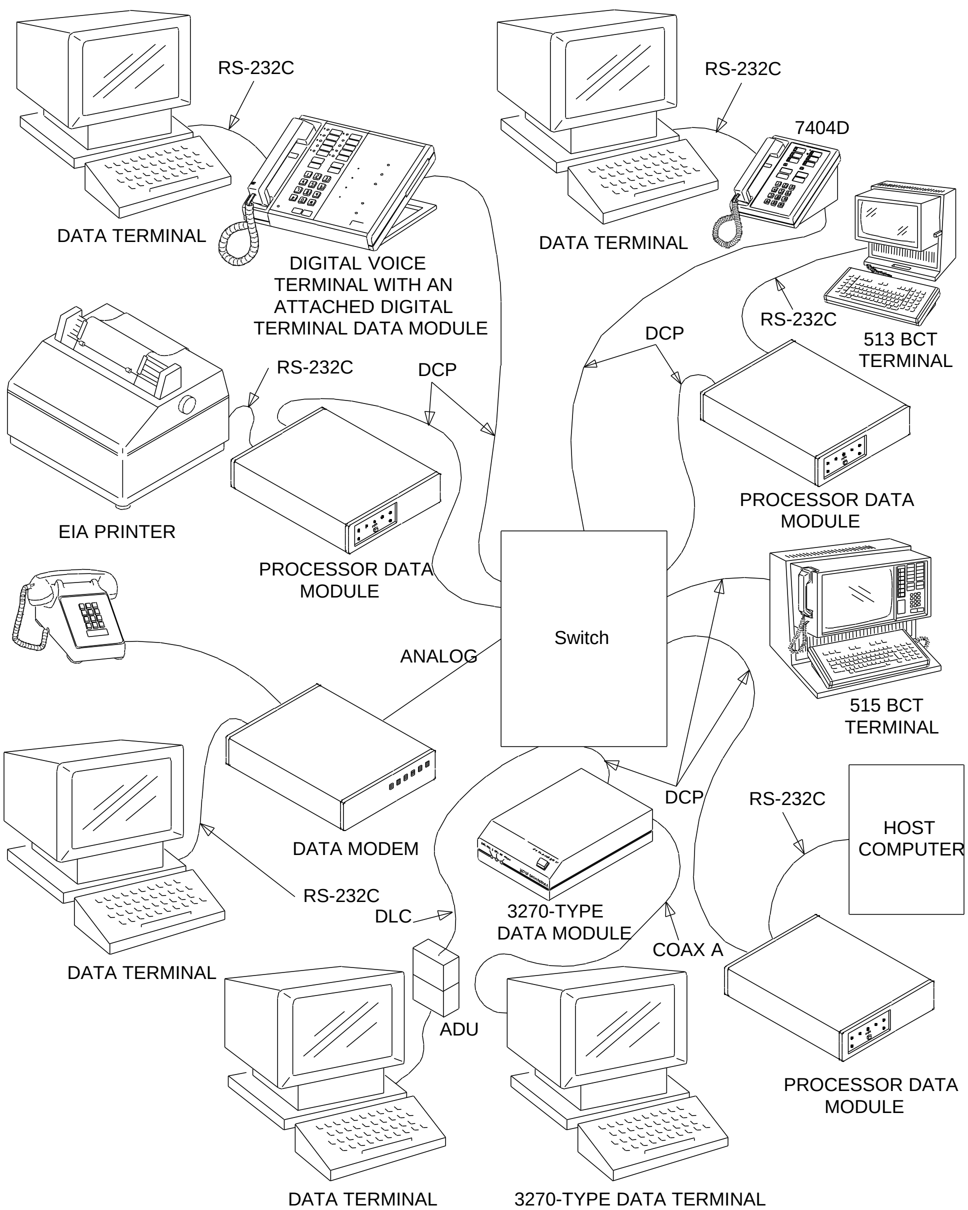


Figure 1-1. System Data Communications Configuration

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 (LANs)**

  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 1-2.

  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)

  — Software Defined Data Network (SDDN) (G3i)

  — 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)

  — Software Defined Network (SDN)

  — Software Defined Data Network (SDDN)

  — Tandem tie trunk

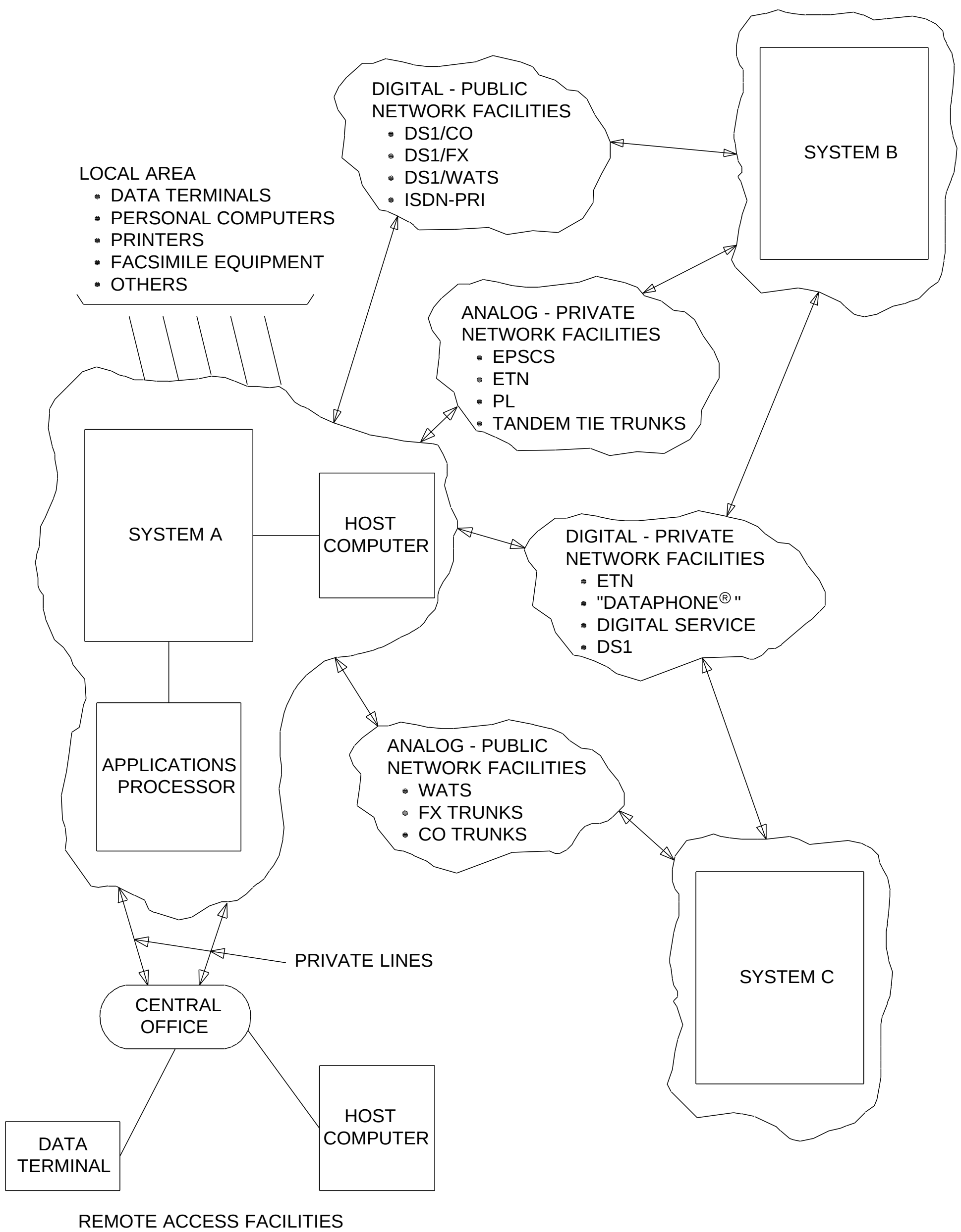


Figure 1-2. System Networking Configurations

Data Management Features

The following features are associated with Data Management:

1. Administered Connections (G3i)

2. Alphanumeric Dialing (G3i)
3. Data Call Setup [including Default Dialing (G3i) and Alphanumeric Dialing (G3i)]
4. Data Hot Line
5. Data-Only Off-Premises Extensions
6. Data Privacy
7. Data Restriction
8. Default Dialing (G3i)
9. Digital Multiplexed Interface
10. DS1 Tie Trunk Service
11. EIA Interface
12. Information System Network (ISN) Interface
13. Modem Pooling
14. PC/PBX Connection
15. Uniform Call Distribution

Data Communications Protocols and Interfaces

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 and communications processing software provide the following protocols:

- ISDN Protocols
- EIA 232C
- RS-449
- RS-366
- Standard Serial Interface (SSI)
- Teletypewriter (TTY) Modes

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

ISDN Protocols

The ISDN Q.931 Protocol is used to support Layer 3 call control signaling for both the network and user sides of an ISDN Primary Rate Interface (PRI). Both T1 and E1 digital transmission standards are supported on a per-interface basis. This implementation provides call state transition, proper message content, and error recovery, as well as protocol support for other related features. With G3i, the ISDN Q.932 Protocol is used to support the CallVisor Adjunct Switch Applications Interface (ASAI) required for the Inbound Call Management feature.

Electronic Industries Association (EIA)

EIA-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 EIA-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 EIA-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 EIA-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 — 1,600 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 EIA-232C interface when used with a 37- to 25-pin cable adapter. Since the AP RS-449 interface is compatible with the EIA-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 ACUs. This permits a computer to originate data calls over a switched telephone network. The AP provides one RS-366 interface for each six EIA-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-AWG standard building cable at distances up to 5,000 feet (1,524 meters). Cable connections are made through the eight-pin modular-type connectors.

Teletypewriter (TTY) Modes

The AP EIA-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 Intercharge (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 precoded 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, four-wire serial data link that operates full-duplex over standard twisted-pair building cable. For data-only transmission, the maximum cable length is 5,000 feet (1,524 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 1-3). The first field is a unique three-bit framing pattern that defines the frame boundary. The second field is a one-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 eight bits each and are used to send digitized voice or digital data.

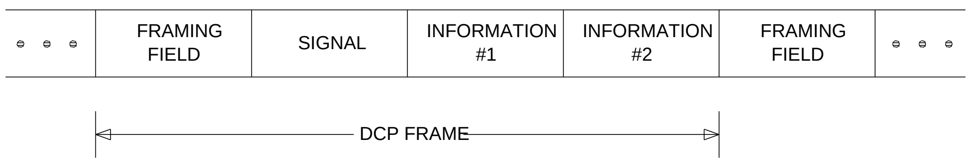


Figure 1-3. Digital Communications Protocol Frame Structure

There are 8,000 frames per second. Therefore, the bit rate available is eight 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 8,000 per second and eight 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 Application, Presentation, and Session. The Application and Presentation layers (see Table 1-1) 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

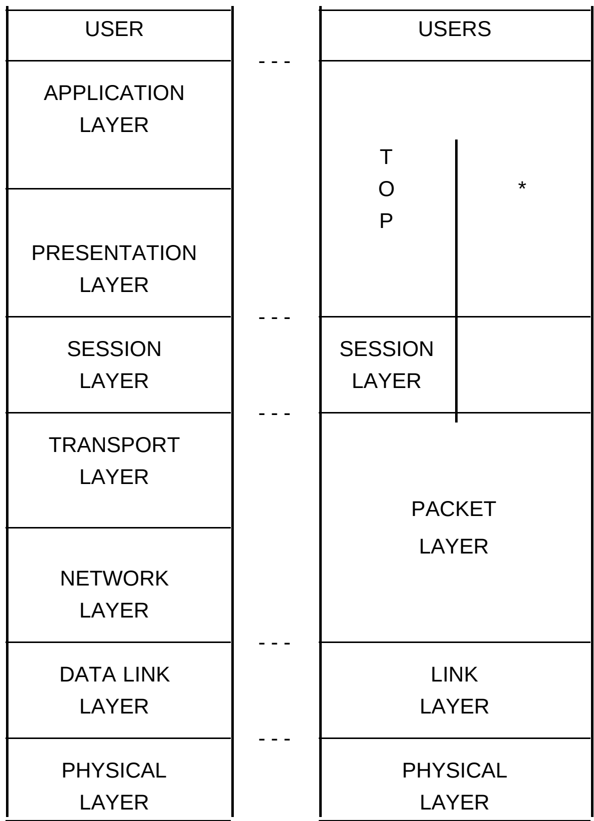
continental United States.

- 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

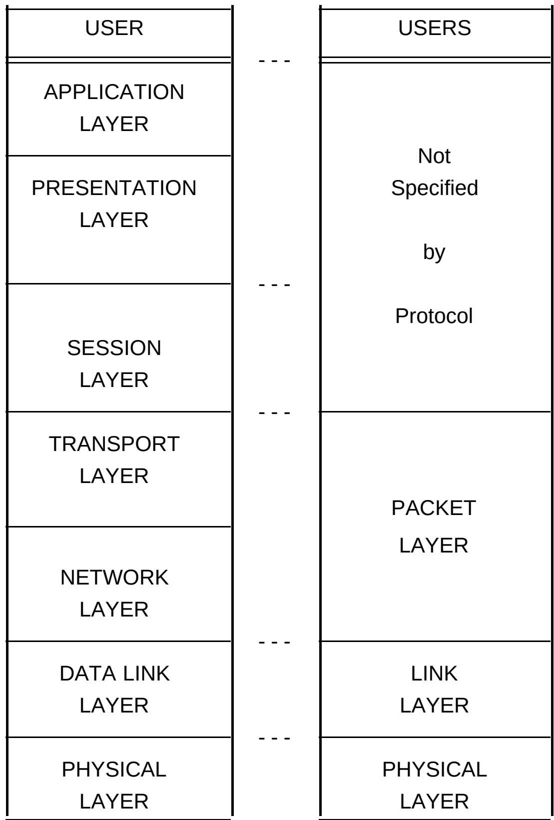
Basic elements of the Application and Presentation layers must be user defined under both protocols. Table 1-1 shows the relationship and similarity between the BX.25 and X.25 protocols.

The BX.25 protocol is used in the system to provide communications between the switch and the switch-related features. The BX.25 protocol is also used in the system to provide communications between the switch and the AUDIX and to provide communication between DCS switches.

Table 1-1. Packet-Switching Protocols



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 EIA-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 EIA-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

Network Services allows a group of switches (consisting of DEFINITY Generic 1, Generic 2, Generic 3i, System 75 and System 85, 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 the *System 75/85 AT&T Network and Data Services — Reference Manual 555-025-201*, for differences between this system and other AT&T systems.

Network Services Features

The following features are associated with Network Services:

1. AAR/ARS Partitioning
2. Automatic Alternate Routing
3. Automatic Circuit Assurance
4. Automatic Route Selection
5. Automatic Route Selection
6. DCS Alphanumeric Display for Terminals
7. DCS Attendant Control of Trunk Group Access
8. DCS Attendant Direct Trunk Group Selection
9. DCS Attendant Display
10. DCS Automatic Callback
11. DCS Automatic Circuit Assurance
12. DCS Busy Verification of Terminals and Trunks
13. DCS Call Forwarding All Calls
14. DCS Call Waiting
15. DCS Distinctive Ringing
16. DCS Leave Word Calling
17. DCS Multi-Appearance Conference/Transfer
18. DCS Trunk Group Busy/Warning Indication
19. Facility and Non-Facility Associated Signaling (G3i)
20. Facility Restriction Levels and Traveling Class Marks
21. Generalized Route Selection

22. Integrated Services Digital Network — Primary Rate Interface
23. Network Access — Private
24. Network Access — Public
25. Off-Premises Station
26. Restriction — Toll
27. Subnet Trunking
28. Ten-Digit to Seven-Digit Conversion
29. Time of Day Routing
30. Uniform Dial Plan

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 3i and Generic 1, however, are 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 nonhierarchical 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 1-4 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.

Within an ETN each location is identified by a unique private network office code. With G1.1, this private network office code is called an RNX. With G3i, this private network office code may be of the form RN, RNX, RX, XX, RNX, RXX, XXX, and RNXX, depending on administration (R = digits 2 through 9, N = digits 2 through 9, and X = digits 0 through 9). After accessing the ETN, the user simply dials the private network office code plus the desired extension number, for a total of seven digits.

Public network office codes (NXXs) are unique within an Area Code, whereas private network office codes are unique within an ETN. Private network office codes are assigned when the ETN is established. 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 controls call routing over an ETN is called Automatic Alternate Routing (AAR). AAR not only determines the route for a call, but, through the Facilities 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, Facilities Restriction Level, and Subnet Trunking are given in Chapter 2.

With G3i, AAR digit conversion is used to convert private network numbers to other private network numbers or public network numbers. This allows the system to steer some AAR calls to other switches in the private network or, by changing specific dialed digits to a public network number, eventually route some calls via ARS. Also, unauthorized private network calls can be routed to an attendant or receive intercept treatment. Details on AAR digit conversion can be found in the Automatic Alternate Routing (G3i) feature description in Chapter 2.

Distributed Communications System (DCS)

A DCS is a cluster of private communications switches (nodes) interconnected among several geographic locations. These switches can be either a DEFINITY Generic 1, DEFINITY Generic 2, DEFINITY Generic 3i, System 75, System 85, or DIMENSION® PBX. If all nodes are System 75s, DEFINITY Generic 1s, or DEFINITY Generic 3is the DCS can have as many as 64 nodes. This limit is removed if the DCS includes a System 85 or DEFINITY Generic 2. 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 or ISDN-PRI trunks (DEFINITY Generic 1 and Generic 3i) 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. DEFINITY Generic 3i and Generic 1 can serve as either an endpoint node or a regular (tandem) node. Figure 1-5 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.

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.

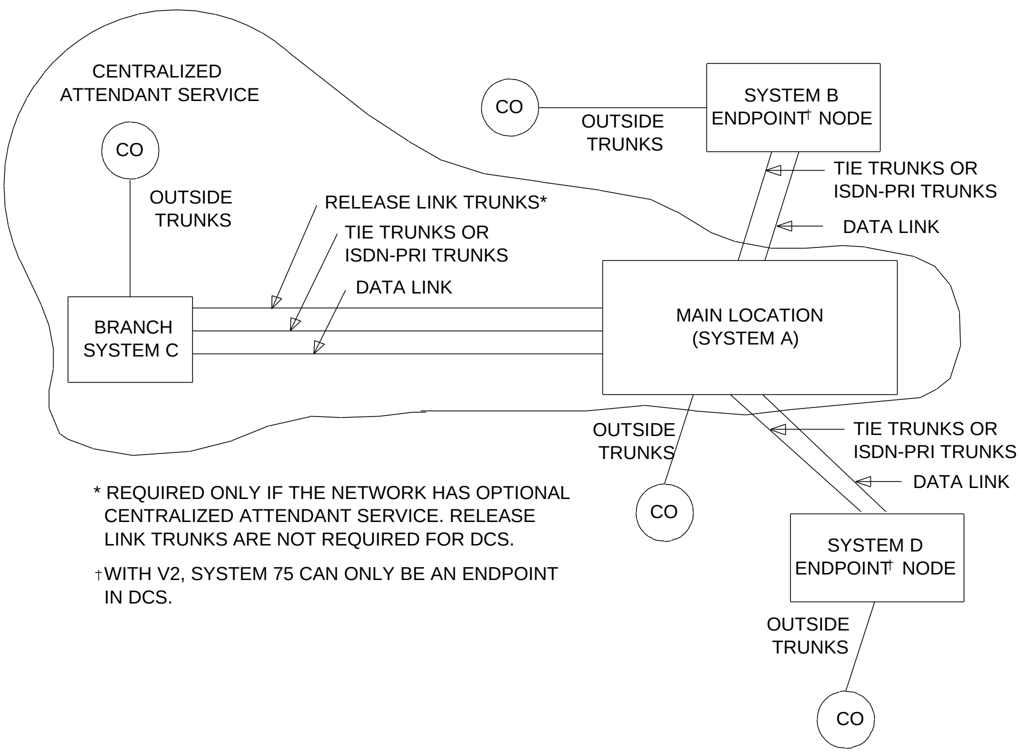


Figure 1-4. Typical Distributed Communications System

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 2:

- 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
- Enhanced DCS (described under Enhanced DCS)

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 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.

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 features.)

Certain feature capabilities are unique to a particular type of node (for example, a DEFINITY Generic 3i or 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. DEFINITY Generic 3i and Generic 1 can serve as the main location for CAS attendants. Centralized Attendant Service capabilities are given in detail in Chapter 2 of this manual.

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 1-6 shows a Main/Satellite/Tributary configuration. It can function independently or serve as an 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.

DEFINITY Generic 3i and 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.

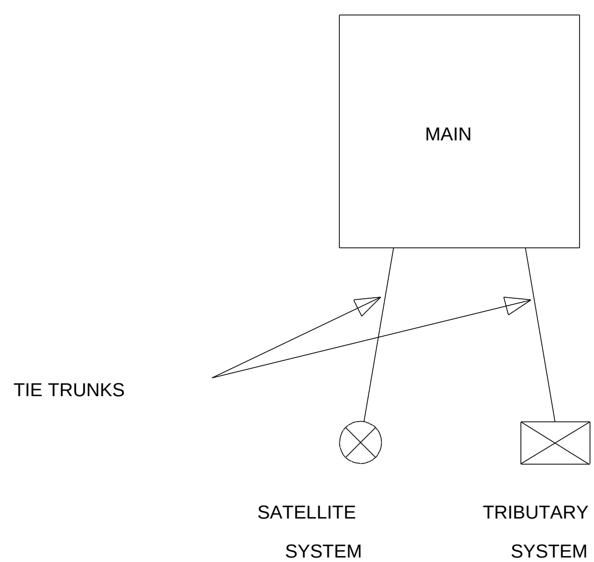


Figure 1-5. Main/Satellite/Tributary Configuration

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.
- 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 four-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, DEFINITY Generic 1, or DEFINITY Generic 3i and a System 85 or DEFINITY Generic 2. DS1 tie trunks also can be used with Release Link trunks for Centralized Attendant Service, and can be used with MEGACOM Telecommunications Service.
- FX — Provides a link with a CO other than the local CO.
- 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 telecommunications service, WATS, ACCUNET digital service, and SDDN, and calls destined for different inter-exchange carriers can be processed.
- Tie and Release Link — 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 ETN switch
  - An EPSCS or Common Control Switching Arrangement (CCSA) office
  - MEGACOM 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 one-burst ringing and will redirect according to the redirection criteria administered for internal calls.
- Incoming external tie trunk calls cause two-burst ringing and will redirect according to the redirection criteria administered for external calls.

The number of bursts for any type of call is administrable.

The Call Coverage feature is described in detail in Chapter 2 of this manual.

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 one-way incoming, one-way outgoing, or two-way. Whether the trunk group is available for incoming, outgoing, or two-way traffic is called direction. A two-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, DEFINITY Generic 1, or DEFINITY Generic 3i at the same time System 75, DEFINITY Generic 1, or DEFINITY Generic 3i 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 two-way trunk group compounds the possibility of glare and is, therefore, not recommended.

Each non-DCS outgoing and two-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 one 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 (touch-tone) 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 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 (system default; however, this may be administered as higher or lower on the trunk group form) after the last digit is dialed. 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 (DDC) or Uniform Call Distribution (UCD) 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

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 2. Functions are more fully described in the following documents:

- *DEFINITY® Communications System Generic 1 — Implementation, 555-204-654*
- *DEFINITY® Communications System Generic 3i — Implementation, 555-*

230-650

- *DEFINITY® Communications System Generic 1 and Generic 3 — System Management, 555-230-500*
- *DEFINITY® Communications System Generic 1 and Generic 3i — System Reports, 555-204-510*
- *DEFINITY® Communications System Generic 1 and Generic 3 — 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:

1. Administration Without Hardware
2. Customer-Provided Equipment (CPE) Alarm
3. Facility Test Calls
4. Move Agent From CMS
5. Recent Change History
6. Report Scheduler and System Printer
7. Security Violation Notification
8. Call Detail Recording
9. System Measurements
10. System Status Report

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 Manager I terminal (G1), the G3 Management Terminal, a Remote Administration terminal, or AT&T location. The Manager I terminal (G1) and the G3 Management Terminal 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 (must be optioned as a 513 BCT)
- 715 BCS (G3i)
- MS-DOS® compatible PC with 4410 emulation software (G3i)
- SAT PC (G3i)
- VT220

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 that 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.

Designated AT&T service locations have 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 terminal performs the same functions as the local administration terminal.

The 513 BCT, 515 BCT, 610 BCT, 615 MT BCT, 715 BCS, 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, 610 BCT, 615 MT, or 715 BCS, it must be connected to the system through a PDM, 7400A data module, 7400B data module, or Data Line circuit pack. If a 4425 terminal or 515 BCT is used as a remote terminal, a PDM, 7400A, or 7400B 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 — 5,000 feet using 24-AWG wire or 4,000 feet using 26-AWG wire
- Remote terminal to 7400A data module (G3i) — 6,000 feet using 24-AWG wire or 5,000 feet using 26-AWG wire
- Remote terminal to 7400B data module (G3i) — 6,000 feet using 24-AWG wire or 5,000 feet using 26-AWG wire. This is with the 7400B in the data-only mode. With DCP voice terminals that require phantom power, this distance is limited to 3,000 feet using 26-AWG wire.

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

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

Technical Service Center (TSC)

Allows system administration and maintenance from a remote location.

The TSC 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 TSC users. The TSC can also be used to perform maintenance routines.

During system access, the TSC automatically receives major and minor alarm notifications from the system. When an alarm is received, TSC 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

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 Property Management System Interface Check-In/Check-Out Controlled Restriction Housekeeping Status Message Waiting Notification Room Change/Room Swap Guest Information Input/Change

Telemarketing

The Telemarketing features of the system support industries such as airlines, travel agencies, and catalog s, that have a large number of similar incoming and/or outgoing calls. These features can provide balanced call distribution to a large group of voice terminals.

Telemarketing Features

The following features are associated with Telemarketing:

Abandoned Call Search Agent Call Handling

- — Stroke Counts (G3i)
- — Call Work Codes (G3i)
- — Forced Entry of Stroke Counts and Call Work Codes (G3i)

Automatic Call Distribution (ACD)
Basic Call Management System
Call Prompting (G3i)
Call Vectoring (G3i)
Intraflow and Interflow
Look Ahead Interflow (G3i)
Inbound Call Management (G3i)
Move Agent From CMS
Queue Status Indications
Service Observing

Feature Descriptions

2

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

  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

Description

Provides for the Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) services to be partitioned among as many as eight different groups of users within a single DEFINITY Generic 1 and Generic 3i. 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 interstate 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 translation tables 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

  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.

- 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 only will 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.

- CDR

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

- 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 (UDP)

  Since UDP calls expand the dialed digits into seven-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:

- With G1.1, different FNPA, HNPA, and RNX tables must be administered for each partitioned user group.
- With G3i, different Digit Analysis tables must be administered for each partitioned user group.
- A PGN must be assigned to each COR table. Up to eight PGNs can be used. If the Time of Day Routing feature is assigned, a Time of Day Plan Number is assigned to the COR instead of the PGN.

Hardware and Software Requirements

No additional hardware or software is required.

Abandoned Call Search

Description

Provides identification of abandoned calls for CO offices that do not provide timely disconnect supervision.

Before an incoming 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. Abandoned Call Search adds an overhead of up to one second to each call delivered to an agent.

To see if the calling party has abandoned the call, after the call has been abandoned by an announcement, 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.

Outside of the U.S., a flash of this duration may be responded to differently. Please see the Trunk Flash feature for more information.

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 is only suitable for older COs that do not provide timely answer supervision. Most COs provide timely disconnect supervision, and therefore do not require the Abandoned Call Search feature. Some older COs can take as long as two minutes to notify the PBX of a disconnect and, thus, require the PBX to determine, within one second, whether the call has been abandoned, prior to extending the call. 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 significantly reduce that probability.

Abandoned Call Search works only with ground-start analog trunks.

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 CMS are more accurate, because the CMS knows when a call is abandoned.

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.

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. Certain stored numbers can also be accessed by attendants.

List Types

Desired called numbers are stored in any of four 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 one-, two-, or three-digit 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. A user can have up to three Personal Number Lists with five or ten entries per list. G3i allows a maximum of 2,000 Personal Number Lists for the entire system, while G1.1 allows a maximum of 1,600 Personal Number Lists. 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 (five or ten 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 90 list entries (in multiples of five). An individual user can access up to three specific Group Number Lists, as set by the System Manager.

- System Number List

  Can have up to 90 entries (in multiples of five). 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

  Can have up to 1,000 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 one through nine, and zero. List entries for the Group Number Lists are numbered 11 through 99 and 00. List entries for the Enhanced Number List are numbered 000 through 999. This numbering scheme is used because the system expects either one, two, or three 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. The three lists may be any combination of the above as long as there is no more than one System and/or Enhanced 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 FRL checking. [FRLs are associated with the Automatic Route Selection and Automatic Alternate Routing 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. (This interval is administrable.) 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 5 to 25 seconds or until dial tone is detected, 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 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 DTMF data call can be made over a dial pulse trunk (for example, retrieving messages from AUDIX).

- 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 needed to delay outpulsing of the initial digits following access of an outgoing trunk if the system does not know when to start outpulsing over a trunk (for example, in Europe). 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 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 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, the list entry number, and the number to be stored (up to 24 digits), 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 24 digits), 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. 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 2,000 lists and a maximum of 10,000 entries are allowed for the G3i. (G1.1 allows a maximum of 1,600 lists and a maximum of 8,000 entries.) The 2,000 lists include a maximum of 2,000 Personal Number Lists, 100 Group Number Lists, a 7103A Group Number List, a System Number List, and an Enhanced Number List. (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 24 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:

- AUDIX Interface

  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.

  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
- — 7103A Group Number List

- Wait Delay Interval (5 to 25 seconds)
- Attendant Console Parameters

Hardware and Software Requirements

Additional 748B tone detectors (up to five per system) may be required if the special "wait" character is used frequently. Other tone detector circuit packs may be more appropriate; please see the System Description manual for more specifics.

Optional software is required for the enhanced Abbreviated Dialing list.

Administered Connections (G3i)

Automatically establishes an end-to-end connection between two access/data endpoints. The Administered Connection feature replaces the Permanent Switched Calls feature from previous releases of System 75 and DEFINITY Generic 1.1, and provides the following enhanced capabilities:

- Support of both permanent and scheduled connections
- Auto Restoration (preserving the active session) for connections routed over Software Defined Data Network (SDDN) trunks
- Administrable retry interval (from 1 to 60 minutes) per Administered Connection
- Administrable alarm strategy per Administered Connection
- Establishment/retry/auto restoration order based on administered priority

The status of an Administered Connection (disabled, connected, and so on) can be displayed by entering the **status administered-connection** command from the Manager I terminal (G1) or the G3 Management Terminal.

The endpoints which can be connected via the Administered Connection feature are either access endpoints or data endpoints. Access endpoints are non-signaling trunks and data endpoints are devices that connect the switch to data terminal/communication equipment. Throughout this section the term ‘‘endpoint’’ is used to mean either data endpoint or access endpoint.

Access Endpoints

An access endpoint is either a non-signaling channel on a DS1 interface or a non-signaling port on an Analog Tie Trunk circuit pack that is assigned a unique extension. Since an access endpoint is non-signaling, it will neither generate nor respond to signaling. As a result, an access endpoint cannot be used as a trunking facility (it cannot receive incoming calls or route outgoing calls). An access endpoint is used primarily to support devices, switches, or services that have a trunk interface but do not support signaling for the trunk. An access endpoint may be designated as the originating (local) endpoint or destination endpoint in an Administered Connection. The status of an access endpoint can be displayed by entering the **status access-endpoint** command from the Manager I Terminal (G1) or the G3 Management Terminal.

If a data call/connection between two access endpoints is set up from a voice station via the Transfer feature, the call can only be dropped (and the endpoints freed) by busying out either one of the Access Endpoints or a trunk over which the connection is routed (if one exists) from the Manager I Terminal (G1) or the G3 Management Terminal. This is required since neither of the endpoints can initiate a drop (access endpoints are non-signaling).

Typical Administered Connection Configurations

The Administered Connection feature allows a great amount of flexibility in assigning the destination address of the connection. As a result, many different configurations are possible with an Administered Connection. An Administered Connection can be established between two endpoints on the same switch; between two endpoints in the same private network, but on different switches; or between an endpoint on the controlling switch and another endpoint off the private network. In all configurations, the Administered Connection must be administered on the same switch as the originating endpoint.

If the two endpoints of the Administered Connection are on two different switches within a private network, normally, the connection will be routed through tie trunks (such as ISDN-PRI, DS1 or analog tie trunks) and possibly intermediate switches. However, if preferred, the connection can be routed through the public network.

The following are typical Administered Connection configurations and their application examples:

- A local data endpoint connects to a local or a remote access endpoint.

  One example of this is an MPDM connecting to a T1 Multiplexer via a DS0.

- A local access endpoint connects to a local/remote access endpoint.

  Two examples are a DS0 cross-connect and a four-wire leased line modem to a four-wire leased line modem connection via analog tie trunks.

- A local data endpoint connects to a local/remote data endpoint.

  One example is a connection between two 3270 data modules.

Establishment of Administered Connections

The originating switch will only attempt to establish an Administered Connection if the following conditions exist:

a. The Administered Connection is enabled,

b. The Administered Connection is due to be active (either a permanent Administered Connection or the time of day requirements are satisfied if a scheduled Administered Connection), and

c. The originating endpoint is in the in-service/idle state.

If the originating endpoint is not in-service/idle, no activity will take place for the Administered Connection until the endpoint transitions to the desired state. The destination address is used by the originating switch to route the connection to the desired endpoint. When two or more Administered Connections are to be established at the same time, they are established in priority order.

Administered Connection Establishment Retries

Administered Connection establishment attempts can fail for the following reasons:

- Resources are unavailable to route to the destination
- A required conversion resource is not available
- Access is denied. COR, FRL, BCC, or an attempt is made to route voice-band-data over SDDN trunks in the 4ESS™ switch network (or other public switch network)
- Incorrect destination address
- Destination endpoint is busy
- Other network or signaling failure

In the event of a failure, an error will be logged in the error log and an alarm will be generated, if it is warranted by the alarming strategy. The reason an Administered Connection has failed can be displayed by the System Manager via the **status administered-connection** command. This information is also contained in the error log.

As long as an Administered Connection is due to be active, continued attempts to establish an Administered Connection will be made by the originating switch unless the establishment attempt failed because of an administrative error (like a wrong number) or service blocking condition (like outgoing calls barred). Establishment attempts for Administered Connections that fail as a result of one of these conditions will resume when the problem is resolved (that is, Administered Connection administration has been changed). The frequency at which failed establishment attempts are retried is determined by the administered retry interval (1 to 60 minutes) of each Administered Connection. Retries will be made after the retry interval has elapsed regardless of the restorable attribute of the Administered Connection. If more than one Administered Connection is to be retried at the same time, they will be retried in priority order. When the customer changes the time of day on the switch, an attempt will be made to establish all Administered Connections in the ‘‘waiting for retry’’ state.

Dropping an Administered Connection

Once established, an Administered Connection will remain active until one of the following events occurs:

- The Administered Connection is changed, disabled, or removed. (See the Administration section for identification of which attributes, when changed, will result in the dropping of an active Administered Connection.)
- The time of day requirements of a scheduled Administered Connection are no longer satisfied.
- One of the endpoints initiates dropping the connection. This could be a result of a user initiated drop (in the case of a data endpoint), maintenance activity resulting from an endpoint failure, or the busying out of the

endpoint or handshake failure. If the endpoints involved in an Administered Connection are incompatible, the connection will successfully connect before the handshake failure occurs.

NOTE:
Administered Connections between access endpoints will remain connected even if the attached access equipment fails to handshake.

- An interruption (that is, facility failure) occurs in the path between the endpoints involved in the Administered Connection.

No action is taken if an Administered Connection drops because it was disabled/removed or is no longer due to be active. If an Administered Connection drops because of changed Administered Connection attributes, an immediate attempt will be made to establish the connection with the changed attributes if it is still due to be active. Existing entries in the error/alarm log are resolved if they no longer apply. If it can be determined that handshake failure resulted in the dropping of the connection, in the case of an Administered Connection involving at least one data endpoint, no action will be taken for that Administered Connection until the **change administered-connection** command has been executed.

Administered Connection Failure: Auto Restoration and Fast Retry

When an active (established) Administered Connection drops prematurely, either auto restoration or fast retry will be invoked. It can be determined whether or not auto restoration will be attempted for an active Administered Connection by observing the contents of the restorable field displayed on the Status Administered Connection screen.

Auto restoration will be attempted if the Administered Connection was optioned for auto restoration and the connection was routed over SDDN trunks. During restoration, connections are maintained between the switch and endpoint at both ends of the connection. In addition to allowing the active session to be maintained, this also provides a high level of security by prohibiting other connections from intervening in active sessions. The auto restoration feature cannot guarantee restoration within a certain time period, but successful restorations (involving remote endpoints on a G3i switch) must be completed before the expiration of the 60-second endpoint holdover timer utilized during restoration. If auto restoration is successful, the session that was active when the failure occurred might be maintained (no guarantee). If the session is maintained, the restoration is transparent to the user with the exception of a temporary disruption of service while the restoration is in progress. A successful restoration is reflected by the ‘‘restored’’ state on the Status Administered Connection screen. The restored status will be displayed, even if the destination endpoint was idle (that is, already dropped) when the restoration attempt arrived at the destination node. (Although the restoration was successful, the data session may not have been preserved.)

If the auto restoration function is not optioned or the Administered Connection is not routed over SDDN trunk(s), the switch will immediately attempt to reestablish the connection (fast retry). Fast retry will also be attempted if the originating endpoint initiated the drop. In the event of a fast retry, connections are not maintained on both ends. Fast retry will not be attempted for an Administered Connection which was last established via fast retry, unless the Administered Connection has been active for at least two minutes.

If the auto restoration or fast retry attempt fails to restore/reestablish the connection, the connection will be dropped and the Administered Connection will go into retry mode. Retry attempts will continue, at the administered retry interval, as long as the Administered Connection is due to be active.

Considerations

A maximum of 128 Administered Connections may be administered on a switch.

The maximum number of trunks (including access endpoints) that can be administered on a switch is 400.

Interactions

The following features and functions interact with the Administered Connection feature:

- Abbreviated Dialing

  Abbreviated dialing entries can be used in the destination address field. Entries must comply with the restrictions of the dial plan.

- AAR/ARS/Generalized Routing Selection (GRS)

  These features may be used in the routing of an Administered Connection.

- Busy Verification of Stations and Trunks

  This feature does not apply to access endpoints because access endpoints are used only for data.

- Class of Restriction

  A COR should be reserved for Administered Connection endpoints and SDDN trunks. This would restrict endpoints, not involved in Administered Connections, from connecting to SDDN trunks or endpoints involved in Administered Connections.

- Class of Service/Call Forwarding

  An Administered Connection endpoint should be assigned a Class of Service that will block call forwarding activation of the endpoint.

- Data Call Setup

  A Default Dialing destination should not be assigned to a data module that is used in an Administered Connection.

- Data Call Hotline

  A hotline destination should not be assigned to a data module that is used in an Administered Connection.

- Digital Multiplexed Interface (DMI)

  DMI endpoints can be used as the destination in an Administered Connection. DMI endpoints do not have associated extensions, so they cannot be used as the originator in an Administered Connection.

- Facility Test Calls

  The feature does not apply to access endpoints because an access endpoint acts as an endpoint rather than as a trunk.

- Hunting

  A hunt group extension is not allowed to be used as the origination extension of an Administered Connection.

- Modem Pooling

  If a conversion resource (pooled modem) is required in an Administered Connection, one will be inserted. If no conversion resource is available, the connection will be dropped.

- Non-Facility Associated Signaling (NFAS) and D-Channel Backup

  Auto Restoration for Administered Connections, initially routed over an NFAS facility, may fail if the only backup route is over the facility on which the backup D-Channel is administered, since the backup D-channel may not come into service in time to handle the restoration attempt.

- Set Time Command

  When the System Manager changes the system time via the **set time** command, all scheduled Administered Connections are examined. If the system time change causes an active Administered Connection to be outside its scheduled period, the Administered Connection will be dropped. If the time change causes an inactive Administered Connection to now be within its scheduled period, the switch will attempt to establish the Administered Connection.

  Also, if any Administered Connection (scheduled or continuous) is in the retry mode and the system time changes, the switch will attempt to establish the Administered Connection immediately.

- CDR

  For an Administered Connection that uses a trunk which has CDR enabled, the origination extension of the Administered Connection will be used as the originator of the call.

  CDR is not available for access endpoints.

- System Measurements

Access endpoints are not measured. All other trunks in an Administered Connection are measured as usual.

- Terminal Dialing

  It is recommended that the terminal dialing capability be turned off for data modules involved in an Administered Connection.

  NOTE:
  This will stop call processing related messages (INCOMING CALL, ...) from being displayed on the terminal.

- Trunk Groups

  In order for auto restoration to be invoked, an Administered Connection must be routed over SDDN trunks. Since a successful restoration depends on there being an SDDN path over which to route the restoration attempt, some SDDN trunks should be kept idle to be used in the event of failure for restoration. SDDN trunk group usage should be restricted to Alternating Current (AC) related traffic.

Administration

Each Administered Connection is administered by the System Manager. The following items require administration:

- Endpoints

  If an Administered Connection involves local endpoints, the endpoints must be administered before the Administered Connection using those endpoints can be administered. An endpoint cannot be removed if it is involved in a locally administered Administered Connection, or if it is currently involved in an active Administered Connection. If the user desires to change any of the translation data associated with an endpoint (except for the Name field which may be changed at any time) that is involved in an active Administered Connection, the Administered Connection must first be disabled or removed.

- Administered Connection

  An Administered Connection must be administered on the same switch as the originating endpoint. The System Manager may change the attributes of an Administered Connection at any time, but not all changes take effect immediately. These attributes are as follows (included in each description is a statement as to whether or not changes take effect immediately):

  — **Originating Address** — The address of the originating endpoint is its local extension on the originating switch. When this attribute is changed for an active Administered Connection, the connection will be dropped and reestablished using the new originating address.

  — **Destination Address** — The destination address is used to route the Administered Connection to the desired destination. When this attribute is changed for an active Administered Connection, the connection will be dropped and reestablished using the new destination address.

— **Enable** — The Enable field allows the System Manager to specify whether the system should attempt to establish the connection when it is due to be active. Answering ‘‘yes’’ to the enable option indicates that the system should be established when the Administered Connection is due to be active. Answering ‘‘no’’ indicates that the System Manager does not want the Administered Connection to be considered for activation at this time (that is, held for future use). A disabled Administer Connection is displayed along with the other Administered Connections administered locally in response to the **list administered-connection** command. A disabled Administered Connection may be enabled at any time. Since Administered Connection administration is done on the originating switch, disabling an Administered Connection can only be done on the originating switch.

  If an Administered Connection is currently active, answering ‘‘no’’ causes the Administered Connection to be dropped immediately. If an Administered Connection is disabled, answering ‘‘yes’’ will cause the originating switch to attempt to establish the connection immediately if the Administered Connection is due to be active. *The disabling and enabling of an Administered Connection after an attribute of the Administered Connection has been changed guarantees that the change will take effect immediately.*

— **Name** — A one through 15 character long, optional `Name` field is provided to allow for additional identification information. Changing this field has no effect on the Administered Connection connection.

— **Authorized Time of Day** — An Administered Connection may be continuous (permanent) or scheduled. Scheduled Administered Connections are described by indicating the days of the week, start time, and the duration for which the Administered Connection is to be active. The modification of any of the attributes associated with the authorized time of day will not affect the current status of an Administered Connection unless the change results in activating or deactivating an Administered Connection.

— **Priority** — The System Manager can specify the priority of a given Administered Connection. This priority is used to determine the order in which Administered Connections are established if two or more Administered Connections are due to be active at the same time. The `Priority` field allows the user to specify a number between one and eight (with one being the highest and eight being the lowest). Changes to the priority attribute have no effect on an active Administered Connection.

— **Auto Restoration** — The System Manager may specify whether an attempt should be made to restore an Administered Connection, via the auto restoration feature, if the connection is dropped due to failure and the connection was routed over SDDN trunk(s).

Reestablishment (retry) of dropped connections is attempted regardless of the value specified in this field. This field has no effect on Administered Connections routed over non-SDDN trunks. The System Manager must disable and enable an active Administered Connection to have changes to this attribute take effect.

— **Retry Interval** — The System Manager must specify a retry interval of 1 to 60 minutes. The default is two minutes. This interval is the number of minutes waited before a retry is attempted. When this field is changed, the new interval will be used for the next retry. An Administered Connection, which is in retry mode when this field is changed, will retry after the old interval has elapsed and then use the new interval for the next retry time. Twenty-three of the Administered Connections will be restored within the required time. The remaining Administered Connections will be restored, but after the time limit.

— **Alarm Type** — An alarm type of none, warning, minor or major must be chosen. The default will be warning. ''None'' indicates that no alarms will be generated on establishment or restoration failure. A choice of ''warning'' will cause alarms to be generated and logged in the alarm and error log. A minor or major indication will also cause alarms to be generated and logged in the alarm and error log and forwarded to an Operations Support System such as INADS if OSS is administered.

Changing this field to ''none'' will cause an existing alarm to be cleared. Changing the field to one of the other values will cause the upgrading or the downgrading of an existing alarm.

— **Alarm Threshold** — The Alarm Threshold field indicates the number of consecutive failures (1 to 10) that must occur before an Administered Connection alarm is generated. Entering 1 in this field will cause alarms to be generated immediately upon failure to establish or reestablish an Administered Connection.

Changes to this field take effect immediately. A comparison of the new value and the current retry count will be made to determine if an alarm should be generated or possibly cleared due to the change.

- Access Endpoint

  The access endpoint has the following attributes which must be administered:

  — **Extension** — This is a unique one- to five-digit identifier, consistent with the current dial plan, by which this access endpoint is addressed.

  — **Port** — This is the port address of the DS1 or analog tie trunk port. A DS1 trunk can be used regardless of the signaling mode of the DS1 circuit pack.

  — **Name** — This can be any alphanumeric string (up to ten characters) representing a name that is useful to the customer.

- **Communication Type** — A communication type of 64K data, 56K data, or voice-band data must be assigned. An access endpoint on an analog tie trunk port is restricted to a communication type of voice-band data. In addition, a communication type of 64K is not allowed for access endpoints on DS1 circuit packs administered for robbed-bit signaling.
- **COR** — A Class of Restriction may be administered for each access endpoint.
- **COS** — A Class of Service may be administered for each access endpoint. Class of Service administration should be used to block call forwarding activation of an endpoint.

Hardware and Software Requirements

Hardware requirements vary depending on the type of Administered Connection desired. The following hardware may be required for Administered Connections:

- Access Endpoint Circuit Packs — TN767 DS1 Interface circuit pack (TN464B/C/D support A-law), TN760B Analog Tie Trunk (TN760D supports A-law).
- Data Endpoint Circuit Packs — TN726 Data Line or TN754 Digital Line (TN413, TN754B support A-law).
- Data Modules — 700A/700D PDM or MPDM, 700B/700C/700E TDM or MTDM, 7400D series voice terminal with DTDM or 7400B Data Module, PC/PBX, 510D, 515BCT.
- Trunk Circuit Packs — TN767 DS1 Interface circuit pack (TN464B/C/D support A-law), TN760 Analog Tie Trunk (TN760D supports A-law).
- TN758 Pooled Modem circuit pack.

No additional software is required.

Administration Without Hardware (G3i)

Description

Provides the ability to administer station forms without specifying a port location. Stations administered as such will not cause alarms or errors to be generated when the station is translated but not yet installed. These station types are referred to as ‘‘phantom’’ stations. The Administration Without Hardware (AWOH) feature supports the following applications:

- Ability to administer station forms without specifying a port location.
- Ability by use of a phantom extension to provide call coverage (including AUDIX) for users who do not have stations physically located on the switch.
- Ability to use phantom extensions for ACD Dialed Number Identification Service (DNIS). This application allows a phantom extension to be administered on the switch for each call type that needs to be identified to ACD agents. The phantom extension is either ‘‘Call Forwarded’’ (via an attendant console) to an ACD split, or its coverage path is defined to include the ACD split. The `Name` field that is administered for the phantom extension will identify to the ACD agent which service the caller is attempting to reach, allowing the agent to properly address the caller.
- Ability to store station templates that can later be used with the **duplicate station** command when implementing many station forms of the same type in the system.

Considerations

The primary use of the AWOH feature is to streamline system initializations, major additions, and rearrangement/changes by allowing voice terminal translations to be entered before the actual ports are assigned. Port assignments can be done at a later time, as required.

The use of this feature is limited to analog, DCP (7400D series of terminals), and hybrid terminal types.

Interactions

None.

Administration

AWOH is administered on a per-voice terminal basis by the System Manager. Normal station administration is required with the exception of entering an X in the `port` field to indicate that there is no hardware associated with the station.

Hardware and Software Requirements

No additional hardware or software is required.

Agent Call Handling

Description

Provides 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 (zip tone)
  - Manual Answer
- ACD Work Modes
  - Auxiliary Work Mode
  - After Call Work
  - Auto-In
  - Manual-In
- Agent Request for Supervisor Assistance
- ACD Call Disconnecting (Release button)

NOTE:
All of the agent capabilities listed above are also supported through the CallVisor ASAI. For information on CallVisor ASAI, consult the Adjunct/Switch Application Interface feature description presented in this chapter.

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 will be required to enter a log-in identification number when logging in if the hunt group is measured via CMS. If the hunt group is not measured, entry of a login ID is optional (probably only required for security).

To log in, an agent must go off-hook and dial the log-in feature access code, followed by the two-digit 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 and the agent hears confirmation tone. At the same time, the system sends two messages to the CMS or BCMS (if it is a measured split): a message that the agent has logged in (including the identification number for CMS) 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 (that is, the agent dials the number of a split to which he or she is not assigned).
- 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 or BCMS and logs out, a message is sent to the BCMS/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 than 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. If the agent is in the Automatic Answer mode and using a handset, he or she can log out of the split by going on-hook. If the agent is in the Automatic Answer mode and using a headset, he or she can log out of the split by turning off the headset. If the agent has a multi-appearance voice terminal, he or she can log out of the split by depressing the Logout button.

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) at all times. While off-hook, the agent hears zip tone through the handset or speakerphone.

NOTE:
Automatic Answer applies to all calls terminating to the agent's set. If the agent will receive direct calls, he or she should always activate Call Forwarding or Send All Calls when leaving his or her position and make himself or herself unavailable for ACD calls (by logging out or entering AUX work mode) so calls will not terminate to an unmanned station).

Manual Answer

If the agent is assigned Manual Answer, the agent hears ringing, and then goes off-hook to answer the incoming call. If the agent does not go off-hook, the call will continue ringing. 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 does not have a call on hold, the mode change is immediate. However, if an agent tries to change modes while he or she is active on a call or has a call on hold, the mode is not changed until the agent is disconnected from 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

- Auto-In
- Manual-In
- After Call Work

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 (and the agent is not in the most idle agent [MIA] queue), but BCMS/CMS tracking of the agent continues.

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 BCMS/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 AUX lamp lights steadily and the BCMS/CMS is informed of the agent's state change. 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.

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 may be available for ACD calls to other splits that the agent is logged into depending on the agent's state in those splits and the agent is still available for non-ACD calls. The BCMS/CMS is notified whenever an agent in the Auxiliary Work 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 lamp lights steadily and 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 (described later) for that split, and is not available for any ACD calls. The agent must then manually reenter 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 lamp lights steadily and 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.

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, although the agent is placed in the MIA queue upon entering this state and advances in queue until entering the AUX state or until he or she becomes available and reaches the front of the MIA queue and receives a call.

When an agent is in the Manual-In mode 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 BCMS/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 BCMS/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. (The agent is automatically placed in the AUX work mode for any other split(s).) However, the agent is still available for non-ACD calls. The BCMS/CMS is notified whenever an agent in the ACW mode receives an incoming non-ACD call or makes an outgoing call.

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 by putting the call on hold and dialing the Assist feature access code, followed by the split group number. Assist generates three burst ring at the supervisor's station. 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 BCMS/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 **Start** before pressing
  Assist. 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 **Assist**. This automatically places a call to the split supervisor. The BCMS/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 BCMS/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 BCMS/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 four ways:

- The agent can press **Release** (if provided). Dial tone is not heard after **Release** is pressed.

- The agent can press **Drop** (if provided). The agent hears dial tone after pressing
  Drop and is not available for calls.
- The call can be dropped by the calling party.
- The agent can go on-hook (hang up).

  Agents using Automatic Answer are logged out of all splits when they disconnect from an ACD call by going on hook (hanging up). The preferred method of operation is to use **Release** (if provided).

Stroke Counts (G3i)

Stroke Counts provide ACD agents with the ability to record up to nine customer-defined events on a per-call basis when the adjunct CMS is active. It also provides a tenth event count to track audio difficulty. Stroke Count ‘‘0’’ is reserved for Audio Difficulty and the other nine Stroke Counts are customer definable.

Stroke counts are reported to the CMS in real time. The switch does not store any Stroke Count information. Therefore, Stroke Counts are only useful when CMS is connected and ACD splits are administered to be measured by CMS.

ACD Stroke Counts allow agents to record relevant events on a per-call basis. A Stroke Count represents an event that the customer wants to measure. For example, a Stroke Count may be used to keep track of the number of inquiries about a specific item. Each time an agent receives an inquiry on a specific item, he or she can enter the Stroke Count (one through nine) assigned to that item.

Stroke count "0" is used to indicate audio difficulty. This count refers to calls with poor transmission qualities experienced by ACD agents. Audio transmission quality is not improved by pressing the Audio Difficulty Stroke Count button, the specific trunk used for the call is recorded.

Ten button types may be assigned for use with Stroke Counts: 0-Stroke, 1-Stroke, 2-Stroke, 3-Stroke, and so on.

An ACD agent enters a Stroke Count by pressing a Stroke Count button while off-hook. The system then validates that the activating agent is either active on an ACD call or in the ACW work mode for an ACD split. If these conditions are met, the feature lamp lights steadily for two seconds to indicate successful activation and a message is sent to the CMS containing the Stroke Count. If the preceding conditions are not met, the feature lamp will flutter to indicate unsuccessful activation and no message will be sent to the CMS. The lamp goes dark after two seconds.

Call Work Codes (G3i)

Call Work Codes allow ACD agents to enter up to 16 digits for an ACD call to record the occurrence of customer-defined events (such as account codes, social security numbers, or phone numbers). The switch does not store any Call Work Code information. Call Work Codes are sent to an adjunct CMS for storage. Release 3 of the CMS is required to record Call Work Code information.

Data will be sent to the CMS only for the splits that are measured by the CMS and only when the link to the CMS is up. The activating agent must be on an ACD call or in the ACW mode after disconnecting from a call while in the Manual-In mode. Activation in any other work mode is denied.

To enter a Call Work Code, the ACD agent presses the Call Work Code button while off-hook. The system then validates that the activating agent is either active on an ACD call, in the ACW mode after disconnecting from a call while in the manual-in mode, or the ACW mode is pending for the call. If these conditions are met, the associated lamp lights steadily and a `C:` prompt appears on the display. This indicates that the feature is ready for digit collection. After receiving this visual indication, the agent can enter the desired digits. (The agent must wait for the visual indication before entering the digits, or the calling party will hear the touch-tone digits being dialed). The agent may enter up to 16 digits. Any digits beyond 16 will be ignored by the system and will not appear on the display. The agent then presses **#** to send the Call Work Code entry to the CMS. The Call Work Code lamp will go dark and the display will return to normal mode. If an error is made while entering digits, the user may press ***** to erase all previous digits and begin entering digits again.

If any button is pressed at the agent's voice terminal, or the agent hangs up during digit collection, the Call Work Code entry is aborted and no message is sent to the CMS. Also, the Call Work Code lamp is extinguished and the display is cleared.

Although display-equipped voice terminals are recommended, this Call Work Code can also be entered from voice terminals without displays. This feature requires display-equipped voice terminals (for example, CallMaster).

The Call Work Codes function may be used by as many as 40 agents simultaneously. If 40 agents are simultaneously using this function, and another agent attempts to enter a call work code, this agent will receive a display message telling the agent to try again later.

Forced Entry of Stroke Counts and Call Work Codes (G3i)

An agent is always allowed to enter a Stroke Count and/or Call Work Code for an ACD call (as long as the agent is on an ACD call or in the ACW mode after disconnecting from a call while in the manual-in mode). Activation in any other work mode is denied. However, each split can be administered so that agents in that split are forced to complete a Stroke Count and/or a Call Work Code entry for every call answered in the Manual-In mode. For details on Forced Entry of Stroke Counts and Call Work Codes, see the Automatic Call Distribution feature description elsewhere in this chapter.

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. (An agent active on a non-ACD call will not receive an ACD call.)

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 on external CMS only. To track individual agent data, the login ID must be at least one digit. The system checks the number of digits in the identification number and verifies that it is not already active. It does not check to see if the identification number is a valid number although the external CMS utilizes the login 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.
- Calls 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.)

NOTE:
The agent can, however, log out.

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.

On direct calls to ACD agents with Auto-Answer, incoming notification will be sent instead of ringing the station once. This includes attendants.

Any calls to non-ACD Auto-Answer agents will be announced by Incoming Call ID tone. Ringing will no longer be heard.

Calls for CALLMASTER digital voice terminals and attendant stations will be announced by double tones. The user hears part of the first tone and all of the second tone. The tones that are doubled are Zip (Auto-Answer ACD agent calls) and Incoming Call ID (all other Auto-Answer calls.)

Agents should not be used for hunt group calls and ACD split calls simultaneously. Otherwise, all of the calls from one split (either ACD or hunt group) will be answered first. For example, if the ACD calls are answered first, none of the hunt group calls will be answered until all of the ACD calls are answered.

The oldest call waiting termination is only supported for agents who are servicing ACD calls only.

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 BCMS/CMS may or may not be administered. The Agent Call Handling features function the same whether it is administered or not. If the BCMS/CMS is not administered, the system does not send information to the BCMS/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 BCMS/CMS. Therefore, the BCMS/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 (for placing calls)

The following items are optional:

- Manual-In button
- Auto-In button
- Auxiliary Work button
- After Call Work button
- Assist button
- Release button (required with CallMaster voice terminal)
- Stroke Count buttons (G3i)
- Call Work Code buttons (G3i)

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

With G3i, each split must be assigned Forced Entry of Stroke Counts and Call Work Codes if agents in that split are to be required to enter these items.

Hardware and Software Requirements

No additional hardware is required, although CallMaster voice terminals are recommended for ACD agents. ACD software is required.

Alphanumeric Dialing (G3i)

Alphanumeric Dialing enhances Data Terminal Dialing by allowing data terminal users to place a data call by entering an alphanumeric name. This capability makes Data Terminal Dialing both convenient and user-friendly. Instead of dialing a long string of numbers, the users can enter a simple alphanumeric name.

When an alphanumeric name is entered from a user's terminal, the system's call processing software converts the name to a sequence of digits by searching through an administered Alphanumeric Dialing Table. The system then dials those digits just as if the user had entered the digits. If the entered name is not found in the Alphanumeric Dialing Table, the call attempt is denied and the user receives either an `Invalid Address` message (for DCP) or a `Wrong Address` message (for ISDN-BRI).

The Alphanumeric Dialing Table can be assigned as many as 200 alphanumeric names and matching digit strings. Each alphanumeric name can have up to eight characters. The first character in each alphanumeric name must be a letter of the alphabet from A to Z (case insensitive). The remaining characters (up to seven) may be any alphanumeric character. The mapped digit string may contain up to 24 dialing characters.

Since data terminals access the switch via DCP or ISDN-BRI data modules, the procedures for using Default Dialing vary. For data terminals using DCP, users type the alphanumeric name and enter a carriage return at the `DIAL:` prompt. For data terminals using ISDN-BRI, users type "d", enter a space, type the alphanumeric string, and enter a carriage return at the `CMD:` prompt.

More than one alphanumeric name can refer to the same digit string. Also, multiple names (mixed with number strings) can be used to dial a number. For example, a company may administer the Alphanumeric Dialing Table to convert the alphanumeric name ‘‘home’’ to the digit string for the area code and office code of the home office. In this example, a data terminal user with a DCP data module could access extension 3797 at the home office by typing `home 3797` and entering a carriage return at the `DIAL:` prompt. A data terminal user with an ISDN-BRI module could access extension 3797 at the home office by typing d `home 3797` and entering a carriage return at the `CMD:` prompt.

Considerations

Alphanumeric Dialing allows a data terminal user to place a data call by entering an alphanumeric name. This makes Data Terminal Dialing both convenient and user-friendly. Instead of dialing a long string of numbers, the user can enter a simple alphanumeric name.

Interactions

The following feature interacts with the Alphanumeric Dialing feature:

- Data Call Setup

  Alphanumeric Dialing enhances Data Terminal Dialing by allowing a data terminal user to place a data call by entering an alphanumeric name.

Administration

Alphanumeric Dialing is administered by the System Manager. In addition to those items listed in the Data Call Setup feature description, elsewhere in this chapter, alphanumeric names and associated digit strings must be assigned in the Alphanumeric Dialing Table.

Hardware and Software Requirements

No additional hardware or software is required.

Answer Detection

Description

Improves the accuracy of the call duration in CDR call detail records by detecting the state of outgoing trunk calls that do not receive Network Answer Supervision.

Since Network Answer Supervision may or may not be sent back to G3i for a normal outgoing trunk call, G3i relies upon the Far End Answer Supervision timer once an outgoing call is placed. The Far End Answer Supervision timer is an internal, administerable timer that estimates when the far end should have answered the call. The interval for the Far End Answer Supervision timer is specified on the trunk group form. As soon as the specified Far End Answer Supervision time interval is reached, CDR generates a call detail record. However, if the call terminates before the timeout interval is reached, CDR does not generate a call detail record. Answer Detection allows the system administrator to set the answer supervision timeout to larger values while generating CDR call detail reports on short duration calls. By providing more accurate timing of call duration, Answer Detection also improves call classification.

Considerations

The Answer Detection feature does not accurately detect all types of tones especially in countries whose tone schemes are not similar to the U.S. For a normal answered call, the call will usually be correctly classified as "answer". However, some calls may be misclassified as "Fast Busy" when they are actually "answer". Miscellaneous tones, such as the PBX tones (that is, confirmation) will be classified as "answer". In addition, loud background noise may activate Answer Detection, causing the call to be classified as "answer", even if the call is not connected. If Answer Detection is accidentally activated, but the call is not connected, the call is recorded in CDR records on the PBX. If Answer Detection is accidentally activated and the call is connected, the caller is billed erroneously for the time that elapses between Answer Detection activating and the called party answering.

Interactions

The following features interact with the Answer Detection feature:

- CallVisor ASAI

  Answer Detection competes with CallVisor ASAI switch-classified calls for ports on the TN744.

- Call Prompting

  Answer Detection competes with Call Prompting for ports on the TN744.

Hardware and Software Requirements

Requires a TN744 Call Classifier circuit pack.

Attendant Auto-Manual Splitting/Don't Split

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 **Start**, 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 three-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

The CAS attendant must use Don't Split to extend calls.

Administration

There must be one Don't Split button assigned per attendant.

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 or administered for a single-line station 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 two-burst tone. The two-burst tone is heard only by the called voice terminal user. (The number of bursts is administrable.)

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 Time-out or Timed Reminder on Hold) expires. If the interval expires, the call returns to the console. 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 two-burst tone and knows a call is waiting. The voice terminal user at extension 123 can then terminate the call in progress, or place the call in progress on hold, 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 (two to nine 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 Time-out 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.

- DDC and 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 1,020 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. Attendant Call Waiting is assigned to single-line voice terminals on a per-terminal basis. The call 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. 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 a basic 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. 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 six (basic console) or 12 (enhanced console) different trunk groups.

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 (UDP)

  Activating Attendant Control of Trunk Group Access removes the controlled trunk group(s) from preferences. Deactivating the feature enables the (UDP) 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 2,000 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 UDP 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. Twelve additional Group Select buttons 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 four-digit extension numbers, the Group Select buttons could be labeled 2400, 2500, 2800, and so on. Likewise, a three-digit system could have these buttons labeled as 100, 200, 300, and so on. A two-digit system would have a 0 Group Select number. A five-digit system, for example, could have group select buttons labeled 28400, 28500, 28600, and so on.

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 2,000 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

Need and introductory sentence for this list:

- 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.

- 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 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. In addition, 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 a basic 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, FX, or WATS. Pressing the button selects an idle trunk in the desired group.

Each attendant console has 12 designated Trunk Group Select buttons. 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.

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.

Attendants may select a display message language. The language choices are English (default), French, Italian, or Spanish. Please see the description under Multi-Language Displays for more information.

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

  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 ISDN-PRI feature, additional calling party information is provided. See the ISDN-PRI feature description, elsewhere in this chapter, for details.

- Called Party Identification

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

- Internal COR

  All system users have a COR to define their calling privileges. The COR is a two-digit number followed immediately by a hyphen and a four-character identifier. 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 Progress Feedback including "ringing", "busy", and "call waiting" will be presented if a G3i-Global switch is used.

- 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 — 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 — 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 — 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.

  ld — DID LDN Call — indicates that the incoming call is a Listed Directory Number (LDN) call on a DID trunk.

  n — night service — indicates that night service is on and calls will go to the night service station.

  na — night service for both attendants — indicates that both the main attendant and attendant 2 are in night service and position busy when a call comes in. Attendant 2 must press position busy in order to take the

call.

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

qf — Queue Full — indicates that the Attendant is in night service and that emergency calls are being routed to a different station.

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."

c — Cover All — indicates that the called voice terminal user has had all calls redirected to coverage.

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

Internal call originated by the attendant:

a=3602

then

a= TOM BROWN 3062

or

a= EXT 3602 3602

Outgoing trunk call originated by the attendant:

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:

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.

Call Progress Feedback including "ringing", "busy", and "call waiting" will be presented if a G3i-Global switch is used.

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. The Return Call button can be used to dial prefixed extensions, because the 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 Intrusion (Call Offer)

This is a new feature for G3. The Attendant Intrusion (Call Offer) feature enables an attendant to enter an existing call on either a digital station or analog station to offer a new call or message to the called party. Upon intrusion, tone may be applied if administered. Upon the attendant's release from the intruded call, the calling party's call is held by the Call Waiting feature for the intruded- on party or rings on an available digital line appearance. The intruded-on party must be on an analog station in order for the Call Waiting feature to be activated.

Considerations

Only attendant users can activate this feature.

Interactions

The following are interactions with other features:

- If a station is on a conference call with administered maximum number of conferees, the attempt to intrude on the station will be denied.
- If there is one call already Call Waiting for the intruded-on party, the source (split from attendant) party will not be able to wait for the intruded party using Call Waiting.
- If a call is established with Data Privacy activated, the attempt to intrude on the call will be denied.
- If a station in a call is administered with Data Restriction, the attempt to intrude on the call will be denied.
- If an attendant attempt to intrude on a call on a station which is a forward-to point of another station, the intrusion will be denied.
- If an attendant attempts to intrude on a busy station where the station is talking to another attendant, the intrusion will be denied.

Administration

An Attendant Intrusion button is required at the attendant console. There can be only one button per console. If intrusion tone is desired, this must be administered.

Hardware/Software Requirements

No special hardware is required for this feature in a stand alone system.

Attendant Override of Diversion Features

This is a new feature for G3. Diversion Override enables an attendant to bypass any diversion features invoked by and/or associated with a dialed extension. Diversion Features are any feature which when activated causes the call to alert at a point different from the dialed station. Specifically, the diversion features are Send All Calls, Call Coverage and Call Forward including cases in which the call alerts at the dialed station and is later transferred (as in the case of Busy Don't Answer).

Depressing the OVERRIDE button before originating a call, invokes the feature and causes the OVERRIDE lamp to light.

If an invalid extension is dialed the override button lamp gives denial. If the Release or Forced Release button is depressed during dialing, the Diversion Override feature is deactivated. A second depression of the button deactivates the feature and extinguishes the lamp. The feature is also deactivated when the call to the dialed extension is terminated, or if a trunk access code is dialed.

Activation or Deactivation of the Override feature while dialing is permitted.

Considerations

This feature together with Attendant Intrusion (Call Offer) can be used to get an emergency or urgent call through to a station user.

Interactions

Calls directed to a station on which diversion features are active, which would normally cause the call to terminate on another extension, are treated as though the diversion features are not active.

Administration

Only one OVERRIDE button is allowed per console.

Hardware/Software Requirements

No special hardware requirements exist for this feature.

Attendant Priority Queue

This is a new feature for G3. The Attendant Priority Queue feature is used to handle incoming calls to the attendant group or to an individual attendant when the call cannot be immediately terminated to an attendant. Such calls are placed in the Attendant Priority Queue in an ordered fashion based on a Priority Queue level and timestamp associated with the call. The calling party hears ringback until the call is answered by an attendant.

Twelve different categories of incoming attendant calls are defined which each have a designated Attendant Priority Queue level. Although each of these categories is given a default level, the customer may specify a Priority Queue level for any or all categories via a system administration function.

The category for "Emergency Call to the Attendant Group" is always defaulted to the highest queue level priority, with all other categories defaulted to a lower priority.

A call placed in the Attendant Priority Queue is associated with one of the following twelve Priority Queue categories:

- Emergency Call to the Attendant Group — Originated by a station user who has dialed the customer-administerable emergency access code.
- Assistance Call to the Attendant Group — Originated by a station user who has dialed the attendant group access code, or originated by a station which has the Manual Origination feature activated.
- Attendant Group Call over a CO/FX/WATS Trunk — An incoming trunk call directed to an attendant group (does not include trunk calls returned to the attendant group after a timeout or some type of deferred attendant recall).
- Attendant Group Call over a DID Trunk — Same as previous category except for a DID trunk.
- Attendant Group Call over a Tie Trunk — Same as previous category except for a tie trunk (dial-repeating or direct types).
- Redirected DID or Redirected ACD Call — A DID or ACD call which times out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable and is rerouted back to the attendant group.
- Attendant Redirected Call — A call assigned to terminate at an individual attendant, but subsequently rerouted to the attendant group because the individual attendant console is busy on another call.
- Attendant Return Call — A call returned to the attendant after a timeout of an extended station or trunk call. Such a call is intended to return to the attendant who extended it. However, if that console is busy on another call, the extended call is returned to the attendant group.This category is a type of Attendant Redirected Call with its own identity to allow assignment of a Attendant Priority Queue level.

- Serial Call — Originated by the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a station, and then the station user goes on-hook. If the attendant who extended the serial call is busy on another call, the serial call is redirected to the attendant group.
- Individual Attendant Access Call — Originated by a station user, incoming trunk call, or a system feature which uses the Individual Attendant Access (IAA) extension to direct a call to a specific attendant. If the individual attendant is busy on another call, the Individual Attendant Access call is queued until the individual console is idle. Then the queued IAA call is terminated to the individual attendant console.
- Interposition Call — Originated by one attendant who directs a call to another attendant by dialing the Individual Attendant Access extension. This category is a type of Individual Attendant Access call with its own identity to allow assignment of a Attendant Priority Queue level.
- Miscellaneous Call — Other calls (such as Automatic Circuit Assurance calls) not covered in the above call categories.

A priority level is assigned to each category, so that calls are answered on a priority basis. The assignment of a priority level to each category is administerable. The same priority level can be assigned to more than one category. By assigning all categories the same priority level, a first-in/first-out queue is achieved.

Calls are prioritized within a Attendant Priority Queue level on a first come first serve basis by a timestamp associated with each call. This timestamp indicates the relative time (with respect to all calls in the queue) when a call was placed in the Attendant Priority Queue after attempting to terminate to the attendant group or an individual attendant.

When at least one call is queued in the Attendant Priority Queue, the Calls Waiting lamp is lighted steady on all active attendant consoles. If the number of calls in the queue reaches the customer-administrable attendant group calls-waiting threshold, the Queue Warning lamp is lighted steady on all active attendant consoles.

Considerations

Note that even though the Attendant Priority Queue feature may reroute an incoming call from an individual attendant to the attendant group under some conditions, the routing reason (and hence the associated Attendant Priority Queue level) for the call is not typically changed. Hence, the reason code display on the answering attendant's alphanumeric display remains the same for the rerouted call

Interactions

An Individual Attendant Access (IAA) call requires special handling by the Attendant Priority Queue feature. If the console to which an IAA call is directed is in the active mode because, the handset/headset is plugged in and the module associated with the individual console is operational, a queued IAA call is not terminated to the individual attendant until that attendant is able to receive the call.

If the handset/headset for the individual attendant console is unplugged, or the module associated with that attendant is not operational, a queued IAA call is rerouted to the global attendant queue.

In general, interaction with System Features which cause a call to be routed to the attendant group or an individual attendant assign a "routing reason" to the call. Each routing reason is used by the Attendant Priority Queue feature to determine the Attendant Priority Queue category (and hence the particular Priority Queue level) for the call.

Therefore, the routing reason designated by a feature determines the priority of a call in the Attendant Priority Queue. Note that even though the Attendant Priority Queue feature may reroute an incoming call from an individual attendant to the attendant group under some conditions, the routing reason (and hence the associated Attendant Priority Queue level) for the call is not typically changed.

Administration

System administration functions are provided to allow the customer to:

- Designate Attendant Priority Queue levels for any or all Attendant Priority Queue call categories (including the highest priority level).
- Designate the attendant group calls-waiting threshold, which is used to light the Queue Warning Indication lamp on all active consoles if that threshold is exceeded.

Note that the "Emergency Call to the Attendant Group" category is set as a default to the highest level of 1, and all other call categories are set to a lower priority of 2. If an attempt is made to change the Emergency Call category from the highest priority level, a warning message is displayed on the administration terminal.

Hardware/Software Requirements

There is no special hardware required for this feature.

Attendant Recall

Description

Allows voice terminal users on a two-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.

Attendant Serial Calling

This is a new feature for G3. The Serial Call feature enables the attendant to transfer trunk calls that return to the same attendant position after the called party hangs up. The returned call may then be transferred to another station within the switch and this can continue to recur. This feature was developed in response to international needs. In some places, trunks are scarce and Direct Inward Dialing services unavailable. This can cause an outside caller to have to redial often to get through to a location because trunks are so busy. Once callers have been able to get through to a switch attendant and have several calls to make to others on the switch, this feature permits them to keep the use of the line into the switch until all their calls are completed.

The Attendant's display shows that the incoming call is a Serial Call. This information is displayed in the Call Purpose area (far right hand side) of the display. The reason code displayed is "sc".

Once the Serial Call feature has been activated it will remain activated until either the trunk drops from the switch or the attendant deactivates the feature manually (by depressing the Serial Call button). Once the attendant answers the serialized call the lamp associated with the serial call button will be turned on. If that button is not administered then the feature will still be activated, however no external indication will show that the feature is active (except the attendant's display). If an attendant received a serialized call but has no serial call button then the feature cannot be deactivated until the trunk hangs up or until an attendant with a serial call button becomes the controlling party.

If no attendants are available then the call will be placed in the attendant's priority queue.

Considerations

Only attendant users can activate this feature. The Serial Call feature is only valid on calls that have only one trunk on the connection. Only one serial call button may be administered per attendant console. The Serial Call button cannot be assigned to an analog or digital station. The feature can be activated on a conference call as long as only one trunk is on the conference.

Interactions

The Following features interact with the Serial Call feature:

- Priority Queue: A special priority exists for serialized calls.
- CDR/Call Charging: CDR/Call Charging must consider that a single outgoing trunk call may have to be broken up and charged to the different parties that dealt with the call.
- The Serial Call feature will only work in a DCS environment if the attendant who is activating the Serial Call feature resides on the same node as

the trunk that the attendant is currently connected to. In addition, the attendant must not be conferenced in with a DCS party when activating Serial Call since this would have two trunks on the connection that is not allowed for Serial Call activation.

Administration

The Serial Call button can only be administered on an attendant console on the console parameter form.

Hardware/Software Requirements

No special hardware is required for this feature.

Audio Information Exchange (AUDIX) Interface

Description

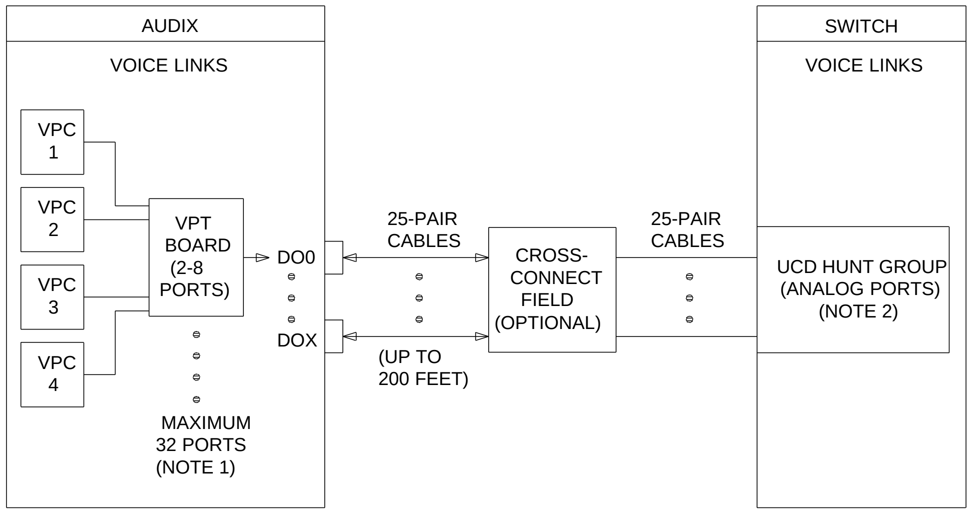
AUDIX is a message-handling system for recording and distributing spoken messages or voice mail. It contains stored voice prompts that guide users to create, send, retrieve, answer, save, and forward spoken messages.

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 Mail— 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 Mail— 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.
- Voice Power Voice Messaging Solution — Send and receive network messages, status information, and administrative update information to and from other members of the AUDIX product family.

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. An AUDIX adjunct 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 or TN746B (A-law) circuit packs. Figure 2-1 shows typical voice connections between the system and the AUDIX adjunct.

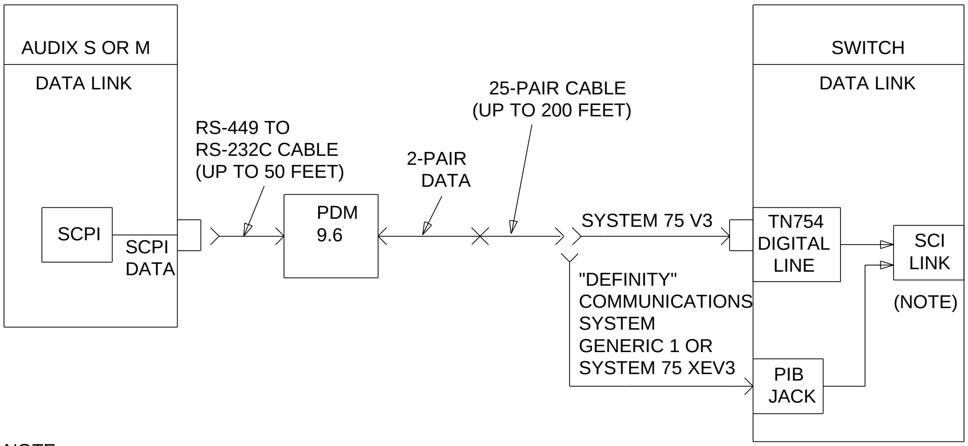


NOTES:
1. AUDIX-S AND AUDIX-M PROVIDE UP TO 16 VOICE PORTS AND IN MOST CASES WILL BE ADEQUATE. HOWEVER, IF THE NEED ARISES (FOR EXAMPLE, FOR DCS, ETC.), AUDIX-L MAY BE USED TO PROIVDE UP TO 32 PORTS.
2. ANALOG PORT CIRCUIT PACKS MAY BE: TN742 (8 PORTS), TN746 (16 PORTS), OR TN769 (8 PORTS).

Figure 2-1. Voice Connections — DEFINITY Generic 1 or Generic 3i to AUDIX

The control link (data link) connection between AUDIX and the switch is via a TN754 Digital Line circuit pack (TN413, TN754B support A-law) or Processor Interface (PI) Board to the Switch Communication Interface (SCI).

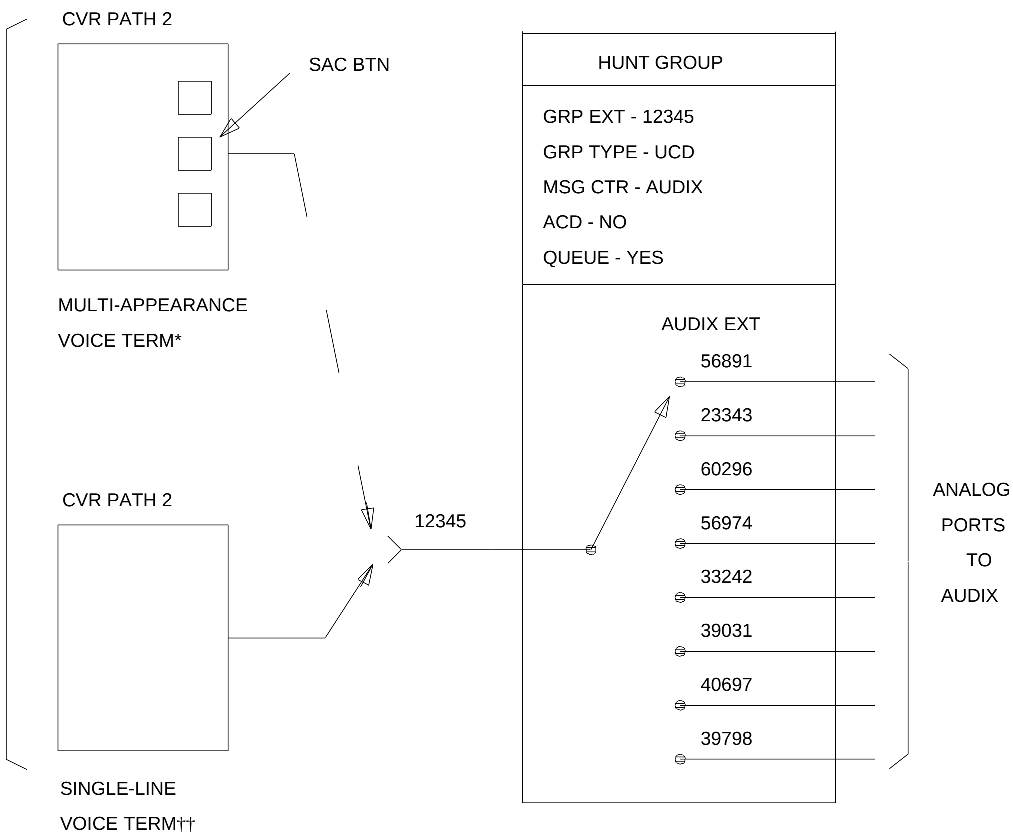
The control link connection is directly to the PI jack (connected to the TN765 Processor Interface circuit pack). If the PI jack is already in use for another adjunct, a TN754 Digital Line circuit pack (TN413, TN754B support A-law) is required. The SCI is provided by the TN765 Processor Interface circuit pack. Figure 2-2 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 2-2. Data Link Connection — AUDIX

System analog ports connected to AUDIX must be assigned to a 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, and so on. Figure 2-3 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 2-3. Simplified AUDIX Arrangement

DEFINITY Generic 1 and Generic 3i allow AUDIX as an 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 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.

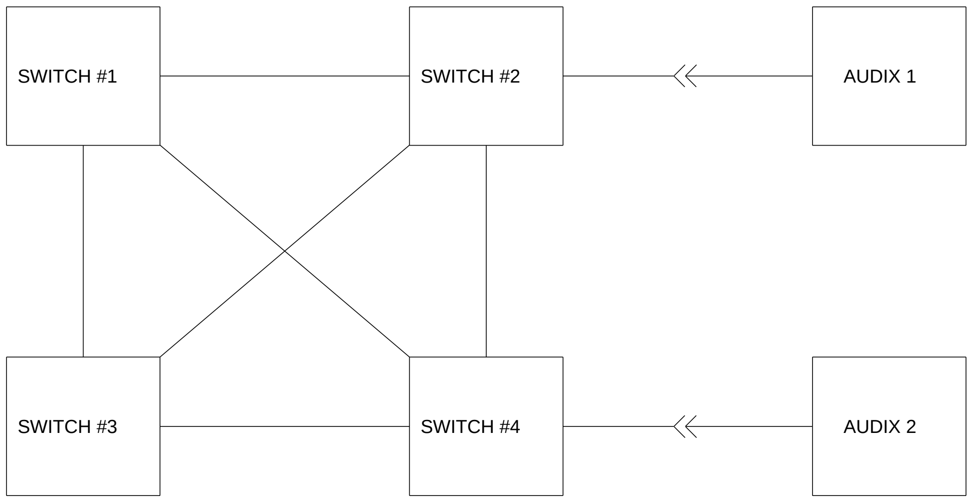
For those systems not equipped with an ACD and CMS, AUDIX traffic measurements can be obtained using the System Measurements feature functionality and AUDIX traffic measurements.

DEFINITY Generic 1 and Generic 3i provide 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 . 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.

DEFINITY Generic 1 and Generic 3i provide 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, Generic 1 and Generic 3i allow the use of AUDIX in a 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, DEFINITY Generic 3i, or other switches that support DCS AUDIX (for example, System 85).

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 2-4 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 2-4. 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 or ISDN-PRI 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, and so on), 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, and so on).

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 mailby 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 mail.

DCS AUDIX

An AUDIX subscriber on any switch in the network may access his or her voice mailby 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 mail.

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 mail.

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 a 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, and so on) 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 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, and so on) 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, and so on) 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 description.

Considerations

Only one AUDIX can be directly connected to a system. However, Generic 1 and Generic 3i allow a system connected to an AUDIX to be a member of a DCS network, with that AUDIX serving the entire network, or any part of the network (see Figure 2-4).

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.

Analog ports on the system connected to an AUDIX adjunct must be assigned to a UCD hunt group. These ports are administered as audix-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 (one 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, and so on, 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.

Attendants are not allowed to transfer out of AUDIX because only the attendant can transfer a call which involves the attendant and this transfer must take place from the attendant's console.

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 AUDIX Interface feature:

- Abbreviated Dialing

  The 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.

- 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 Transfer Out of Audix interacts with Call Coverage as shown in the following table:

Transfer out of AUDIX (Enhanced) and Coverage Interactions

Source	Transfer Destination	Coverage Type
External	Local Station	Internal
	Remote Station (DCS)	External
	Remote Station (ISDN)	Internal
Internal (local)	Local Station	Internal
	Remote Station (DCS)	Internal
	Remote Station (ISDN)	Internal
Internal (remote)	Local Station	Internal
	Remote Station (DCS)	Internal
	Remote Station (ISDN)	Internal

- 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 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).

- LWC

  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 LWC for an AUDIX-covered principal are denied and the reorder tone is returned.

  If a caller attempts to cancel a LWC message sent to AUDIX, the caller receives an 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 a 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/Deactivation

  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.

- 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, a 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 (three-, four-, or five-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 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. 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:

- Processor Data Module (for SCI)
  - Assign Data Module (Processor Interface) when SCI is provided by TN765 circuit pack.
- Interface Link
  - Assign link extension number for Processor Interface.
  - Assign Destination Number = extension number of MPDM assigned to AUDIX (when MPDM is used).
  - Assign Destination Number = eia, 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 — G1.1, and G3i with single-carrier cabinets; 1-8 — G1.1 or G3i 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 7405D (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 = audix.
  - 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) one through four.
- — Assign Chan (two fields) one through 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 one-, two-, or three-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. "ap-spe" may be entered 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 (FAC) 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) are provided by TN742 or TN746B (A-law) circuit packs.

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

- DEFINITY Generic 1 or Generic 3i connected to a two-cabinet AUDIX

  AUDIX data link connection between DEFINITY Generic 1 or Generic 3i 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 Digital Line circuit pack (TN413, TN754B support A-law) and an MTDM and an MPDM (see Note) is required.
- DEFINITY Generic 1 or Generic 3i connected to a one-cabinet AUDIX

AUDIX data link connection between DEFINITY Generic 1 or Generic 3i 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 Digital Line circuit pack (TN413, TN754B support A-law) and an MPDM is required.

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

For DCS AUDIX, DCS software is required.

Authorization Codes

Description

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

The Authorization Codes feature is optional, is closely linked to the FRL feature, and can be used with the ARS, 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 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 CDR 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, 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 end-points, 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 use DID/Tie trunk intercept treatment. Internal calls receive intercept tone.

The System Manager can obtain a printed list of the system's authorization codes by entering **list auth p** at the Manager I terminal (G1) or the G3 Management 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. 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.

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 Time-out 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 come 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 LDN for access to the attendant or, after dialing the remote access number, wait 10 seconds for Time-out 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 DEFINITY Generic 1s or Generic 3is. If a System 85, DIMENSION PBX, DEFINITY Generic 1 switch, or DEFINITY Generic 3i switch is part of the network, then the digit 1 is used as the CACR code. If the digit one is used as the CACR code, then it cannot also be used as the first digit of an authorization code.

If the Time-out 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 that 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 PGNs are assigned according to 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.

- COR and 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 CDR

  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. 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 CDR 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 Time-out 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
  - If possible, 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 (AAR) (G1.1)

Description

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

AAR provides up to six 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 eight. The called number may be a seven-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 North American Numbering Plan numbers are directed to the 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 by, Subnet Trunking on the Routing Pattern.

The 640 private network RNXs may match public network central office codes (NXXs). Therefore, the only way to determine the intended network for seven-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 seven-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 discussed separately in this section.

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 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 home 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 attendant-seeking call and the number of digits deleted in the routing pattern is not seven, 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 seven, the system will delete all the digits and replace these digits with the DDD number given in the routing pattern.

Each RNX can point to any one of 254 Routing Patterns, numbered one 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 FRL assignments. FRLs are fully described 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 of Intertandem and regular (access and by-pass) Tie trunks is important with respect to the routing of certain calls and the outpulsing of digits.

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:

- 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.

- CDR

  An AAR call using a trunk group marked for CDR 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 CDR.

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

  If CDR 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 CDR account code is to be dialed with an ARS call, it must be dialed before the ARS access code is dialed.

- UDP

  The leading one to four digits of the four- or five-digit called DCS extension (PBX Code on Dial Plan form) are converted into an ETN number by inserting an RNX. RNX tables are used to route the UDP call.

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 that 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 Alternate Routing (G3i)

Description

Provides alternative routing choices for private on-network calls. With AAR, the system automatically selects the most desirable (normally the least expensive) route over various trunking facilities for private network calls. AAR also provides digit modification to allow on-network calls to route through the public network when an on-network route is not available.

The private network of PBXs that utilizes the AAR feature is called an 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.

Within an ETN, each switching facility is identified by a unique private network office code. Private network office codes may be two to four digits in length. Throughout the rest of this description, the private network office code will simply be referred to as the "office code."

ETN addresses for DCS or UDP destinations are limited to a seven-digit format. This means that the location code part for UDP/DCS is a three-digit code of the form RNX and the extension number is a four-digit number in the XXXX format (along with limitations that the UDP/DCS number cannot start with a 0). For other destinations, ETN addresses are not limited to the seven-digit RNX format.

The principal use of AAR is to provide routing of private network calls. Private network calls are those 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 an on-network number. AAR then selects the route for the call and performs any necessary digit manipulation. AAR selects the most desirable route for the call. If the first choice route is not available, another route is chosen automatically. AAR provides up to six routes for each office code.

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. Normally, the called number is a private network number. However, it may also be a public network number, a service code, an IDDD number, an operator code 0 (or any other digit assigned to the operator), or a CDOS (Customer Dialed/Operator Serviced) number (0+ or 01+ the number).

Private network (on-network) numbers are handled by the AAR feature. An on-network number can be changed into a public network direct-distance dialing number, a CDOS number, or an IDDD number by, administering the "ars" call-type for such numbers.

The private network location codes may match public network central office codes. Therefore, the only way to determine the intended network for seven-digit calls is by the dialed AAR or ARS access code or by specific administration of the "ars" call-type on the AAR Analysis form.

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 office code is, of course, reserved to represent a particular country and city. At the final on-network switch, the office code is deleted. The international prefix code 011 (in the U.S., 00 in most of Europe, and so on), 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 discussed elsewhere 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 office code 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 ETN tandem or main switch. Toll charges, if any, are from the final ETN switch to the destination.

Each office code can point to any one of 254 Routing Patterns, numbered 1 through 254. More than one office code can point to the same pattern. A blank pattern provides intercept treatment and pattern 254 is the default for all office codes. Routing Patterns are shared with ARS. Access to a route within the pattern is controlled by FRL assignments. FRLs are fully described elsewhere in this chapter. For outgoing ISDN calls, route selection is dependent on Bearer Capability Class (BCC), FRL, and type of facility.

The system may serve as an ETN tandem switch. This distinction as a tandem switch is important with respect to the routing of certain calls. As a tandem switch, 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. Traveling Class Marks (TCMs) are appended to numbers outpulsed on tandem trunks. (TCMs represent the originating user’s FRL.) The system can also access Bypass Tie Trunks to an ETN main switch.

AAR Dialing

AAR begins when a user dials the AAR access code (normally the digit 8), followed by the number to be called.

As soon as the user dials the AAR access code, the system checks to see if the user’s voice terminal extension has been Origination Restricted or Outward Restricted by its assigned COS. The system also checks to see if the user has a Controlled Restriction of either Outward or Total. If any of these restrictions apply, intercept treatment is applied to the call. Otherwise, the AAR call continues and the user can enter the number to be called.

A second dial tone may or may not be heard after the AAR access code is dialed, depending on the system administration.

Inter-Digit Time-out

The system uses a short inter-digit timer and a long inter-digit timer during the dialing process. Normally, a 10-second inter-digit timer is used between each digit for the user to continue dialing. If the digits dialed so far point to a valid destination, but there is a similar string of digits which is of different length, the short three-second inter-digit timer will be started. If dialing does not continue before the timer expires, it is assumed that no more digits will follow, and # is appended by the system to indicate end of dialing. To override the timer for faster call processing, the originator may dial # to indicate end of dialing.

A 10-second long inter-digit timer is used when the digits dialed so far are not a valid destination. But more digits may be required, and a 10-second timer is allowed for dialing the next digit. Time out of this timer results in Intercept tone to the caller if it is not a valid number. To indicate the end of dialing for valid strings with a lesser number of digits, the user may, however, dial # on any of these calls to cancel the time-out interval and indicate end of dialing.

Digit Conversion

Once the AAR access code and the called number are dialed, the dialed number is compared to entries in the Matching Pattern fields of the AAR Digit Conversion Table screen. An example of this screen follows. If all or part of the dialed number matches one of the Matching Patterns on the screen, the dialed number is replaced by a new number from the Replacement String field on the screen. This new number is then used to route the call, the call becomes an ARS call, and is routed using the ARS Analysis Table. This function may be used to route specific dialed number strings to a different number, intercept, and so on. The Digit Conversion Table is only used once per call.

```
                         AAR Digit Conversion Table
                                                          Percent Full:
 Matching Pattern   Replacement String    Matching Pattern   Replacement String
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
-----------------   -----------------     -----------------   -----------------
```

The purpose of AAR digit conversion is simply to convert private network numbers to public network numbers. This allows the system to change specific dialed digits to a public network number and route some calls via ARS. Also, unauthorized private network calls can be routed to an attendant or receive intercept treatment.

Time of Day Routing

After an AAR call passes through AAR digit conversion, the Time of Day Plan Number of the calling party is used to make the choice of an associated Time of Day Routing screen. On this screen, a Routing Plan Number (RPN) is identified based on the time of day. This plan is then used to select the AAR Analysis Table screen, discussed later in this feature description, which will determine how the call is routed.

If Time of Day Routing is not assigned, the user’s PGN is used to select the ARS Analysis screen.

See the AAR/ARS Partitioning and Time of Day Routing feature descriptions elsewhere in this chapter for more information on these features.

AAR Analysis

After an AAR call passes through AAR Digit Conversion and Time of Day Routing, AAR Analysis is performed based on the Time of Day Routing Plan Number or (if Time of Day Routing is not assigned) the user’s PGN.

The system uses AAR Analysis to compare the dialed number with entries in an AAR Analysis Table. When the system finds a dialed string entry in the table that matches the dialed number, the AAR Analysis Table maps the dialed number to a specific Routing Pattern (discussed later) and Call Type. The selected Routing Pattern and Call Type are then used to route the call. If the Call Type field on the AAR Analysis Table for a digit string is "ars," call processing crosses over to ARS and the call is processed as an ARS call. The AAR Analysis Table screen form also shows the minimum and maximum number of digits required for digit analysis of each dialed number. An example of the AAR Analysis Table screen form is shown in Screen 2-2.

Dialed string entries may contain the letter "x" or "X", which is used as a ‘‘wildcard’’ character. This wildcard character matches any of the digits 0-9. For example, a dialed string entry of ‘‘3x’’ applies to all calls beginning with 30 through 39. This ‘‘wildcard’’ makes it possible for traditional three-digit RNXs to be represented in several ways in the AAR Analysis Table. For example, RNXs 200 through 299 can be assigned to the AAR Analysis Table in either of the following ways:

Dialed String	Min. # of Digits	Max. # of Digits
2	7	7
	or	
20	7	7
21	7	7
22	7	7
...	...	...
29	7	7
	or	
2xx	7	7
	or	
20x	7	7
21x	7	7
22x	7	7
...	...	...
29x	7	7

It is possible that some numbers may overlap other numbers. For example, the AAR Analysis Table may have dialed string entries of ‘‘645’’ and ‘‘6452’’ In this case, for example, the number 645-2045 will be routed according to the 6452 entry (the longest dialed string).

When the UDP is used, the three-digit RNX dial string representation must be used on the AAR Analysis Table to match the administration in the UDP table.

Possible Call Types in the AAR Analysis Table are as follows:

- **aar** — Regular AAR call
- **ars** — Crossover to ARS call
- **attd** — Attendant (indicates that the call will be terminated to remote attendant).
- **haar** — Home ETN address (indicates that the call should be terminated locally on the home switch instead of routing to another ETN node. If the

UDP is administered, then that dial plan is used to convert the number to a local extension number. Otherwise, the location code is deleted from the dialed number, and the remaining digits are used to route the call to a local extension).

- **hatd** — Indicates that the AAR call should be sent to to the local attendant group.

```
                          AAR Analysis Table
                        Partitioned Group Number:  1          Percent Full:___
    Dialed                Route  Call        Dialed                Route  Call
    String        Min Max  Pat   Type        String        Min Max  Pat   Type
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
_________________  __  __  ___   ____    _________________  __  __  ___   ____
```

Some special dialing patterns are automatically mapped to a specific Routing Pattern and Call Type. These special dialing patterns, shown in Table 2-1, are preassigned and require no administration. However, they can be changed by the System Manager.

Table 2-1. AAR Analysis Default Translations

Dialed String	Number Of Digits Min.	Number Of Digits Max.	Route Pattern	Call Type
0	1	23	-	ars
1	4	23	-	ars
2	7	7	254	aar
3	7	7	254	aar
4	7	7	254	aar
5	7	7	254	aar
6	7	7	254	aar
7	7	7	254	aar
8	7	7	254	aar
9	7	7	254	aar
X11	3	3	-	ars

Normally, the `Route Pat` (routing pattern) field on the Analysis Table screen contains a routing pattern number (1 through 254). However, this field may instead contain a Remote Home Numbering Plan Area (RHNPA) table number (r1 through r32). When an AAR Analysis table points to an RHNPA table, the next three dialed digits are compared with codes in the selected RHNPA table. Each code on the table is then mapped to a specific routing pattern number (1 through 254).

In summary, AAR Digit Analysis is merely a method of selecting a routing pattern. The routing pattern may be selected in two ways:

- It may be selected directly from the AAR Analysis table.
- The AAR Analysis table may first have to select an RHNPA table which will in turn select the routing pattern.

Routing Patterns

The digit translations performed on an AAR call by the AAR Analysis and RHNPA tables cause a specific Routing Pattern to be selected for the call. The Routing Patterns are numbered 1 through 254. More than one combination of dialed digits can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with AAR, digit translation should always point to a Routing Pattern. If calls to some numbers are to be denied, this should be handled by FRL assignment, not by intercept on the codes. FRLs are discussed elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of up to six trunk groups that can be used for the call. Trunk group access is controlled by FRLs. The digit manipulation necessary to route the call is controlled by the Subnet Trunking feature. (Subnet Trunking is discussed elsewhere in this chapter.)

Otherwise, the digit string to be outpulsed is controlled by AAR.

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 32 RHNPA tables, shared with ARS, can be provided.

Up to eight ARS and AAR partitions may be administered. Routing patterns and RHNPA tables are shared between partitions.

ARS and AAR Analysis tables together can have up to 2,000 entries.

If a customer changes AAR routing assignments, it is the customer's responsibility to notify the RSC network designer and the 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.

Internal memory resources used for AAR Digit Analysis are shared by ARS, AAR, Digit Conversion, and Toll Analysis features. A `Percent Full` field on the ARS and AAR Digit Analysis screens indicates how many of these resources have been used.

Interactions

The following features interact with the AAR Feature:

- ARS

  ARS and AAR can access the same trunk groups and share the same Routing Patterns and RHNPAs. Also, AAR calls can be administered to cross over to ARS via digit analysis and digit conversion.

- 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 required for an incoming trunk call, then an authorization code must be dialed even if the originating FRL was adequate to complete the call.

- CAS

  A CAS Attendant may extend a call out of a Branch PBX by use of AAR. The call is extended over an RLT by dialing the appropriate feature access code and number. The call is routed as determined by AAR administration at the Branch PBX.

- Toll Restriction

  Toll Restriction is not checked on AAR calls, unless they cross over to ARS or digit conversion occurs.

- 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.

  If a multi-appearance voice terminal user has an Automatic Callback button, makes an AAR call, and all trunks are busy, Ringback Queuing is activated automatically.

- CDR Account Codes

  An CDR Account Code may be required for an AAR call if it crosses over and becomes an ARS call via AAR Analysis or AAR Digit Conversion.

- CDR

  An AAR call using a trunk group marked for CDR 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 CDR. The dialed digits are recorded, not the outpulsed digits.

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

  If CDR 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 CDR account code is to be dialed with an AAR call, it must be dialed before the AAR access code is dialed.

- Voice Terminal Display

  The voice terminal display shows the dialed digits (not outpulsed digits). The called-party shown on the display is that of the trunk group actually used. The miscellaneous call identification field on the display will show AAR.

An ISDN-BRI station may format these display fields differently, and the timing of display updates may be different.

- UDP

  When a UDP number is dialed (four or five digits), the routing software initially converts the dialed number to a seven-digit format. The location code in the UDP table equates to a three-digit RNX plus four digits. In a four-digit UDP, the number created is the RNX plus the four extension digits originally dialed. In a five-digit UDP, the number created is the RNX plus the last four extension digits dialed.

  UDP destinations are limited to a seven-digit format. This means that the location code part for UDP/DCS is a three-digit code of the form RNX and the extension number is a 4-digit number in the XXXX format (along with limitations that the UDP/DCS number cannot start with a zero).

  When the UDP is used, the three-digit RNX dial string representation must be used on the AAR Analysis Table to match the administration in the UDP table.

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)
- AAR Analysis Table (one per PGN)
- Up to 32 RHNPA Tables
- Up to 254 Routing Patterns
- FRLs — Assigned via Class of Restriction to each originating facility
- Trunk Groups to be used with AAR
- 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.

Optional ARS/AAR Digit Conversion software is required. This is present in the system when ARS and either UDP or Private Networking are ordered.

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 three-burst ringing (the number of bursts is adminstrable). 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 160 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 redial 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 COS 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 call-back 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
- A VDN Extension (G3i)

Administration

The System Manager assigns Automatic Callback to individual voice terminals by their COS. 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 (ACD)

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. ACD data, transmitted from the switch to the CMS or BCMS, 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 algorithms (depending on administration) to select an idle available terminal or console. The two types of hunting that can be used are ‘‘direct’’ hunting (administered as DDC hunting) and ‘‘most-idle agent’’ hunting (administered as UCD 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 (discussed later), the call routes to the next split agent with all call appearances idle, 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 waited the longest period of time since completing an ACD call (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. However, at any one time, an agent can be logged into a maximum of three ACD splits.

In addition to the agents, a split supervisor can be assigned to each split. The split supervisor can listen in on agent calls, monitor the split queue status (discussed 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

When all agents are active on a call or in After Call Work, the queue allows incoming calls to await an idle terminal.

NOTE:
If no agents are logged in or all agents are in AUX work mode, calls are not allowed to queue at the split. A busy signal is returned to the caller unless the call has come in via an automatic-in central office facility, in which case the caller hears ringback from the central office and the system continues trying to place the call in queue. Automatic-in central office facilities are incoming trunks that do not send address digits (for example, an incoming destination for the trunk would have been assigned and the calls would go to that destination with no switch intervention.)

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 (zero 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.

NOTE:
If a call forwards from a dummy split to a destination that has forced first announcement administered, the caller will hear the first announcement if not forced. The caller will not hear the first announcement if it is forced.

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.

After a forced first announcement, the caller always hears ringback until the call is answered or until it is connected to a second delay announcement. After a first or second delay announcement, the caller hears music on hold (if administered).

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 (either on an ACD call or in ACW mode), the call enters the split queue. If a split queue is not assigned, if the queue is full, if there are no agents logged in, or if all the logged in agents are in AUX, and the incoming facility is a digit-oriented facility (digits are being sent to the PBX as in DID, incoming wink, or immediate tie trunks), the caller receives busy tone or the call is redirected by the Intraflow feature (discussed later) associated with Call Coverage and Call Forwarding.

NOTE:
Central office trunk (non-DID) calls receive ringback from the central office, so the PBX cannot provide a busy signal to these callers. The system will keep trying to put such calls into queue until successful or until the caller abandons.

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.

After the first delay announcement, the caller will hear music on hold (if administered).

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 or music 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.

Announcement Rules: The following rules govern which announcements a caller hears:

1. Calls coming directly into a split will always hear a forced first announcement if assigned, regardless of subsequent treatment such as Call Coverage, Call Forwarding, Night Service, or busy signal. If these calls queue for a sufficient period of time, they will hear delay first and second announcements.
2. Calls that reach a split via Call Coverage from another split or a station will NOT receive a forced first or delay first announcement at the destination split. They will hear a delay second announcement if administered and if the delay interval is met. The assumption is the caller has heard a first announcement at the original destination or that a call redirected from a station should not receive the first announcement.
3. Calls that reach a split via Call Forwarding from another split or a station WILL receive delay first and second announcements at the destination split if administered and the delay interval is met. These calls will receive a forced first announcement at the original split (if administered) but will NOT receive a forced first announcement at the "forwarded-to" split.

Intraflow and Interflow: The Intraflow feature allows splits to be redirected to other destinations on the system. This is accomplished via the Call Forwarding or Call Coverage features. Splits can 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 via Call Coverage 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. The Coverage Don't Answer Interval does not begin until after the first announcement completes (if assigned). 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, consult the feature 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, consult the Queue Status Indications feature description in this chapter.

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 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 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. An agent performs the following steps to log in:
  1. Dial the login Feature Access Code (FAC).
  2. Enter the two-digit split ID number. All split ID numbers consist of two digits. If an agent wants to access split 4, he or she would enter **0** and **4** .
  3. Enter the agent ID if CMS is being used.

  In addition, an agent can log out to let the system know that he or she is unavailable for ACD calls. To log out, the agent performs the following steps:
  1. Dial the logout FAC.
  2. Dial the two-digit split ID number.
- 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, zip tone is not heard. Therefore, trunk groups terminating to auto answer positions and auto answer agent positions 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. Entering the AUX work mode in one split does not affect the agent's status in other splits. The agent is not in the Most Idle Agent queue while in AUX work mode.

- After Call Work — An agent can enter the 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 (the agent is placed in the AUX work mode for other splits). The agent is in the Most Idle Agent queue, but he or she is unavailable while in ACW.

- Auto-In or Manual-In — An agent can enter either the Auto-In mode or the Manual-In mode to become available for ACD calls.

  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 reenter either the Auto-In mode or Manual-In mode to become available for ACD calls.

  With G3i, an agent may be required to enter a Stroke Count or Call Work Code when in the Manual Mode. For details on this interaction, see the ''Forced Entry of Stroke Counts and Call Work Codes'' later in this ACD description.

- 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. If the agent presses the Drop button, he or she will receive dial tone and be unavailable for calls. The Drop button is not recommended for disconnecting calls.

CMS

CMS is an optional adjunct to the system that collects and processes ACD data. CMS uses this data to generate various reports on the status of agents, measured splits, measured trunks and, on G3i, measured VDNs and vectors. These reports can be stored for later use or can be displayed on a terminal for real-time information.

For information on CMS, see the *3B Call Management System Administration,* 585-215-504.

BCMS

The BCMS feature provides real-time and historical reports that assist a customer in managing individual agents, ACD splits (hunt groups), trunk groups and VDNs. 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 (G1), the G3 Management Terminal, or printed on demand on the printer associated with the Manager I terminal or G3 Management 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 manual.

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.

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.

Direct Agent Calling (G3i)

Direct Agent Calling allows an adjunct to transfer a call to a particular ACD agent and have the call treated as an ACD call.

Calls that originally enter the switch as ACD calls and are rerouted to a particular agent via adjunct routing are treated as ACD calls for the duration of the call. This is important for a number of reasons. Agents need to receive zip tone when these calls are delivered. Agents may have After Call Work associated with these calls. The CMS and the BCMS correctly measure these calls as ACD calls.

Delivery of Direct Agent Calls

If the agent receiving the direct agent call is available to answer an ACD call in the associated split, the direct agent call is delivered to the agent. Zip tone is applied if the agent is in the automatic answer mode.

If the receiving agent is not available to answer an ACD call, (for example, the agent is busy on a call, in the After Call Work mode, or in the Auxiliary Work Mode), the receiving agent is notified with a ring-ping if the agent has a multi-function voice terminal or is on-hook. If the receiving agent has a single line voice terminal and is not available, the receiving agent will hear call waiting tone (even when the Call Waiting feature is not assigned) if the agent is off-hook. The ring-ping or call waiting tone is given only once per call when the call is queued. The Auto-In and Manual-In button lamps for the associated split on the receiving agent’s voice terminal flash, indicating a direct agent call is waiting. Flashing starts when the call queues and stops when all direct agent calls leave the queue (answered, abandoned, or sent to coverage).

Direct agent calls are queued and served in a first-in first-out order before any non-Direct Agent Call (including priority calls). Therefore, when an agent becomes available, the switch first checks for any direct agent calls before serving normal ACD calls in queue.

The voice terminal display for the receiving agent before a transfer is complete shows the originating agent’s name and number. The voice terminal display for the receiving agent after the transfer is complete shows the caller identification (CPN/BN or trunk group name for external calls, and name/number for internal calls) and the original split or VDN name.

Direct agent calls follow the receiving agent's coverage and call forwarding, if activated. Once the call goes to coverage or is forwarded, the call is no longer treated as a direct agent call. CMS is informed that the call has been forwarded.

Answering a Direct Agent Call

The receiving agent answers a direct agent call by becoming available in the split with which the direct agent call is associated. While on a direct agent call, the agent becomes unavailable to subsequent ACD calls.

If the receiving agent logs off by unplugging the headset, the agent may still answer a direct agent call in queue by logging back in and becoming available. Agents who have direct agent calls waiting will be denied if they attempt to log-off using a feature access code.

Vector-Controlled Splits

For detailed information on vector-controlled splits, see ‘‘ACD Split/Hunt Group Operation with Call Vectoring’’ in the Call Vectoring feature description elsewhere in this chapter.

Stroke Counts (G3i)

Stroke Counts provide ACD agents with the ability to record customer-defined events on a per-call basis when the CMS is active. For details on the Stroke Counts function, see the Agent Call Handling feature description elsewhere in this manual.

Call Work Codes (G3i)

Call Work Codes allow ACD agents to enter up to 16 digits for an ACD call to record the occurrence of customer-defined events (such as account codes, social security numbers, or phone numbers). For details on the Call Work Codes function, see the Agent Call Handling feature description elsewhere in this manual.

Forced Entry of Stroke Counts and Call Work Codes (G3i)

An agent is always allowed to enter a Stroke Count and/or Call Work Code for an ACD call. However, each split can be administered so that agents in that split are forced to complete a Stroke Count and/or a Call Work Code entry for every call answered in the Manual-In mode. The rest of this discussion assumes this has been done.

An agent can enter the Stroke Count and/or Call Work Code while on the call, or while in the ACW mode after the call releases. After a call has been released by an agent in the Manual-In mode, the agent automatically enters the ACW mode.

The agent is not permitted to return to the Manual-In mode until a Stroke Count or a Call Work Code is completed. If the Manual-In button is depressed before a Stroke Count or a Call Work Code has been completed, the Manual-In lamp flashes. If the Manual-In FAC is used before a Stroke Count or a Call Work Code has been completed, intercept tone is given.

Once a Stroke Count or a Call Work Code entry is completed, pressing the Manual-In button (or FAC) returns to Manual-In mode, and lights the Manual-In lamp.

ACD agents with an attendant console or multi-appearance voice terminals can enter Stroke Counts or Call Work Codes.

An Agent is permitted to be logged into three splits at the same time. Any of these splits may have the Forced Entry option active. A transition into the Aux-Work mode in any split will remove the Forced Entry requirement for all other splits.

The ACD feature must be enabled on the System Parameters Customer Options form. The Call Work Code feature may also be enabled on this screen. If Call Work Code is not selected, the Forced Entry capability applies only to Stroke Counts.

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 ACD parameters are as follows:

Parameter	Maximum G1.1	Maximum G3i
ACD Splits	99	99
Agents Per Split	200	200
Agents Per System	500	500
Split Supervisors Per System	99	99
Agents Measured By CMS	400	400
Agents Measured By BCMS	30	200
Agents Per Split Measured By BCMS	30	30
Splits Measured By CMS	60	99
Splits Measured By BCMS	30	99
Calls In Queue, Per Split	200	200
Calls In Queue, Per System	1000	1000

A voice terminal or individual attendant can be an agent in one or more splits. However, an agent cannot be logged into more than 3 splits simultaneously. If an agent is assigned to more than one split, each assignment applies to the maximum number of agents.

Each system can contain up to 64 (G1.1) or 128 (G3i) 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-channel announcement board and queuing is based on whether or not one of the 16 channels is available. When a channel becomes available, any of the announcements on the board can be accessed. Therefore, a caller may be in queue for an announcement (because a channel 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 announcement at the same time.

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.

Agents 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. The agent may, however, receive non-ACD calls while active on an ACD call.

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 system-wide 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 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.

For MEGACOM 800 Service with DNIS over a wink/wink trunk, 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 (unlike other automatic-in trunk groups which receive ringback from the central office).

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.

CO switches will usually drop calls which remain unanswered after a period of two to three minutes. Therefore, if an incoming CO call queues to a split without hearing an announcement or music, and the caller listens to CO ringback for two to three minutes, the call will be dropped by the CO.

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.

Agents should not be used for hunt group calls and ACD split calls simultaneously. Otherwise, all of the calls from one split (either ACD or hunt group) will be answered first. For example, if the ACD calls are answered first, none of the hunt group calls will be answered until all of the ACD calls are answered.

The oldest call waiting termination is only supported for agents who are servicing ACD calls only.

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.

  A vector-controlled split cannot be assigned a coverage path.

  For a call to an ACD split to be redirected to Call Coverage on the busy criterion, one of the following conditions must exist:

  - Each agent in the split must be active on at least one call appearance and the queue, if there is one, must be full
  - No agents are logged in
  - All agents are in AUX work mode

  If the queue is not full, a call will enter the queue when at least one agent is on an ACD call or in ACW mode. Queued calls remain in queue for a time interval equal to the Coverage Don't Answer Interval before redirecting to coverage. If any agent in the split becomes 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 or a first delay announcement at the covering split, if administered. The redirected call will receive a second delay announcement only.

  Calls to a split that are directed to an agent's voice terminal will not follow the agent's call coverage path. Activating Send All Calls for an agent terminal will not affect the distribution of ACD calls. An ACD call directed to an agent's station will follow the split's call coverage path if the specified "don't answer" interval is met at the agent's set.

- 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 removed from the queue and forwarded.

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

  Calls can be forwarded to destinations outside your PBX (that is, domestic and international phone numbers on the public-switched telephone network).

  On calls forwarded to an ACD split, the caller will hear the forwarded-to split's first and second delay announcement(s), if assigned. A forced first announcement at the forwarded-to split will not be delivered.

- 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 activated will not hear the zip tone that is normally heard.

- DDC and 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 Auxiliary 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.

- 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.

- Dial Intercom

  An agent with origination and termination restriction can receive ACD calls and can make and receive Dial Intercom calls.

- 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.

- Intrusion

  Intrusion will not work with ACD since an ACD extension has many agent extensions. Therefore, it would not be possible for the switch to determine which agent extension to intrude upon.

- 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 for the original split, 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 night destination hunt group are busy, the caller will hear the following, if assigned: forced or delayed first announcement, ringback, 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 three-burst ringing is heard (if three bursts have been administered for priority calls).

  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.

- CDR

  When a CO call enters a full ACD split queue, CDR and the CMS may show different measurements. CDR 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.

- Termination Restriction (COR)

  A station that is in a COR with Termination Restriction can receive ACD calls.

- Transfer

  Calls cannot be transferred to a busy split. The transfer operation will fail and the transferring party will be re-connected to the call. If the transferring party depresses **Transfer**, dials the hunt group extension number, and then disconnects (and the split is busy), the call will be disconnected.

- 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) are 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, BCMS or both.
- Whether or not the split is adjunct-controlled.
- Whether or not each split is vector-controlled (G3i). If a split is vector-controlled, announcement attributes, night service destination, intraflow, coverage path, and message information cannot be administered for the split.
- Whether split measurements are to be internal, external, both, or neither.
- 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.
- four-Digit security code.
- Type of Message Center the split serves as (AUDIX or blank).
- Whether or not the split is served by a queue.
- Queue length (1 to 200 calls).
- Queue Warning Threshold for number of calls (1 to 200 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).
- Whether or not agents in the split are required to enter Stroke Counts and/or Call Work Codes (G3i).

Hardware and Software Requirements

Each auxiliary queue warning level lamp requires one port on a TN742 or TN746B (A-law) Analog Line circuit pack. A beehive-type indicator lamp may be used as a queue warning level lamp. This lamp is available from the Custom Work Group. The lamp operates on ringing voltage and can be mounted at a location convenient to the split.

Each delay announcement requires either one port on a TN750 Integrated Announcement circuit pack or external announcement equipment and one port on a TN742 or TN746B (A-law) Analog Line circuit pack.

NOTE:
There are 16 "ports" available for assignment as extension numbers to be associated with a given announcement. However, any announcement may be played over any one or more physical channels on the integrated announcement board.

If music is to be heard after an announcement, a music source and a port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) 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. Additionally, if a CMS is to be used, CMS software is required. If a split is to be vector-controlled (G3i), Call Vectoring software is required. If Call Work Codes are to be entered, Call Work Codes software must be enabled (G3i).

Automatic Circuit Assurance

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 three 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 one 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 one 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 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.

If ACA referral calls are sent off the PBX generating the referral, the display information indicating the failed trunk will be lost even if the referral call is made over a DCS network.

Interactions

The following features interact with the Automatic Circuit Assurance feature:

- 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 Hold

Automatic Incoming Call Display

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 7507D voice terminal
- Line 1 of the display on a 7506D voice terminal
- 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 Data Communications 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 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 (ARS) (G1.1)

Description

Routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

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 — 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 central office or by a carrier access code.
- Foreign exchange — Used to emulate local calling in an area not served by the local central office. Like the local central office, the foreign exchange office provides a choice of long-distance carriers.
- WATS — Used to provide calling to predefined geographic areas at a rate based on expected usage.
- Tie trunks — Used to provide access to an ETN, or to an EPSCS or 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. 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 followed by a Direct Distance Dialing (DDD) number is also possible. In that case, route selection is based on the DDD number, not on the 10xxx.

To use ARS, the user simply dials the ARS access code 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 three-digit translators are provided: one for the office codes within the home NPA and one for the foreign NPAs. Thirty-two six-digit translators are provided, allowing call routing based on the office codes within the foreign NPA rather than on the NPA alone. (Six-digit translators are actually three-digit translators that are accessed from the foreign NPA translator. The foreign NPA translator can yield one of the six-digit translators based on the NPA. The six-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 six-digit translators. These translators are also known as 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 one 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 central office, not by the system. By allowing the unassigned codes to be intercepted by the central office, 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 FRL assignment, not by intercept on the codes. FRLs are discussed 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 discussed 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 discussed 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, some areas may require the prefix.

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. This situation is reflected on the Dial Plan form in the 1 Prefix Required field. The digit tells the system whether to route the call as a seven-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 for seven-digit calls 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 such adjacent NPA calls, then it must be dialed on the home NPA toll calls so the system can differentiate between the two 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 by the corresponding toll tables. 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. Five choices are available and are identified in translations by a Prefix Mark. Values and their meaning are as follows:

- Prefix Mark 0 — Suppress a user-dialed Prefix digit 1 for 10-digit FNPA calls, but leave a user-dialed Prefix digit 1 for the following types of calls:
  - seven-digit HNPA calls
  - 10-digit calls that are not administered as FNPA or HNPA types
- Prefix Mark 1 — Send a 1 on 10-digit calls, but not on seven-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.
- Prefix Mark 4 — Always suppress a user-dialed 1.

NOTE:
This capability is required, for example, when routing ISDN-PRI calls to an AT&T 4ESS. If the prefix digit 1 were not suppressed, then

the 4ESS would reject such calls.

Which of the five 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 prevents the system from sending a 1 prefix digit for 10-digit FNPA calls. However, the system leaves a user-dialed prefix digit 1 for 7-digit HNPA calls. 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 the toll information for all seven-digit calls only. 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. All 10-digit calls are considered to be toll calls.

Prefix Marks are only applicable on public network destinations. Requirements for a 1 are specified via Prefix Marks when the call accesses the public network. With G1.1, the digit 1 is transmitted if the Prefix Mark is non-zero.

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. On a 10-digit call, 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.

The NPA is inserted on seven-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.

Another case of NPA insertion occurs when an ARS call accesses an ETN intertandem trunk. 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 switch. Therefore, all ARS calls (to a tandem switch) accessing an ETN intertandem trunk are 10-digit calls. This enables the system to distinguish between ARS off-network calls and the seven-digit on-network calls. For prefix digit 1 outpulsing, see the last paragraph in the previous section on Digit 1 Outpulsing.

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 three 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, unless it is a rotary or ISDN trunk.

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 special codes in the FNPA translator, meaning that six-digit translation can be used. This is particularly useful on international calls, since the six-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 three-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 six-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 2-2 which lists the special dialing patterns along with the associated HNPA or FNPA table entry through which that type of call is routed.

Table 2-2. 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)	001X...X	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
LOCAL/TOLL CALLS	(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
TOLL CALLS WITHIN HOME NPA	(1)HNPA-NXX-XXXX	NXX	HNPA
LONG DISTANCE FOREIGN 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	011	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-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 FOREIGN 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 three-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 inserted via subnet trunking 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 RSC network designer and the 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 the Automatic Route Selection feature:

- 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

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 GRS 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 assigned, makes an ARS call, and all trunks are busy, Ringback Queuing is activated automatically.

- CDR

  An ARS call using a trunk group marked for CDR 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 CDR.

  If CDR 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 CDR account code is to be dialed with an ARS call, it must be dialed before the ARS access code is dialed.

- 10-Digit to Seven-Digit Conversion

  If 10-Digit to Seven-Digit Conversion is used, normal ARS call routing may be affected. This feature converts 10-digit public network numbers to seven-digit private network numbers prior to call routing. See the 10-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 (one to three digits)
- ARS Access Code 2 (one to three digits)
- 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 six-digit translator so the call will be routed on both the NPA and the office code.
- Up to 32 Remote Home NPA Tables — Provides six-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 area code of the distant end of the trunk group) is a local or toll call.
- FRLs — Must be assigned via a COR to each originating facility. 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.
- 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 FRL and Subnet Trunking descriptions for details.)
- Whether or not the system returns dial tone after the ARS 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 (TN420C, TN744 support A-law). These additions are, however, cost effective when compared to the alternatives for call routing.

Optional ARS software is required.

Automatic Route Selection (ARS) (G3i)

Description

Routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

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** — 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 central office or by a carrier access code.
- **Foreign exchange** — Used to emulate local calling in an area not served by the local central office. Like the local central office, the foreign exchange office provides a choice of long-distance carriers.
- **WATS** — Used to provide calling to predefined geographic areas at a rate based on expected usage.
- **Tie trunks** — Used to provide access to an ETN, or to an EPSCS or CCSA office. (In some cases, it is preferable to allow a private network to handle the routing of calls destined for the public network.)
- **ISDN-PRI** — Used for calls over an ISDN and provides users access to a variety of switched nodal services such as MEGACOM telecommunications service, INWATS, and ACCUNET digital service and allows access to other inter-exchange carriers.

A variety of numbers can be called using ARS, including seven-digit numbers, 10-digit numbers, International Direct Distance Dialing (IDDD) numbers, service codes, Customer-Dialed Operator-Serviced (CDOS) numbers (for example, 0+ or 01+), and Inter-Exchange Carrier (IXC) numbers.

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 assigned in translations to best benefit the customer on any given call. These codes are inserted as needed to guarantee automatic carrier selection.

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.

ARS Dialing

ARS begins when a user dials the ARS access code (normally the digit 9), followed by the number to be called.

As soon as the user dials the ARS access code, the system checks to see if the user's voice terminal extension has been Origination Restricted or Outward Restricted by its assigned COR. The system also checks to see if the user has a Controlled Restriction of either Outward or Total. If any of these restrictions apply, intercept treatment is applied to the call. Otherwise, the ARS call continues and the user can enter the number to be called.

A second dial tone may or may not be heard after the ARS access code is dialed, depending on the system administration.

Inter-Digit Time-out

The system uses a short inter-digit timer and a long inter-digit timer during the dialing process. If the digits dialed so far are a valid destination (for example, 0) but more digits may follow, the short three-second inter-digit timer will be started. If dialing does not continue before the timer expires, it is assumed that no more digits will follow. To override the timer for faster call processing, the originator may dial # to indicate end of dialing. To prevent a time-out race condition between the caller's switch and other network switches, a 10-second short inter-digit timer is used for incoming tie trunk calls.

A 10-second long inter-digit timer is used when the digits dialed so far are not a valid destination. Time-out of this timer results in Intercept tone to the caller. The user may, however, dial # on any of these calls to cancel the time-out interval and indicate end of dialing. Use # to indicate end of dialing is applicable only with touch-tone stations in the public network.

Special Dialing Patterns

The system recognizes certain dialing patterns on outgoing calls and routes these calls accordingly. The descriptions of these dialing patterns reflect the system defaults as used in the United States. Other countries may require different administration of these values. The following dialing patterns are recognized:

- **DDD Calls With Prefix Digit 1 Required**

  The user may or may not be required to dial a 1 before dialing a seven- or 10-digit number, depending on the system's dial plan administration. 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 central office codes. Therefore, it is possible that some NPA codes, also called ''area codes'' may also serve as CO codes. In this case, the digit 1 must be dialed if a 10-digit call is intended. The first digit tells the system whether to route the call as a seven-digit call within the home NPA (1 not dialed) or as a 10-digit call to another NPA (1 dialed). In this case, the dial plan should be administered so that the user is required to dial 1 for 10-digit calls.
  - 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, central office codes may be duplicated in the home and adjacent NPAs. Also a CO code in the home NPA may be a toll call. Therefore, if the digit 1 is not required on certain adjacent NPA local calls, then it must be dialed on the home NPA seven-digit toll calls so the system can differentiate between the intended destinations.

- **DDD Calls with Prefix Digit 1 Not Required**

  The first digit following the ARS access code may or may not be a 1. In systems where the 1 prefix is dialed, but not required (as administered on the Dial Plan form), dialing the 1 prefix before a 10-digit call is optional and the prefix will be ignored.

- **IDDD Calls**

  IDDD numbers consist of a Country Code and a National Number. The National Number is simply the number used when calling within the country. The Country Code can be from one to three digits in length. In the NANP the National Number is 10 digits in length. The Country Code and National Number together cannot exceed 12 digits. In the NANP, international numbers are recognized by special prefix codes:

  — **011** — Indicates that the caller is making a station paid direct international call. The Country Code and National Number follow the 011 prefix.

  — **01** — Indicates the caller desires operator assistance on an international call, such as person-to-person, credit card, collect call, and so on. The Country Code and National Number follow the 01 prefix.

- **Operator Assistance Calls**

  The first digit following the ARS access code is a 0. If a 0 is dialed by itself to access an operator, a special inter-digit time-out occurs, the route for dial 0 calls is selected and a 0# is outpulsed. If the user dials another 0, the route for 00 is selected and a 00# is outpulsed. The call is routed to the toll operator (if one exists) instead of the local operator in this case.

- **Operator Assisted and International Calls**

  The first digits following the ARS access code are 0 (operator) or 00 (toll operator) optionally followed by a 10-digit DDD number, or 01 or 010 (international operator) for international dialing followed by international destination address digits. Because of the variable number of digits required on these calls, an inter-digit time-out is used to recognize end of dialing.

- **Special Service Codes**

  The first three digits following the ARS access code are of the form X11 (where X = 0 through 9) with or without dialing the 1 prefix digit. This is called a service code. These are recognized as complete addresses if no further digits are dialed, and are routed to the appropriate facility. If it is administered with a length of three to seven digits, the inter-digit time-out determines whether the call is a three- or seven-digit call. For example, if the user dials 911, the call will route to the police/emergency operator; if the user dials 811-XXXX, the call will be translated as a seven-digit call for the repair bureau corresponding to the last four digits (811 is a service code for repair). In any case, the call is routed based on the first three digits (X11) for these special services. If the first three digits after the prefix digit (if any) are not in the form X11, further processing is required to route the call.

- **Calls Dialed with Inter-Exchange Carrier (IXC) Access**

  The first digits following the ARS access code are an IXC Access Code. The access code may be followed by a DDD or an IDDD number. This gives the user control over which carrier or facilities should be used for routing the call. The call is routed based on the administration of the IXC prefixes in the ARS Digit Analysis Table discussed later in this chapter.

The system supports access to three general IXC arrangements which are commonly referred to as Feature Groups A, B, and D:

- Feature Group A access dialing is of the form NXX-XXXX (where N is any digit 2 through 9, and X is any digit from 0 through 9) and may be followed by a Personal Identification Number (PIN) (for example, 800-XXXX).
- Feature Group B access dialing is of the form 950-0XXX or 950-1XXX (where X is any digit from 0 through 9) and may be followed by a PIN.
- Feature Group D access dialing is of the form 10XXX (where X is any digit from 0 through 9).

From a caller's perspective, the major differences between use of the various groups are:

- Access to Feature Groups A and B requires the dialing of seven digits, whereas access to Feature Group D requires just five digits.
- Single-stage dialing is supported for access to Feature Group D, whereas access to Feature Groups A and B requires two-stage dialing. Two-stage dialing means that there is a pause for dial tone between the two groups of dialed digits.
- No customer identification digits are required for access to Feature Group D.
- A touch-tone telephone is required to enter a PIN code when accessing Feature Group A or B. A rotary or touch-tone telephone may be used with Feature Group D.

Digit Conversion

Once the ARS access code and the called number are dialed, the dialed number is compared to entries in the Matching Pattern fields of the ARS Digit Conversion Table screen. An example of this screen is shown below. If all or part of the dialed number matches one of the Matching Patterns on the form, the dialed number is replaced by a new number from the `Replace` field on the form. This new number is then used to route the call. The new number will route the call via AAR over a private network. If no corresponding entry is found in the AAR Digit Analysis table, the ARS Digit Analysis table will be searched for a match with the new modified number and routed accordingly.

```
                         ARS Digit Conversion Table
                                                         Percent Full:
 Matching Pattern   Replacement String    Matching Pattern   Replacement String
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
-----------------   -----------------    -----------------  -----------------
```

If no match is found on the ARS Digit Conversion Table screen, the dialed number is compared with digit string entries on the system's Toll Analysis form for a relationship with the system's Restricted Call List or Unrestricted Call List. If the dialed number matches a number assigned to a Restricted Call List, the call may not be allowed to complete. Also, if the user is toll restricted and the dialed number matches a number on the Toll List, the call may not be allowed to complete. Calls that are not allowed to complete receive intercept treatment.

The purpose of ARS digit conversion is simply to convert public network numbers (to be routed with ARS) to private network numbers (to be routed with AAR). This can save toll charges and allow users to route calls on the customer's private network facilities. Also, unauthorized public network calls can be blocked and routed to an attendant or receive intercept treatment via digit conversion.

The following table shows several ARS Digit Conversion examples.

The following conditions are assumed for the examples: ARS Access Code = 9, AAR Access Code = 8, Home RNX (Private Network Office Code) = 222, Prefix 1 is required on all long-distance DDD calls, Dashes "-" shown in the table are for readability only.

Table 2-3. ARS Digit Conversion Examples

Operation	Actual Digits Dialed	Matching Pattern	Replacement String	Modified Address	Notes
DDD call to ETN	9-1-303-538-1345	1-303-538	362	362-1345	The call will be routed via AAR on the route selected for RNX 362.
Long-distance call to presubscribed carrier	9-10222	10222+DDD	(blank)	(blank)	The call will be routed as dialed with the DDD number over the customer's network facilities.
Terminating a local DDD call to an internal station	1-201-957-5567 or 957-5567	1-201-957-5 or 957-5	222-5	222-5567	The call goes to the home RNX 222, Extension 5567
Unauthorized call to intercept treatment	9-1-212-976-1616	1-XXX-976	#	(blank)	The "#" Signifies the end of dialing. Any digits dialed after 976 are ignored by ARS. The user will receive intercept treatment.
International calls to an attendant	9-011-91-672530	011-91	222-0111#	222-0111	The call is routed to local switch (RNX 222), then to the attendant (222-0111). This method may also be used to block unauthorized IDDD calls. The call can be routed to an announcement by replacing 0111 with an announcement extension.
International call from certain European countries	0-00-XXXXXXXX	00	+00+	w00wXXXX 0	The first zero denotes ARS, the second pair of zeroes denotes an international call, the pluses denote "wait", and the modified address "w" shows how wait is modified.

NOTE:
The dialed digits are matched to the Matching Pattern that most closely matches the dialed number. For example, if the dialed string is 957-1234 and matching patterns 957-1 and 957-123 are in the table, the match is on

pattern 957-123. The call will be routed as dialed.

Time of Day Routing

After an ARS call passes through ARS digit conversion (with no Matching Pattern found) and toll analysis allows the call, the Time of Day Plan Number of the calling party is used to make the choice of an associated Time of Day Routing form. On this form, a RPN is identified based on the time of day. This plan is then used to select the specific partition of the ARS Digit Analysis form, discussed later in this chapter, which will determine how the call is routed.

If Time of Day Routing is not assigned and partitioning is enabled, the user's PGN is used to select the specific ARS Digit Analysis form.

See the AAR/ARS Partitioning and Time of Day Routing feature descriptions elsewhere in this chapter for more information on these features.

ARS Digit Analysis

After an ARS call passes through ARS Digit Conversion, Toll Analysis, and Time of Day Routing, ARS Digit Analysis is performed based on the Time of Day Routing Plan Number or (if Time of Day Routing is not assigned) the user's PGN.

The system uses ARS Digit Analysis to compare the dialed number with entries in an ARS Digit Analysis Table. When the system finds a `Dialed String` entry in the table that matches the dialed number, the ARS Digit Analysis Table maps the dialed number to a specific Routing Pattern, discussed later in this chapter, and Call Type. The selected Routing Pattern will then be used to route the call. The ARS Digit Analysis Table screen also shows the minimum and maximum number of trailing digits required for digit analysis of each dialed number. An example of the ARS Digit Analysis Table screen follows.

Possible Call Types in the ARS Digit Analysis Table are as follows:

- **fnpa** — 10-digit call within North America
- **hnpa** — 7-digit call within North America
- **int** — International call (In the United States, the international prefix is 011)
- **iop** — International operator
- **op** — Operator-assisted call (0+)
- **svc** — Service call (such as 811 for repair or 911 for emergency)
- **natl** — National numbers within a country (used outside North America)
- **unk** — Unknown call

```
                          ARS Digit Analysis Table
                        Partitioned Group Number:  1          Percent Full:___
    Dialed                    Route  Call          Dialed                  Route  Call
    String         Min Max     Pat   Type          String        Min Max    Pat   Type
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
_________________  __  __  ___  ____    _________________  __  __  ___  ____
```

Some special dialing patterns are automatically mapped to a specific Routing Pattern and Call Type. These special dialing patterns, shown in the following table, are preassigned and require no administration. They can, however, be changed by the system administrator and probably will be changed for PBXs used outside North America.

Table 2-4. ARS Digit Analysis Default Translations

Dialed String	Trailing Digits Min.	Trailing Digits Max.	Pattern	Route Type
0	1	1		op
0	11	11		op
00	2	2		op
00	12	12		op
01	10	23		iop
010	3	3		iop
011	10	23		int
1	11	11		fnpa
10XXX	5	5		op
10XXX0	6	6		op
10XXX0	16	16		op
10XXX00	7	7		op
10XXX00	17	17		op
10XXX01	15	23		iop
10XXX010	8	8		iop
10XXX011	15	23		int
2	7	7	1	hnpa
3	7	7	1	hnpa
4	7	7	1	hnpa
5	7	7	1	hnpa
6	7	7	1	hnpa
7	7	7	1	hnpa
8	7	7	1	hnpa
9	7	7	1	hnpa
X11	3	3		svc

Legend: fnpa - foreign number plan area (10-digit call)
hnpa - home number plan area (7-digit call)
int - international
iop - international operator
op - operator
svc - service
X - any digit (0-9)

Normally, the `Route Pat` (routing pattern) field on the ARS Digit Analysis Table screen contains a routing pattern number (one through 254). However, this field may instead contain a Remote Home Numbering Plan Area (RHNPA) table number (r1 through r32). An RHNPA is simply a concentrator for up to 1,000 calls. Calls are routed to these tables by ARS/AAR Digit Analysis when an ARS Digit Analysis table points to an RHNPA table. The next three dialed digits (the code) are compared with the codes in the selected RHNPA table. Each code on the table is then mapped to a specific routing pattern number (one through 254).

The RHNPA tables allow up to 1000 codes to be handled by one entry in the Digit Analysis table.

In summary, Digit Analysis is merely a method of selecting a routing pattern. The routing pattern may be selected in two ways:

- It may be selected directly from the Digit Analysis table.
- The Digit Analysis table may first have to select an RHNPA table which will in turn select the routing pattern.

Routing Patterns

The digit translations performed on an ARS call by the Digit Analysis and RHNPA tables cause a specific Routing Pattern to be selected for the call. The Routing Patterns are numbered 1 through 254. More than one combination of dialed digits 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 central office, not by the system. By allowing the unassigned codes to be intercepted by the central office, 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 FRL assignment, not by intercept on the codes. FRLs are discussed elsewhere in this chapter.

The Routing Pattern applicable for a given call contains a list of up to six 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 switch to one of the public network offices serving the main switch. The digit manipulation necessary to route the call is controlled by the Subnet Trunking feature. (Subnet Trunking is discussed 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.

The following paragraphs describe how the switch decides what digits to outpulse in specific situations.

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 outpulsing 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 route. Five choices are available and are identified in translations by a Prefix Mark. Prefix Mark operations only apply to FNPA and HNPA call types. The values and meanings of the Prefix Marks are as follows:

- Prefix Mark 0 — Suppress a user-dialed Prefix digit 1 for 10-digit FNPA calls, but leave a user-dialed Prefix digit 1 for the following types of calls:
  - 10-digit calls that are not administered as FNPA or HNPA types in the ARS Routing Table.
  - 7-digit HNPA calls
- 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 ensure that all toll calls are 10-digit calls.
- Prefix Mark 4 — Always suppress a user-dialed Prefix digit 1.

NOTE:
This capability is required, for example, when routing ISDN-PRI calls to an AT&T 4ESS. If the prefix digit 1 were not suppressed, then the 4ESS would reject such calls.

Which of the five possible treatments of the 1 prefix digit should be administered on a given route is based on the characteristics of the distant office. Prefix Mark 0 is prevents the system from sending a 1 prefix digit for 10-digit FNPA calls. However, the system leaves a user-dialed prefix digit 1 for 7-digit HNPA calls and 10-digit calls that are not administered as FNPA or HNPA types in the ARS Routing Table.

Prefix Mark 1 causes the system to send a 1 prefix on all 10-digit FNPA 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 7- or 10-digit DDD public network calls. Requirements for outpulsing a 1 are specified via Prefix Marks and go into effect when the call accesses is outpulsed. Digit 1 outpulsing only applies to calls administered as "fnpa" or "hnpa" in the ARS Digit Analysis table.

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.

NPA deletion and insertion only applies to calls administered as "fnpa" or "hnpa" in the ARS Digit Analysis table.

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 an ARS 7- or 10-digit DDD call. An ARS call accessing a tandem trunk is another example of NPA insertion. If the call is a 7-digit ARS call, the system inserts the home NPA before sending the call to the tandem trunk. Therefore, all ARS calls accessing a tandem trunk are 10-digit calls. Whether or not the digit 1 is sent on a tandem call is determined by the prefix rules. 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 using 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. The U.S. ARS default dialing time-out is three seconds for the 0 and 1 prefix digits, but is 10 seconds for the called number. In other countries, the 3 second timer will apply to all numbers administered as valid dialed destinations that also happen to be substrings of a longer valid dialed destination. 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 (unless it is an ISDN trunk or it is suppressed on the trunk form.)

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 type of call. An ARS call can be processed via the RHNPA table. This is particularly useful on international calls, since the RHNPA table 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) with 0 as the attendant access code, require a three-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 home NPA 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.

These examples are for the U.S. and will differ in other countries.

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.

Up to 32 Toll Lists can be provided.

Up to 32 RHNPA tables, shared with AAR, can be provided.

Up to eight ARS and AAR partitions may be administered. Routing Patterns, toll lists, and RHNPA tables are shared between partitions.

ARS and AAR Digit Analysis tables together with the Toll Analysis table can have up to 2,000 entries.

Internal memory resources used for ARS Digit Analysis are shared by ARS, AAR, Digit Conversion, and Toll Analysis features. A `Percent Full` field on the ARS and AAR Digit Analysis screens indicate how many of these resources have been used.

If a customer changes ARS routing assignments, it is the customer's responsibility to notify the RSC network designer and the SCO technician of the changes in order to receive their continued support.

Interactions

The following features interact with the Automatic Route Selection feature:

- AAR

  ARS and AAR can access the same trunk groups and share the same Routing Patterns, toll lists, and RHNPA tables.

- 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.

- CAS

  A CAS Attendant may extend a call out of a Branch PBX by use of ARS.

The call is extended over an RLT by dialing the appropriate feature access code and number. The call is routed as determined by ARS administration at the Branch PBX.

- Toll Restriction

  Toll Restriction is 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. See Forced Entry of Account Codes feature for more information.

- GRS

  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.

- PCOL

  A trunk assigned as a PCOL may not be assigned to any ARS routing pattern.

- 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.

- CDR

  An ARS call using a trunk group marked for CDR 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 CDR.

  If CDR 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 CDR account code is to be dialed with an ARS call, it must be dialed before the ARS access code is dialed.

- Voice Terminal Display

  The voice terminal display shows the dialed digits (not outpulsed digits). The called-party shown on the display is that of the trunk group actually used. The miscellaneous call identification field on the display will show ARS.

An ISDN-BRI station may format these display fields differently, and the timing of display updates may be different.

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 (one to three digits)
- ARS Access Code 2 (one to three digits)
- ARS Digit Analysis Table (1 per PGN)
- Up to 32 RHNPA Tables
- Up to 32 Toll Lists
- FRLs — Assigned via COR to each originating facility.
- Up to 254 ARS Routing Patterns
- Trunk Groups to be used with ARS
- Whether or not the system returns dial tone after the ARS 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 (TN420C, TN744 support A-law). These additions are, however, cost effective when compared to the alternatives for call routing.

Optional ARS software and ARS/AAR Digit Conversion software is required.

Auto Start and Don’t Split

Description

The Auto Start feature allows the attendant to initiate a phone call by depressing any button on the dial keypad. If the attendant is on an active call and presses digits on the keypad, the system automatically splits the call and begins dialing the next call. When the Auto Start feature is enabled, the Start button is disabled. Also, end-to-end signaling is not allowed. That is, digits pressed on the keypad are always interpreted as dialing a call.

To allow end-to-end signaling, the Don’t split button is administered on consoles when the Auto Start feature is active. When the Don’t split is pressed, digits pressed on the console keypad are heard by the parties on the call. For example, use Don’t split if you need to send touch-tones to the far end to pick up answering machine messages.

To extend an active call to another extension, begin dialing the digits of the other extension. The active call is automatically put on hold. Once the called party answers, press Release to extend the call.

To send touch-tones on an active call, press Don’t Split. The call remains active. Press the keypad digits; the tones are sent to the far end. To deactivate Don’t Split, press Cancel.

Considerations

When Auto Start is enabled and an attendant dials an AAR number where the min and max in the AAR analysis table are not equal, then the attendant must dial a "#" after the digit string or the call will not be processed.

Interactions

When Don’t Split is active or if Auto Start is disabled, the attendant and any conference parties will hear touch tones on the connection when a digit on the keypad is pressed.

Release, Forced Release, and Cancel deactivate the Don’t Split feature.

The Auto Start feature is activated when the Don’t Split feature is activated and reactivated upon deactivation of Don’t Split.

If the system is using the Centralized Attendant Service (CAS) feature, Auto Start and Don’t Split cannot be activated.

Administration

This feature is administered on a system wide basis. The feature is administered on the system parameters miscellaneous form.

Hardware and Software Requirements

No special hardware or software is required for this feature.

Automatic Wakeup

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 five 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 five 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 two and one half 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 five-minute intervals. If the call is not completed after the three attempts, the system can leave a 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 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 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 five-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 five 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.
- System Reset — indicates that a system reset level 1 or system reset level 2 occurred while the system attempted to place the wakeup call. Calls affected by these conditions are treated as other wakeup attempts.

If a wakeup call was not completed because of a busy, no answer, ringing blockage, or system reset, the system attempts to place the call two more times at five-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 users 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 Manager I terminal (G1) or G3 Management Terminal, 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:
  - Request — A new wakeup call request has been made.
  - Change — The time has been changed on an existing wakeup call request.
  - Cancel — A wakeup request has been canceled. This event can be caused by a user request, a front desk request, or a room check out.
  - Move To — The wakeup request for this room has been moved to another room. This event occurs when the PMS sends a room change message.
  - Move From — The wakeup request for another room has been moved from that room to this room. This event occurs when the PMS sends a room change message.
  - Move-Cancel — A wakeup request from another room has replaced the request for this room. This event occurs when the PMS sends a room change request.
  - Swap — A room swap has occurred and at least one of the rooms had a wakeup request. Wakeup calls are swapped when a room swap is performed. A journal entry is made for each room. If the room has received a wakeup call as the result of the swap, the time of the call is provided in the entry. If the room has lost a wakeup call as the result of the swap (and has not received another), the

time is not present in the entry.

- — Completed — The wakeup call was completed successfully.
- — Not Completed — The wakeup call failed (not answered, busy, and so on.)
- — Skip — The wakeup call was skipped. This event occurs if the system time is advanced past the requested time of a wakeup call.

- 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 work load 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 Manager I terminal (G1) or G3 Management 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 one and one half 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 300 extension numbers may have active wakeup requests for any 15-minute interval.
- The maximum number of wakeup requests per system is 1,600. This maximum number is shared with Do Not Disturb requests.
- Wakeup requests may be entered from five 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 10 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

  Speech synthesis for Automatic Wakeup is only available for the U.S. Italian and United Kingdom boards do not allow Automatic Wakeup. attendant or to the specially administered Wakeup Messages Extension.
- Wakeup call attempts are not rerouted, forwarded, or sent to coverage.
- An RS-Alert lamp on the attendant console or a station would enable you to know when a reset system levels three to five has occurred. Resets at these levels will erase any AWU requests from the system. When the RS-Alert lamp is lit, check the AWU journal printer to see if there were any outstanding requests. During a reset system 3, the AWU journal printer will notify you of the reset only if there were wakeup requests in the system when the reset occurred. This report will include the time of the reset. Reset system 4 and reset system 5 are always journalled, since the system cannot determine whether or not there were wakeup requests prior to the reset.

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.

- 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 — Choose 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 MPDM and a port on a Digital Line circuit pack or an ADU and a port on a Data Line circuit pack.

No additional software is required.

Basic Call Management System (BCMS)

Description

Provides real-time and historical reports which will assist a customer in managing individual agents, ACD splits (hunt groups), and trunk groups. These reports are a subset of those available on the CMS adjunct. BCMS reports can be accessed and displayed on the Manager I terminal (G1), G3 Management Terminal or printed on demand on the printer associated with the Manager I terminal or G# Management 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 seven days.

The following reports are available with the BCMS:

- Real Time Reports
  - Split Status
  - System Status
- Historical Reports
  - Agent
  - Split
  - System
  - Trunk
  - VDN

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 (G1) or the G3 Management 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 Generic 3 — Basic Call Management System Operations, 555-230-703*.

BCMS Split Status Report

The BCMS Split Status Report provides the current (real-time) status and cumulative measurement data for those agents assigned to the split you specify. This report is reset at the beginning of the time interval (for example, hour or half-hour). The following screen shows the BCMS Split Status Report.

```
monitor bcms split 1                                                  Page 1 of 1
                      BCMS SPLIT (AGENT) STATUS
        Split: ##
   Split Name: xxxxxxxxxxxxxxx               Date: 12:59 pm THU APR 12, 1990
Calls Waiting:  xxx
  Oldest Call: x:xx
   xx=Staffed  xx=Avail  xx=ACD  xx=ACW  xx=AUX  xx=Extn  xx=OtherSplit
AGENT                EXT      STATE      TIME       ACD     EXT IN    EXT OUT
                                                   CALLS     CALLS    CALLS
xxxxxxxxxxxxxxx    xxxxx xxxxxxxxxx      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             12351 OtherSplit      11:58        0         0        0
             Note - Xs are used to show field length and are not displayed.
```

The BCMS Split Status Report fields are described below:

- **Split** — the split number specified with the command line.
- **Split Name** — the administered name of the split. This name usually describes the purpose or service of the split (for example, sales, customer service, and reservations). If no name exists, the split extension (for example, EXT 65222) will be displayed.
- **Calls Waiting** — the number of calls currently queued and calls ringing at an agent's phone. If any of the calls in the queue are Direct Agent calls, an asterisk will appear before the value in this field.
- **Oldest Call** — the number of minutes and seconds that the oldest call in queue has been waiting to be answered. This includes calls ringing at an agent's phone.
- **Staffed** — the number of agents currently logged into the split.
- **Avail** — the number of agents in this split currently available to receive an ACD call. In order to be counted as being available, agents must either be in the Auto-In or Manual-In work mode. If the agent is on another split's call or is performing After Call Work for another split, the agent is not considered available and will not be recorded here.
- **ACD** — the number of agents who are currently on an ACD call for this split. This value also includes direct agent calls and those agents who are currently on ACD calls that flowed in from another split. If an agent puts an ACD call on hold, but does not enter another state, the agent remains in the ACD state.
- **ACW** — the number of agents in this split who are currently in ACW mode

for this split. If an agent is in ACW mode for another split, the agent will be included in the "OtherSplit" state count for this split. Also, if an agent is on a call while in ACW mode, the agent will appear in the Extn Calls state count, and not in the ACW state count.

- **AUX** — the number of agents in this split who are currently in the AUX work mode for this split. If an agent is answering a call from another split or is in ACW work mode for another split, that agent is not considered in AUX work mode for this split and will not be included in this number. The agent will be included in the OtherSplit state count.
- **Extn** — the number of agents in this split who are currently on non-ACD calls. These non-ACD calls may be either incoming (direct to the extension) or outgoing (direct from the extension). Those agents receiving or making extension calls while in Avail, ACW, or AUX work mode will be recorded as being on extension calls.
- **OtherSplit** — the number of agents in this split who are currently answering a call from another split or are in ACW work mode for another split. This measurement is only applicable if the agents belong to multiple splits.

  NOTE:
  The time that an agent spends in OtherSplit is accounted for as AUX time in the BCMS Agent report.

- **AGENT** — the name of the agent. Generally, this will be the agent's first or last name. However, if no name is administered on the station form, this field will be left blank. When the field is blank, the data can be identified by the extension.
- **EXT** — the 2-, 3-, 4-, or 5-digit extension number for the agent.
- **STATE** — the current work state for the agent. Possible work states are Staffed, Avail, ACD, ACW, AUX, Extn, and OtherSplit. A blank work state field indicates that the agent is in the Unstaffed state.
- **TIME** — the 24-hour clock time that the agent entered this work state.
- **ACD CALLS** — the number of ACD calls that the agent has completed since the beginning of the current interval. This value includes any calls that flowed in from other splits.
- **EXT IN CALLS** — the number of non-ACD calls that the agent has received (incoming) since the beginning of the current interval.
- **EXT OUT CALLS** — the number of non-ACD calls that the agent has made (outgoing) since the beginning of the current interval.

BCMS System Status Report

The BCMS System Status Report provides current (real-time) status information for either all BCMS splits or selected BCMS splits. This report is reset at the beginning of the time interval (for example, hour or half-hour). The following screen shows the BCMS System Status Report. A completed call may span more than one time interval. ACD calls that are in-process (have not terminated) will be counted in the time interval in which they terminate. For example, if an ACD call begins in the 10:00 to 11:00 time interval, but terminates in the 11:00 to 12:00 time interval, the data for this call is counted in the 11:00 to 12:00 time interval.

```
monitor bcms system                                                 Page 1 or 1
                              BCMS SYSTEM STATUS
                                           Date:    1:00 pm SUN APR 8 1990
                                 AVG                    AVG                  AVG
                 CALLS  OLDEST   ANSW  AVAIL    #      ABAND   #    AVG    AFTER
SPLIT            WAIT    CALL   SPEED  AGENT  ABAND    TIME   ACD   TALK    CALL
xxxxxxxxxxxxxxx    xxx xxx:xx  xxx:xx  xxxx   xxxx   xxx:xx xxxx  xxx:xx xxx:xx
Services             3   1:03    0:45     0      3     0:30   20     2:30   1:25
Sales                5   0:33    0:15     0     11     0:45   36     1:32   0:35
             Note - Xs are used to show field length and are not displayed.
```

The BCMS System Status Report fields are described below:

- **SPLIT** — the name of the split (for example, sales, service, and help line). If no name exists, the split extension (for example, EXT 12345) will be displayed.
- **CALLS WAIT** — the number of calls in the split's queue that are currently waiting to be answered and calls ringing at an agent's phone. If any of the calls in the queue are Direct Agent Calls, an asterisk appears before this field.
- **OLDEST CALL** — the number of minutes and seconds the oldest call in queue has been waiting to be answered. This includes calls ringing at an agent's phone.
- **AVG ANSW SPEED** — the average amount of time it takes before the calls are being answered. This value includes time waiting in the queue and time ringing at the agent's voice terminal.
- **AVAIL AGENT** — the number of agents in this split who are currently available to receive an ACD call directed to this split.
- **# ABAND** — the total number of ACD calls that have hung up while waiting to be answered. This includes those calls which have abandoned while in queue or while ringing. Calls that are not queued (for example, because the queue is full, the caller receives a forced first announcement and abandons during the announcement, or no agents are staffed) will not be counted as abandoned.
- **AVG ABAND TIME** — the average time before an ACD call abandons. This does not include any time spent in another split's queue before intraflowing to this split. This value does not include time spent listening to a forced first announcement.
- **# ACD** — the number of ACD calls completed during the current interval. This number also includes those calls that flow in from other splits.

- **AVG TALK TIME** — the average duration of ACD calls for each split. This calculation includes time each agent spent talking and on hold, but does not include ring time at an agent's voice terminal.
- **AVG AFTER CALL** — the average ACW time for ACD calls handled by the split in this time interval. This average includes ACD calls that have no ACW time. However, ExtnIn and ExtnOut type calls are not included in the average.

BCMS Agent Report

The BCMS Agent Report provides traffic information for the specified agent. Depending on specifics from the command line, the information may be displayed as either a time interval or a daily summary. The following screens show the BCMS Agent Time Interval Report and the BCMS Agent Daily Summary Report.

```
list bcms agent 34 time 08:00 14:00                                     Page 1

                             BCMS AGENT REPORT
Agent: 34                                  Date 6:05 pm SUN APR 8, 1990
                #     AVG    AVG     TOTAL   TOTAL   #     AVG    TOTAL
               ACD    TALK  AFTER    AVAIL   AUX     EXTN   EXTN   TIME
TIME           CALLS  TIME  CALL     TIME    TIME    CALLS  TIME  STAFFED
xx:xx-xx:xx     xxxx xxx:xx xxx:xx xxxx:xx xxxx:xx  xxxx  xxx:xx xxxx: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   60:00
11:00-12:00       12   1:21    :47   12:56    9:21     3    7:21   60:00
12:00-13:00        8    :55    :51   10:13   22:11     0     :00   60:00
13:00-14:00       16   1:10   1:04    9:12   14:31     0     :00   60:00
-----------------------------------------------------------------------
SUMMARY           74   1:28    :54   62:55   56:43     6    2:26  360:00
             Note - Xs are used to show field length and are not displayed.
```

```
list bcms agent 1 day 09/11 09/15                                         Page 1
                             BCMS AGENT REPORT
                                             Date 6:05 pm FRI SEP 15, 1989
Agent: ACD-1
                 #      AVG    AVG     TOTAL    TOTAL    #      AVG    TOTAL
                ACD     TALK  AFTER    AVAIL    AUX     EXTN   EXTN    TIME
DAY            CALLS    TIME  CALL     TIME     TIME    CALLS  TIME   STAFFED
xx/xx/xx        xxxx xxx:xx xxx:xx xxxx:xx xxxx:xx   xxxx   xxx:xx xxxx:xx
09/11/89          38   1:28    :54   30:34   10:40      3     2:26  150:00
09/12/89          21   1:31    :32   27:21   11:21      2     2:56  170:00
09/13/89          22   1:11    :21   25:25   08:51      1     3:01  120:00
09/14/89          35   1:43    :31   29:48   13:12      0     0:00  149:00
09/15/89          30   1:21    :45   28:21   12:13      1     2:11  137:00
-----------------------------------------------------------------------
SUMMARY          146   1:26    :36  141:29   56:17      7     1:06  726:00
             Note - Xs are used to show field length and are not displayed.
```

The BCMS Agent Report fields are described below:

- **AGENT** — the name of the agent. If no name is administered, the agent's extension will be displayed in the form "EXT 65432."
- **TIME/DAY** — the time or day interval specified in the command line.
- **# ACD CALLS** — the number of ACD calls answered by this agent for all splits during the reporting interval. This value includes calls that flowed in from other splits.
- **AVG TALK TIME** — the average duration of ACD calls for all splits the agent was logged into. This value includes time spent talking and on hold, but does not include ring time at the agent's voice terminal. If an agent puts a caller on hold, but does not enter another state, BCMS does not receive a message. As a result, the time on hold is credited to talk time until the agent enters another state.
- **AVG AFTER CALL** — the average amount of time per ACD call that the agent spent in the ACW work state for all splits during the reporting interval. This does not include time spent on ExtnIn or ExtnOut calls while in ACW.
- **TOTAL AVAIL TIME** — the sum of the time that the agent:
  - was in Auto-In or Manual-In work modes for at least one split
  - was not in ACW in any split
  - was not on any call
- **TOTAL AUX TIME** — the sum of the time that the agent has the AUX button pressed and is not doing anything else for any of the other splits (that is, the sum of the time that the agent is in AUX work mode for all splits). This value does not include time the agent spent on a direct call or in Manual-In, Auto-In, or ACW mode for another split.
- **# EXTN CALLS** — the total number of non-ACD incoming and outgoing calls for this agent during the reporting interval. Only those non-ACD calls

that are originated/received while the agent is logged into at least one split will be counted.

- **AVG EXTN TIME** — the average amount of time that the agent spent on non-ACD calls while logged into at least one split during the reporting interval.
- **TOTAL TIME STAFFED** — the total time that the agent spent logged into at least one split during the reporting interval. Staff time is clocked for an agent who is in multiple splits as long as the agent is logged into any split. Concurrent times for each split are not summed.
- **SUMMARY** — the total of each of the columns that do not contain averages. Columns that do contain averages are the total time divided by the number of calls.

BCMS Split Report

The BCMS Split Report provides traffic information for the specified split number. Depending on specifics from the command line, the information may be displayed as either a time interval or a daily summary. The following screens show the BCMS Split Time Interval Report and the BCMS Split Daily Summary Report.

```
list bcms split 1 time 08:00 10:00                                          Page 1
                              BCMS SPLIT REPORT
     Split: 1
Split Name: xxxxxxxxxxxxxxx                   Date 18:00 pm FRI APR 13, 1990
                   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 xxxxx xx:xx xxxx xx:xx xx:xx xx:xx  xxxx  xxxx xxxx: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 to show field length and are not displayed.
```

```
list bcms split 1 day 04/12 04/12                                         Page 1
                              BCMS SPLIT REPORT
     Split: 1
Split Name: xxxxxxxxxxxxxxx                  Date 18:00 pm FRI APR 13, 1990
                   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    xxxxx xx:xx xxxx xx:xx xx:xx xx:xx  xxxx  xxxx xxxx:xx xxx.x
04/12/90       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 to show field length and are not displayed.
```

The BCMS Split Report fields are described below:

- **SPLIT** — the split number specified with the command line.
- **SPLIT NAME** — displays the name that is administered for this split number. If no name exists, then the split extension (for example, EXT 65432) will be displayed.
- **TIME/DAY** — the time or day interval specified in the command line.
- **# ACD** — the number of ACD calls completed for this split during the current interval. This number also includes calls that flowed in from other splits.
- **AVG ANSWER SPEED** — the average amount of time ACD calls spent in queue and ringing at an agent's station before being answered during the reporting interval. Calls that flowed in will not have queue time from the previous split included in this average. This value does not include time listening to a forced first announcement.
- **# ABAND** — the total number of ACD calls that have hung up while waiting to be answered. This value includes those calls which have abandoned while in queue or while ringing. Calls that are not queued (because the queue is full, the caller receives a forced first announcement and abandons during the announcement, or no agents are staffed) will not be counted as abandoned.
- **AVG ABAND TIME** — the average time before an ACD call abandons. This value does not include any time spent in another split's queue before flowing in to this split. This value does not include time listening to a forced first announcement.
- **AVG TALK TIME** — the average duration of ACD calls for each split. This includes time spent talking and on hold. The calculation does not include ring time at an agent's voice terminal.
- **AVG AFTER CALL** — the average amount of time that the agents in this split spent in ACW mode during the reporting interval. The calculation does not include time spent on Ext In or Ext Out calls while in ACW.
- **# FLOW IN** — the total number of calls that this split received as a coverage point (intraflowed) from another BCMS-measured split, or are call

forwarded (interflowed) to this split during the reporting interval. This total does not include calls which are interflowed from a remote switch by means of the Look Ahead Interflow feature.

- # **FLOW OUT** — the total number of calls queued to this split that were
  - — successfully sent to the split's coverage point
  - — forwarded-out via call forwarding
  - — answered via the Call Pickup feature
  - — forwarded-out via Look Ahead Interflow
- **AUX TIME** — the total time that logged-in agents in this split were unavailable to receive calls during the reporting interval. This value also includes time agents put calls on hold and did not make another state selection. This value does not include the time agents spent on another split's calls or in ACW for another split.
- **AVG STAFF** — the average number of agents who were logged into this split (staffed) during the reporting interval.
- **SUMMARY** — for those columns that specify averages the summary will also be an average for the entire reporting interval. For the # ACD, # ABAND, FLOW IN, FLOW OUT, and AUX TIME columns, the summary will be the sum of individual time intervals or specified days.

BCMS System Report

The BCMS System Report provides traffic measurement information for all of the BCMS splits. Depending on specifics from the command line, the information may be displayed as either a time interval or daily summary. The following screens show the BCMS System Time Interval Report and the BCMS Daily System Report.

```
list bcms system time 12:00                                                 Page 1
                            BCMS SYSTEM REPORT
Time: 12:00-12:30                                    Date 12:55 pm THU APR 12, 1990
                   AVG          AVG  AVG    AVG     #     #
                #  ANSW   #    ABAND TALK  AFTER  FLOW  FLOW  AUX    AVG
SPLIT          ACD SPEED ABAND TIME  TIME  CALL    IN    OUT  TIME   STAFF
xxxxxxxxxxx xxxxx xx:xx xxxx xx:xx xx:xx xx:xx  xxxx  xxxx xxxx:xx xxx.x
services        32   :25    4   :32  5:15   :30     3     5    3:30   4.0
sales            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 to show field length and are not displayed.
```

NOTE:
Because of space limitations, the calculation for AVAIL TIME is not included on either the Split Report or the System Report. However, AVAIL TIME is displayed on the Agent Report. It can also be calculated as

follows:

$$AVAIL = 60 \times (AVG\ STAFF) - (\#ACD \times AVG\ TALK\ TIME + AVG\ AFTER\ CALL + A$$

The *60* represents a one-hour time interval. If you used a half-hour time interval, the equation would contain *30* instead of *60*.

The result of this calculation is only an approximation since calls can span intervals.

```
list bcms system day 04/12/90                                                Page 1
                                BCMS SYSTEM REPORT
Day:  4/12/90                                    Date 12:55 pm THU APR 12, 1990
                   AVG          AVG  AVG    AVG    #     #
                #  ANSW   #    ABAND TALK  AFTER  FLOW  FLOW  AUX    AVG
SPLIT         ACD SPEED ABAND TIME  TIME  CALL    IN    OUT   TIME  STAFF
xxxxxxxxxxx xxxxx xx:xx xxxx xx:xx xx:xx xx:xx  xxxx  xxxx xxxx:xx xxx.x
services       32   :25    4   :32  5:15   :30     3     5    3:30   4.0
sales           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 to show field length and are not displayed.
```

The BCMS System Report fields are described below:

- **TIME/DAY** — the time or day qualifier, which is entered on the command line and indicates the type of report.
- **SPLIT** — the name of the split (for example, sales, customer service, or reservations). If no name exists, the split extension (such as EXT 12345) will be displayed.
- **# ACD** — the number of ACD calls (inbound and outbound) that ended in each split during the reporting interval.
- **AVG ANSWER SPEED** — the average amount of time ACD calls spent in queue and ringing at an agent's station before being answered during the reporting interval. This value includes time in queue and time ringing at an agent's station. Calls that flowed in will not have queue time from the previous split included in this average. This value does not include time spent listening to a forced first announcement.
- **# ABAND** — the total number of ACD calls that have hung up while waiting to be answered. This value includes those calls which have abandoned while in queue or while ringing. Calls that are not queued (for example, because the queue is full, the caller receives a forced first announcement and abandons during the announcement, or no agents are

staffed) will not be counted as abandoned.

- **AVG ABAND TIME** — the average time before an ACD call abandoned. This value does not include any time spent in another split's queue before flowing into this split. This value does not include time spent listening to a forced first announcement.
- **AVG TALK TIME** — the average duration of ACD calls for each split. This value includes time spent talking and on hold, but does not include ring time at an agent's voice terminal.
- **AVG AFTER CALL** — the average amount of time that the agents spent in ACW during the reporting interval. The calculation does not include time spent on ExtnIn or ExtnOut calls while in ACW.
- **# FLOW IN** — the total number of calls that this split received as a coverage point (intraflowed) from another BCMS-measured split, or are call forwarded (interflowed) to this split during the reporting interval. This total does not include calls which are interflowed from a remote switch by means of the Look Ahead Interflow feature.
- **# FLOW OUT** — the total number of calls queued to this split that were
  - successfully sent to its own coverage point
  - forwarded-out via call forwarding
  - answered via the Call Pickup feature
  - forwarded-out via Look Ahead Interflow
- **AUX TIME** — the total amount of time that the agents within this split spent in AUX work state during the reporting interval. This value also includes time an agent puts a call on hold and does not make another state selection. This value does not include the time agents spent on a call from another split or in ACW for another split. If an agent was in AUX, for this split, and in ACW for another split, then the ACW time is not counted against AUX.
- **AVG STAFF** — the average number of agents who were logged in (staffed) for this split during the reporting interval.
- **SUMMARY** — the sum of the total columns (# ACD, # ABAND, AUX TIME, # FLOW IN, and # FLOW OUT) and average for the columns which are averages (ANSW SPEED, TALK TIME, AFTER CALL, ABAND TIME, and AVG STAFF).

BCMS Trunk Group Report

The BCMS Trunk Group Report gives statistical information for all BCMS trunk groups. The BCMS Trunk Group Report may be used by the ACD administrator/manager to monitor use of the trunk group and to determine the optimal number of trunks for the trunk group. Depending on specifics from the command line, the information may be displayed as either a time interval or a daily summary. The following screens show the BCMS Trunk Group Time Interval Report and the BCMS Trunk Group Daily Report.

```
list bcms trunk 1 time 10:00 10:30                                      Page 1
                             BCMS TRUNK GROUP REPORT
     Trunk Group: 1
Trunk Group Name: xxxxxxxxxxxxxxx
Number of Trunks: xx                         Date: 12:59 pm THU APR 12, 1990
                   INCOMING           |         OUTGOING          |% ALL % TIME
TIME         CALLS ABAND   TIME     CCS| CALLS  COMP   TIME     CCS| BUSY  MAINT
xx:xx-xx:xx xxxxx  xxxx xxx:xx xxxx.xx  xxxxx  xxxx xxx:xx xxxx.xx   xx      xx
10:00-11:00    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 to show field length and are not displayed.
```

```
list bcms trunk 1 day 4/11 4/12                                         Page 1
                             BCMS TRUNK GROUP REPORT
     Trunk Group: 1
Trunk Group Name: xxxxxxxxxxxxxxx
Number of Trunks: xx                         Date: 12:59 pm THU APR 12, 1990
                   INCOMING           |         OUTGOING          |% ALL % TIME
DAY          CALLS ABAND   TIME     CCS| CALLS  COMP   TIME     CCS| BUSY  MAINT
xx/xx/xx     xxxxx  xxxx xxx:xx xxxx.xx  xxxxx  xxxx xxx:xx xxxx.xx   xx      xx
04/11/90        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 to show field length and are not displayed.
```

The BCMS Trunk Group Report fields are described below:

- **Trunk Group** — the trunk group number specified with the command line.
- **Trunk Group Name** — the name that is administered for this trunk group. If no name is administered, then this field is displayed as blank.
- **Number of Trunks** — the number of individual trunks in the trunk group at the end of the first interval being reported.
- **TIME/DAY** — the time or day interval specified in the command line.
- **INCOMING CALLS** — the total number of incoming calls carried by this trunk group.
- **INCOMING ABAND** — the number of incoming calls that queued to ACD splits, then abandoned (without being answered by a staffed agent within this split) during the reporting interval. Calls that cannot queue (for example, queue full, or calls that receive a busy signal from the CO because there aren't any available trunks) are not included in the INCOMING ABAND number.
- **INCOMING TIME** — the average holding time for incoming calls to this trunk group during the specified reporting interval. Holding time is defined as the length of time in minutes and seconds that a facility is used during

a call.

- **INCOMING CCS** — the total holding time (usage) for incoming calls to the trunk group during the specified reporting interval. The units are expressed in hundred call seconds (CCS).
- **OUTGOING CALLS** — the total number of outgoing calls for this trunk group during the specified reporting interval.
- **OUTGOING COMP** — the total number of outgoing calls that were placed over this trunk group and answered during the specified reporting interval. Completion is determined by either return of network answer supervision, or a call that lasts longer than the answer supervision time-out parameter; whichever occurs first.
- **OUTGOING TIME** — the average holding time for outgoing calls during the specified reporting interval.
- **OUTGOING CCS** — the total holding time for outgoing calls from this trunk group. The units are expressed in hundred call seconds (CCS).
- **% ALL BUSY** — the percentage of time that all the trunks in this trunk group were busy.
- **% TIME MAINT** — the percentage of time that one or more trunks have been busied-out for maintenance purposes.

BCMS VDN Report

The BCMS VDN Report provides statistical information for the specified VDN. Depending on specifics from the command line, the information may be displayed as either a time interval or a daily summary. The following screens show the VDN Time Interval Report and the VDN Daily Summary Report.

```
list bcms vdn 12345 time 08:00 9:00                                         Page 1
                      VECTOR DIRECTORY NUMBER REPORT
   VDN Ext: 12345
  VDN Name: xxxxxxxxxxxxxxx                   Date 18:00 pm FRI APR 13, 1990
                               AVG            AVG   AVG
             TOTAL     NUM   TIME TO   NUM   ABAND  TALK   FLOW  OTHER
TIME        ATTEMPTS   ANS   CONNECT  ABAND   TIME  TIME   OUT   CALLS
xx:xx-xx:xx    xxxxx  xxxx    xx:xx   xx:xx  xx:xx  xx:xx  xxxx   xxxx
 8:00- 9:00       79    50      :39       5    :45   2:30     0     24
------------------------------------------------------------------------
SUMMARY           79    50      :39       5    :45   2:30     0     24
             Note - Xs are used to show field length and are not displayed.
```

```
list bcms vdn 12345 day 4/13                                              Page 1
                      VECTOR DIRECTORY NUMBER REPORT
   VDN Ext: 12345
  VDN Name: xxxxxxxxxxxxxxx                       Date 18:00 pm FRI APR 13, 1990
                               AVG           AVG   AVG
             TOTAL    NUM    TIME TO   NUM   ABAND  TALK   FLOW  OTHER
DAY         ATTEMPTS   ANS   CONNECT  ABAND   TIME  TIME   OUT   CALLS
xx/xx/xx       xxxxx  xxxx    xx:xx   xx:xx  xx:xx  xx:xx  xxxx   xxxx
 4/13/90          79    50      :39       5    :45   2:30     0     24
-----------------------------------------------------------------------
SUMMARY           79    50      :39       5    :45   2:30     0     24
             Note - Xs are used to show field length and are not displayed.
```

The BCMS VDN Report fields are described below:

- **VDN EXT** — the VDN specified with the command line.
- **VDN NAME** — the name that is administered for this VDN. If no name exists, then the VDN extension (for example, EXT 64532) will be displayed.
- **TIME/DAY** — the time or day interval specified in the command line.
- **TOTAL ATTEMPTS** — the total number of completed calls that used the VDN during the current interval.
- **NUM ANS** — the total number of calls to the VDN that ended in the specified interval and were answered as a result of a queue to main or check backup split step.
- **AVG TIME TO CONNECT** — the average time that calls spend in a vector before being connected as an ACD call to an agent (for example, via a queue to main split or check backup step) during the current interval. This includes queue time and time ringing at an agent's station.
- **NUM ABAND** — the total number of calls that have abandoned from the VDN before being answered or outflowed to another position during the current interval. This value includes calls that abandoned while in vector processing or while ringing an agent. Calls that abandoned immediately after the agent answered are recorded as NUM ANS.
- **AVG ABAND TIME** — the average time calls spent waiting in this VDN before being abandoned by the caller during the the current interval.
- **AVG TALK TIME** — the average duration of calls (from answer to disconnect) for this VDN during the current interval. This includes time spent talking and on hold. The calculation does not include ring time at an agent's voice terminal.
- **FLOW OUT** — the total number of calls that were advanced to another position via a successful route-to or messaging split command. This includes the following:
  - — adjunct routing
  - — calls routed to another VDN

- — calls answered via the Call Pickup feature
- — calls answered by an attendant (through a route-to command)

FLOW OUT does not include calls that encounter a **goto vector** command.

Once a call outflows, the system does not take further measurements on the call for this VDN. As a result, if an outflowed call later abandons, it will not be recorded in NUM ABAND for this VDN.

- **OTHER CALLS** — the total number of calls that were forced busy or forced disconnect during the current interval. This value does not include abandoned calls.
- **SUMMARY** — for those columns that specify averages, the summary will also be an average for the entire reporting interval. The total of each of the columns that do not contain averages.

Commands

The following list shows the commands that may be used at the Manager I terminal (G1) or the G3 Management Terminal to generate BCMS reports:

- monitor bcms split
- monitor bcms system
- list bcms agent
- list bcms split
- list bcms system
- list bcms trunk
- list bcms vdn

The **monitor** commands display real-time status reports for agents and splits on the Manager I terminal (G1) or the G# Management Terminal. When a status report is displayed on the Manager I terminal (G1) or the G3 Management 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, trunk groups, and VDNs.

Report Scheduler and System Printer

The report scheduler allows the System Manager to use the Manager I terminal (G1) or the G3 Management Terminal to schedule BCMS reports as well as other reports, lists, and so on, 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 Generic 3 — System*

Management, 555-230-500 and *DEFINITY® Communications System Generic 1 and Generic 3i — System Reports*, *555-204-510*.

Considerations

BCMS provides a set of internal switch measurement reports for telemarketing s or customer service s. These reports can help in managing ACD splits (hunt groups) without the need for an adjunct CMS.

The maximum number of measured agents for the BCMS feature is limited to 30 (G1.1) or 200 (G3i). An agent can be a member of up to three 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 (G1.1) or 32 (G3i).

The maximum number of internally measured splits is limited to 30 (G1.1) or 99 (G3i). If a split is assigned more than 30 agents in G1.1, it cannot be measured internally. If a split is assigned more than 200 agents in G3i, it cannot be measured internally.

A maximum of 25 time intervals are allocated for storing data. A time interval can be either a one-hour or a one-half hour interval.

A maximum of seven 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 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 (G1) or G3 Management Terminal is connected to the EPN and the fiber link goes down, the Manager I terminal (G1) or G3 Management 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 BCMS 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 BCMS 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 from an extension will be treated like new incoming calls to that split. INFLOW and OUTFLOW counts will not be affected.

  If a split’s calls are forwarded, inflow and outflow apply. An agent’s call forwarding will not forward ACD calls.

- 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 BCMS Split report.

- Call Vectoring (G3i)

  With Call Vectoring, calls can be queued to up to three splits. ACD call count will be pegged to the answering split. Abandoned calls, inflows, and disconnects will be credited to the first (primary) split. Inflow and outflow cannot occur with vector controlled calls.

- 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 conferenced 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 and Generic 3i 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 sp-link** 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 sp-link** 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 sp-link** 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 PDMs connected to a digital port, or 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 two-lamp button. It does not require the use of 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 two-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 1,600 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 primary voice terminal has three call appearances, then a bridging voice terminal should have three bridged call appearances of that primary 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, UCD Group, Call Coverage Answer Group, or DDC 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).

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 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.

- COR

  The COR 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 both the principal's appearance button and the bridged party's 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).

- LWC

  A LWC 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 LWC, the message is left for the primary extension, even if the call was answered at a bridged call appearance.

  LWC messages left by the primary user can be canceled by a bridged appearance user (for example, the secretary can cancel a LWC message left by the boss).

- PCOL

  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.

- 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 is 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

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 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 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 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 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 station 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 two-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, and so on (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 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 two-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 system technician busyout command, and so on.

A maximum of 1,600 bridged call appearances (any mix of analog and multi-appearance) is allowed per system (G1.1 or G3i).

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, UCD Group, Call Coverage Answer Group, or DDC 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 description 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 (COS, COR, FRL)

  The COS and COR (including restrictions and FRL) 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 would be 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 members 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, and so on), 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 pick up 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 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).

- LWC

  A LWC 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 LWC, the message is left for the primary terminal, even if the call was answered at a bridged call appearance.

  LWC messages left by the primary terminal user can be canceled by a bridged appearance user (for example, the secretary can cancel a LWC message left by the boss).

- PCOL

  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 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, *TERMINAL-BASED* feature).

- CDR

  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.

- 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

Description

Allows attendants and specified multi-appearance voice terminal users to make test calls to trunks, voice terminals, and hunt groups DDC and 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 two-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 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 (two-second burst of 440 Hz tone) to let them know that the attendant is bridging onto the call. A half-second burst of warning tone is repeated every 15 seconds, as long as the attendant is bridged onto the call. If a country requires a different tone than 440 Hz, the attendant should use the intrusion feature rather than busy verification to make these test calls.

- 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 (two-second burst of 440 Hz tone) to let them know that the originator of the busy verification is bridging onto the call. A half-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 reorder 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 the 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:
  - DID
  - CO
  - FX
  - WATS
  - APLT
  - Tie
  - Remote Access
  - 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.

With G3i, busy verification may be activated for a phantom extension that is "administered without hardware." In this case, an Out of Service indication is provided.

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 which 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 which 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

Description

Call-By-Call Service Selection allows a single ISDN-PRI trunk group to carry calls to many services or facilities (such as a SDN, MEGACOM telecommunications service, MEGACOM 800 service, and so on) 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 AAR, ARS, and 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 the following figure. 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 in the U.S. 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 five.
- MEGACOM Service — Provides an AT&T service that provides unbanded long distance services using special access (PBX to 4ESS switch) from an AT&T node.
- MEGACOM 800 Service — Provides an AT&T service that provides unbanded 800 service using special egress (4ESS switch to PBX) from an

AT&T node.

- Network Operator — Provides access to the network operator.
- 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.
- International 800 (G3i) — Allows a subscriber to receive international calls without a charge to the call originating party. The subscriber of the service is charged for the calls.
- MULTIQUEST® Telecommunications Service service between callers at switched-access locations and service providers directly connected to the AT&T switched network. Callers access the service providers by dialing a 700 number.
- Other User-Defined Services — New service types can be assigned as they are developed and defined.

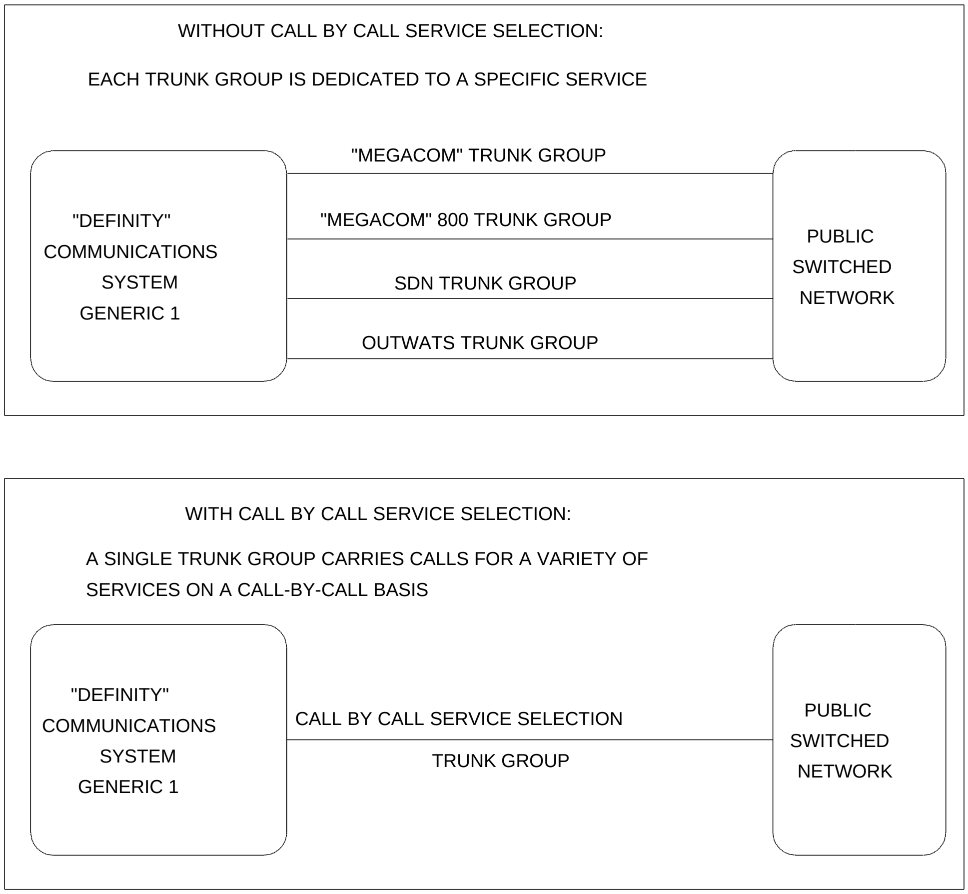


Figure 2-5. Call-By-Call Service Selection Example

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 10 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 10.

If a UAP 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 UAP 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 UAPs. The customer can assign either fixed or scheduled allocation plans for each Call-By-Call Service Selection trunk group, as described below (see the following screens for an example of the screen used to schedule UAPs the actual UAP screen):

- 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    __:__   _    __:__   _    __:__   _    __:__   _     __:__   _
```

```
                    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
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
____________   __    __    ____________   __    __    ____________   __    __
```

System administration allows the customer to have anything from a simple field 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 UAPs:

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 UAPs.

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 that contains this table follows.

```
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 CPN? n
                                                     Send Name? n
  Service/    Called     Called        Del       Insert      CPN/BN     Night
  Feature      Len       Number                                          Serv
_____________ ___  _______________  ___  ______________ __________  _______
_____________ ___  _______________  ___  ______________ __________  _______
_____________ ___  _______________  ___  ______________ __________  _______
_____________ ___  _______________  ___  ______________ __________  _______
_____________ ___  _______________  ___  ______________ __________  _______
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_____________ ___  _______________  ___  ______________ __________  _______
_____________ ___  _______________  ___  ______________ __________  _______
_____________ ___  _______________  ___  ______________ __________  _______
```

As shown in Screen 2-19, 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 discussed 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 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 one 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 (G1.1)** — Specifies your preference of a Station Identification (SID) or Automatic Number Identification (ANI) 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.

— **CPN/BN (G3i)** — Specifies your preference of a SID or ANI Calling Party Number (CPN) or Billing Number (BN) request for this type of call. A blank or "none" indicates that the switch will not request either CPN or BN for any incoming calls of this type. Allowable entries are BN only, prefer BN but accept CPN, CPN only, and prefer CPN but accept BN.

— **Night Serv** — Specifies a night service extension per Service/Feature. An entry other than blank overrides the night destination entry on Page one 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 (G1.1) or CPN/BN (G3i), 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:

- AAR

Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by AAR.

- ARS

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by ARS.

- GRS

  Call-By-Call Service Selection uses the same routing tables and routing preferences that are used by GRS.

- CDR

  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 CDR. CDR 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, CDR will record the three-digit IXC code. CDR may not record the IXC code properly if the dialed code format differs from the U.S. IXC code formats.

  When a Call-By-Call Service Selection call is rejected because of a trunk group usage allocation plan, CDR 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 UAPs are required
  - UAPs

  - — UAP Assignment Schedule
  - — Group Member Assignments

- AAR/ARS Routing Patterns — Routing Patterns can be administered to include a Network Specific Facility and IXC.
- Network Specific Facilities Encoding — New Network Specific Facilities can be added as needed by the system technician.

Hardware and Software Requirements

A TN767 DS1 circuit pack (TN464B/C/D support A-law) 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 (TN464B/C/D support A-law).

Optional ARS software are 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
- UCD group
- DDC group
- TEG
- PCOL group
- ACD split

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.
- AUDIX

Multiple Coverage Paths

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 does not meet any of the redirection criteria 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 call-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 (discussed 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 (discussed 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 (discussed next).

- Don't Answer

  Redirects calls to coverage if unanswered during the assigned Don't Answer Interval (1 to 99 ringing cycles). 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

  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 (discussed 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 discussed 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 discussed 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, discussed 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.

All calls extended by the attendant are treated as external.

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 LWC, 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 LWC button, a ‘‘call me’’ message is left for the principal. 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 (ICI) lamp.

- Coverage 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 one. 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 Coverage Subsequent Redirection No Answer Interval (1 to 99 ringing cycles). 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 below 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 200 Coverage Answer groups with up to eight voice terminals in each group.

Up to 600 coverage paths can be established. Each coverage path can have one, two, or three coverage points.

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

  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 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.

- CAS

  If an incoming CAS call is directed to a hunt group, the call will not redirect to the hunt group's coverage path.

- COR and Controlled Restrictions

  Users who may normally be restricted from receiving calls can still receive calls directed to them via Call Coverage.

- DDC and 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.

- 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.

- LWC

  Call Coverage can be used with or without LWC. However, the two features complement each other. When a covering user activates LWC 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 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. 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.

  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.

- Transfer

  The Transfer feature interacts with Call Coverage as shown in the following table:

Transfer and Coverage Interactions

Source	Transfer Initiator	Destination	Coverage Type
External	Local Station	Local Station	Internal
	Local Station	Remote Station	Internal
	Remote Station	Local Station	Internal
	Remote Station	Remote Station	Internal
	Attendant	Local Station	External
	Attendant	Remote Station	External
Internal	Local Station	Local Station	Internal
	Local Station	Remote Station	Internal
	Remote Station	Local Station	Internal
	Remote Station	Remote Station	Internal
	Attendant	Local Station	External
	Attendant	Remote Station	External

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)
- Cover Answer Groups
- Don’t Answer Interval and Coverage Subsequent Redirection No Answer Interval

  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 Coverage Subsequent Redirection No Answer Interval 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
- 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.)
- 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 Detail Recording (CDR)

Description

Records detailed call information on all incoming and outgoing calls on specified trunk groups and extensions administered for intra-switch recording and sends this information to a CDR output device. The CDR 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 CDR. CDR 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. This includes the unavailable incoming or outgoing trunks due to trunk usage allocation for ISDN Call-By-Call Service Selection trunks and incoming calls rejected by the switch due to NSF mismatch.
- Calls made using the Loudspeaker Paging Access and Code Calling Access features.
- Calls involving an auxiliary trunk.
- TSCs involving a trunk.
- Internal, direct calls that are originated by an extension optioned for intra-switch CDR, or that have a dialed number which is optioned for intra-switch CDR.

NOTE:
If an extension optioned for intra-switch CDR is neither the originator of the call nor the dialed number of the call, no CDR record will be output even though the extension might be a party on the call (via Call Pickup, Call Forwarding, and so on).

Intra-switch CDR is an administerable option that allows CDR records to be generated for some internal calls.

You have the option of turning off CDR generation for incoming calls, specific trunk group(s), intra-switch calls, NCA-TSCs, CA-TSCs, and ineffective call attempts via administration procedures.

Splitting of CDR Records

Since international calls are expensive and difficult to set up, the call transfer feature is commonly used among different parties to optimize the use of the connections. It is, therefore, important to provide accurate cost allocation data for each leg of the call.

An administrable option is provided that creates a new call record from calls that are transferred, attendant handled or conferenced.

CDR Privacy

To ensure the privacy of calls, an administrable option is provided that allows up to 7 digits of the "Dialed Number" to be blanked from the CDR record. Certain countries have requirements that specify a certain number of digits must be blanked from every call.

CDR Data Formats

This part covers the two formats sent to the CDR output device, call detail and date record formats.

Call Detail Record Format

The call detail record format provides detailed information concerning an incoming call, an outgoing call, or an intra-switch call. Call detail records are generated during call processing and are sent to the CDR output device in ASCII.

The following list describes the CDR 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 CDR output device such as TELESEER® Call Detail Recorder (CDR) 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.

The CDR output for intra-switch CDR records contain only the time, duration, condition code, dialed number, calling number, and optionally who disconnected first information (refer to the FRL field description) fields.

- Access Code Dialed (up to three or four digits [24-word formats])

  This field is used only for outgoing calls. This field can be the ARS access code, AAR access code, or the access code of a specific trunk group. This field does not exist in the ISDN 18-word record formats.

  Intra-switch CDR will not output this field.

- Access Code Used (up to three or four digits [24-word formats])

  This field is used only for outgoing calls. This field is used 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).

Intra-switch CDR will not output this field.

- Account Code (up to 15 digits)

  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, AAR access code, or ARS 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 CDR account code.

  Four digit IXCs will use one digit of the Account Code for LSU formats.

  Account code dialing can be optional or mandatory (forced). Forced account code entry is set on a per-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 WATS calls are excluded.

  If the ISDN 18-word format CDR 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.

  Intra-switch CDR will not output this field.

- Attendant Console (two digits) (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.

  Intra-switch CDR will not output this field.

- Authorization Code (seven digits)

  This field contains the four- to seven-digit authorization code used to make the call. 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. On the 59-character record, the authorization code is never recorded.

  Intra-switch CDR will not output this field.

- BCC (one digit) (24-Word Record Only)

  This field contains the BCC for ISDN calls, identifying the type of an ISDN call. It will distinguish between voice and different types of data. Intra-switch CDR will not output this field. 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 five and 10 digits in 24-word format)

  For outgoing or intra-switch calls, this field contains the extension number of the originating voice terminal user. For incoming and tandem calls, this field contains the TAC of the trunk group used for the call. With the 24-word format, the calling number field is 10 digits and contains the SID/ANI (G1.1) or CPN/BN (G3i) information on incoming ISDN calls. The Calling Number field contains the local extension of the NCA-TSC endpoint when the CDR record is for an outgoing (or originating) NCA-TSC. This field is blank for other NCA-TSC CDR records (that is, terminating, tandem, or unsuccessful). Information in this field is right adjusted.

- Condition Code (one character)

  These codes reflect special events relating to the call. The condition codes apply to the printer, TELESEER CDR unit, 94A LSU, and 3B2 CDRU. These condition codes are listed and defined in Table 2-5. Condition codes for the 59-character CDR record are different from the codes in Table 2-5, but can be mapped to these codes as shown below:

Condition Code Mapping for 59-Character Record

59-Character Condition Code	Code From Table 2-5
A	1
D	4
G	7
H	8
I	9
L	C
N	E

Table 2-5. Condition Codes

Condition Codes	Description
0	Identifies an intra-switch call (a call that originates and terminates on the switch).
1	Identifies an attendant-handled call or an attendant-assisted call (except conference calls).
4	Identifies an extremely long call (10 hours or more) or an extremely high message count TSC (9999 messages or more). On a call exceeding 10 hours, 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 AAR or ARS Selection feature.
8	Identifies calls which have been served on a delayed basis via the Ring-back Queuing feature.
9	Identifies an incoming or tandem call.
A	Identifies an outgoing call.
B	Identifies an adjunct-placed outgoing call.
C	Identifies a conference call. For trunk CDR, 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. For intra-switch CDR, if the originator is optioned for intra-switch, each time the originator dials a non-trunk party a separate call record is produced with this condition code. If the originator is not optioned for intra-switch CDR, a separate record with this condition code is produced for each intra-switch party dialed.
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. 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 because of either insufficient calling privileges of the originator (assigned per FRL), ISDN calls rejected by the switch due to an NSF mismatch, or an authorization mismatch which prevents the completion of a data call.

NOTE:
When more than one condition applies to a call, the overriding code is shown in Table 2-6.

When two condition codes apply on the same call, one will override the other. The matrix in Table 2-6 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 11 horizontal rows (0, 1, 4, 7, 8, 9, A, B, C, E, and F) and 11 vertical columns (0, 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 2-6. Condition Code Override Matrix

	Condition Code										
	0	**1**	**4**	**7**	**8**	**9**	**A**	**B**	**C**	**E**	**F**
0	NA	0	4	0	NA	NA	NA	B	C	NA	NA
1	0	NA	4	1	NA	9	1	B	C	E	NA
4	4	4	NA	4	4	4	4	4	4	NA	NA
7	0	1	4	NA	7	9	7	B	C	E	F
8	NA	NA	4	7	NA	NA	8	B	C	E	NA
9	NA	9	4	9	NA	NA	NA	NA	C	E	F
A	NA	1	4	7	8	NA	NA	B	C	E	F
B	B	B	4	B	B	NA	B	NA	B	E	F
C	C	C	4	C	C	C	C	B	NA	NA	NA
E	NA	E	NA	E	E	E	E	E	NA	NA	NA
F	NA	NA	NA	F	NA	F	F	F	NA	NA	NA

- Dialed Number (up to 15 digits)

  This field contains the number dialed by a system user. If more than 15 digits are dialed, the least significant digits are truncated. Intra-switch CDR will output this number.

  The # sign (‘‘E’’ with the 94A LSU format) may be printed in this field in the following cases for both ARS and TAC calls:

  — When the user dials a feature access code that starts with a **#**

  — When the user dials **#** at the end of digit dialing (eg. for WATS and IDDD calls)

  — When the inter-digit time-out occurs before the answer supervision time-out, even if the user has not dialed the # sign

  — If CDR Privacy is enabled for the calling number (this feature is available on a per station basis and is administered on the station form) and this is an outgoing call (that is, not an incoming or intra-switch call). In this case, the trailing digits of the dialed number will be blanked in the CDR for the call. If more than 15 digits are dialed, the dialed number will first be truncated to 15 digits, then the appropriated number of digits will be blanked. The number of blanked digits is administered system wide on the Feature Related

System Parameters form.

For an outgoing (or originating) NCA-TSC or tandem NCA-TSC, this field contains the dialed digits used to establish a route to a far-end PBX. It contains the extension of the local extension used as the NCA-TSC end-point when its for a terminating NCA-TSC. For an unsuccessful NCA-TSC, this field is blank.

- Duration (four 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).

- FRL (one digit)

  FRLs, numbered zero through seven, are associated with the AAR and ARS 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 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 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 TCM sent with the tandem tie trunk call, depending on which was used to complete the call. On ISDN calls, this field always contains the TCM, if it was received.

  — The Feature-Related System Parameters can be administered to have ''Disconnect Information in Place of FRL.'' For trunk CDR, the following call disconnect data is printed in this field in place of the FRL data:

Data	Meaning (for calls)
0	Don't know who dropped first
1	We dropped first
2	The CO dropped first
3	Maintenance got the trunk

For intra-switch CDR, if ''Disconnect Information in Place of FRL'' has been administered, the following call disconnect data is printed in this field in place of the FRL data (which is not output for intra-switch CDR):

Data	Meaning (for calls)
0	Indeterminate
1	calling number dropped first
2	dialed number dropped first

Indeterminate refers to all conference and transfer calls or any other call it could not be determined who dropped first.

- Feature Flag (one digit)

  Intra-switch CDR will not output this field.

  With G1.1, 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.

  With G3i, the digit in this field indicates whether or not the switch has received answer supervision from the network and whether the call was a voice or data call (Data Call CDR):

  — A `4` in this field indicates a voice call with network answer supervision.

  — A `0` in this field indicates a voice call without network answer supervision.

  — A `5` in this field indicates a data call with network answer supervision.

  — A `1` in this field indicates a data call without network answer supervision.

  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 timeout 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.

With G3i, calls are flagged as data calls if they use a conversion resource (such as a modem pool) and/or originate or terminate on a data module.

- Incoming TAG (four digits) (24-Word Records Only)

  This field contains the access code of the incoming trunk group.

  Intra-switch CDR will not output this field.

- INS (three digits)

  This field specifies the INS requested for a call. This field applies only to ISDN calls. Each Network Specific Facility is translated into an INS according to the following table.

Table 2-7. 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	353
MEGACOM	354
INWATS	355
Maximum Banded WATS	356
AT&T Long Distance Service	358
ACCUNET Digital Service	357
OUTWATS Band 0 OUTWATS Band 1 . . . OUTWATS Band 255	33 34 . . . 288
International 800	359
Multiquest	367

Intra-switch CDR will not output this field.

- IXC Code (one digit hexadecimal representation) (three digits with an ISDN format and four digits with IXC format)

  — Non-ISDN Formats

  IXC codes, numbered one through 15 (1 through F hexadecimal), are associated with the AAR and ARS features and depict the carrier used on the call. This information is sent to the CDR 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. In the U.S., this number is of the form 10XXX, 950 — 1XXX, 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 IXC carrier is not used on the call or the carrier is selected at the CO.

  — ISDN Formats

  With an ISDN record format, this field is a three-digit field that identifies the actual IXC used on an ISDN call. This information is determined from the routing pattern administration. On AAR and ARS calls, the three-digit 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 CDR device will put only the last three digits in the CDR record. If a user dials a seven-digit IXC code, this field will contain a zero.

  Intra-switch CDR will not output this field.

- Incoming Circuit Identification (three digits)

  This field contains the member number of a trunk within a trunk group used for an incoming call.

  Intra-switch CDR will not output this field.

- MA-UUI (one digit)

  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 zero to nine and is found only on 24-word records.

  Intra-switch CDR will not output this field.

- Node Number (two-digits) (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.

  Intra-switch CDR will not output this field.

- Outgoing Circuit Identification (three digits)

For outgoing calls, this field contains the member number of the trunk within a trunk group used. This field is blank for incoming calls. Tandem calls include both incoming and outgoing circuit id numbers. For outgoing and tandem NCA-TSCs, this field contains the signaling group used to carry the NCA-TSC.

Intra-switch CDR will not output this field.

- Packet Count (four digits)

  For ISDN TSCs, this field contains the number of ISDN-PRI. USER INFOrmation messages sent, received, or (for tandem TSCs) passing through the switch.

  Intra-switch CDR will not output this field.

- Resource Flag (one digit) (G3i) (24-Word Records Only)

  The digit in this field indicates whether or not a conversion device was used on the call. A ‘‘1’’ in this field indicates a conversion device was used. A ‘‘0’’ in this field indicates a conversion device was not used.

  Intra-switch CDR will not output this field.

- Time in Queue (two digits)

  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 two digits of the INS code. This field is always blank in the 24-word records. The remaining codes are defined in Table 2-5.

  Intra-switch CDR will not output this field.

- TSC Flag (one digit)

  This field distinguishes CDR records pertaining to TSCs. When not equal to zero, this field will indicate the status of the TSC. The following table presents the TSC Flag encoding.

  Intra-switch CDR will not output this field.

Table 2-8. Encoding for CDR TSC Flag

Encoding	Meaning
	Encoding for CDR TSC Flag
0	Circuit-switched call without TSC requests
1	Reserved for future use
2	Reserved for future use
3	Reserved for future use
4	Call Associated TSC requested, accepted in response to SETUP, no congestion control (applicable to originating node)
	Call Associated TSC received and accepted via SETUP, no congestion control (applicable to terminating node)
5	Call Associated TSC requested, accepted in response to SETUP, congestion control (applicable to originating node)
	Call Associated TSC received and accepted via SETUP, congestion control (applicable to terminating node)
6	Call Associated TSC requested, accepted after SETUP, no congestion control (applicable to originating node)
	Call Associated TSC received and accepted after SETUP, no congestion control (applicable to terminating node)
7	Call Associated TSC requested, accepted after SETUP, congestion control (applicable to originating node)
	Call Associated TSC received and accepted after SETUP, congestion control (applicable to terminating node)
8	Call Associated TSC requested, rejected (rejection came from outside the local switch)
9	Call Associated TSC requested, rejected (rejection came from the local switch, that is, lack of resource)
A	Non Call Associated TSC requested, accepted, no congestion control (applicable to originating node)
	Non Call Associated TSC received, accepted, no congestion control (applicable to terminating node)
B	Non Call Associated TSC requested, accepted, congestion control (applicable to originating node)
	Non Call Associated TSC received, accepted, congestion control (applicable to terminating node)
C	Non Call Associated TSC requested, rejected (rejection came from outside the local switch)
D	Non Call Associated TSC requested, rejected (rejection came from the local switch, that is, lack of resource)
E	Reserved for future use
F	Reserved for future use

The call detail information sent to the TELESEER CDR unit is shown in Table 2-9 and Table 2-10 (with ISDN).

The call detail information sent to the printer is shown in Table 2-11 and Table 2-12 (with ISDN).

The call detail information sent to the 94A LSU and 3B2 CDRU is shown in Table 2-13 (with G1.1 and G3i), Table 2-14 (with ISDN), and Table 2-15 (G3i with ISDN).

The call detail information included in the 59-character Direct Output Record is shown in Table 2-16.

The call detail information included in the G1.1 24-word unformatted record and the G1.1 24-word expanded record is shown in Tables 2-17 and 2-18, respectively.

The call detail information included in the G3i 24-word unformatted record and the G3i 24-word expanded record is shown in Tables 2-19 and 2-20, respectively.

Table 2-9. CDR Data Format — TELESEER CDR Unit

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 2-37.
** Data is right justified and padded with blanks.
‡ Data is right justified and padded with Os.

Table 2-10. CDR Data Format — TELESEER CDR Unit With ISDN

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*
13-16	IXC**
17-19	Access Code Used**
20-34	Dialed Number**
35-39	Calling Number**
40-54	Account Code**
55	INS (third digit)
56	FRL
57-59	Incoming Circuit ID‡
60-62	Outgoing Circuit ID‡
63	Feature Flag
64-70	Authorization Code
71-72	INS (first and second digits)
73-76	Space
77	Carriage Return
78	Line Feed
79-81	null

* Refer to Table 2-37.
** Data is right justified and padded with blanks (spaces).
‡ Data is right justified and padded with Os.

Table 2-11. CDR Direct Output Format From System to Printer

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 2-37.
** Data is right justified and padded with blanks.
‡   Data is right justified and padded with Os.

Table 2-12. ISDN CDR Direct Output Format from System to Printer

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 2-37.
** Data is right justified and padded with blanks (spaces).
‡ Data is right justified and padded with Os.

Table 2-13. 94A Local Storage Unit System or 3B2 CDRU

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** ** (second through fifth digits for five-digit dialing plan)
31-35	Account Code (First five Digits)**
36-42	Authorization Code or sixth through twelfth Digits of Account Code**
43-44	Time in Queue or thirteenth and fourteenth digits of account code**
45	FRL or fifteenth Digit of Account Code**
46	Calling Number (first digit of a five-digit calling number)
47-48	Incoming Circuit ID §
49	Feature Flag
50-52	Outgoing Circuit ID §
53	Incoming Circuit ID (third digit) [reserved for future use — data is blank]
54	IXC
55	Carriage Return
56	Line Feed
57-59	null

* Refer to Table 2-37.
** Data is right justified and padded with blanks (spaces).
** For a 4-digit dialing plan, this field will record first four digits of the calling number.
§ Data is right justified and padded with zeros.

Table 2-14. 94A Local Storage Unit System or 3B2 CDRU (G1.1)

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*
06-09	IXC **
10-12	Access Code Used**
13-27	Dialed Number**
28-31	Calling Number**
32-35	Account Code (First five Digits)**
36-42	Authorization Code or sixth Through twelfth 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 2-37.
** Data is right justified and padded with blanks (spaces).
§ Data is right justified and padded with zeros.

Table 2-15. 94A Local Storage Unit System or 3B2 CDRU (G3i)

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 five Digits)**
36-42	Authorization Code or sixth through twelfth Digits of Account Code**
43-44	INS or thirteenth and fourteenth Digits of Account Code**
45	INS (third Digit), FRL, or fifteenth 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	FRL or IXC**
55	Carriage Return
56	Line Feed
57-59	null

*	Refer to Table 2-37.
**	Data is right justified and padded with blanks (spaces).
§	Data is right justified and padded with zeros.

Table 2-16. CDR 59-Character Direct Output Format

ASCII Character Position	Data Field Description
01	Time-Hours (tens)
02	Time-Hours (units)
03	Time-Minutes (tens)
04	Time-Minutes (units)
05	Duration-Hours
06	Duration-Minutes (tens)
07	Duration-Minutes (units)
08	Duration-Minutes (tenths)
09	Condition Code*
10-12	Access Code Dialed**
13-15	Access Code Used**
16-30	Dialed Number**
31-35	Calling Number**
36-50	Account Code**
51	FRL
52	IXC
53-55	Incoming Circuit ID
56-58	Outgoing Circuit ID
59	Carriage Return
60	Line Feed
61-63	null

* Refer to Table 2-37.
** Data is right justified and padded with blanks (spaces).
§ Data is right justified and padded with zeros.

Table 2-17. 24-Word ISDN Unformatted CDR Record Format (G1.1)

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**
86-89	IXC**
90	BCC
91	MA-UUI
92	Resource Flag
93-96	Packet Count
97	TSC Flag
98-100	Reserved
101	Carriage Return
102	Line Feed
103-105	null

* Refer to Table 2-37.

** Data is right justified and padded with blanks.

Table 2-18. 24-Word ISDN Expanded CDR Record Format (G1.1)

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*
12	Space
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
78-79	Space
80-81	Incoming Circuit ID **
82	Space
83	Space
84-85	Outgoing Circuit ID **
86	Space
87	Feature Flag
88	Space

* Refer to Table 2-37.
** Data is right justified and padded with blanks (spaces).

Continued on next page

Table 3-18. 24-Word ISDN Expanded CDR Record Format (G1.1) ***(Continued)***

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 2-37.
** Data is right justified and padded with blanks.

Table 2-19. 24-Word ISDN Unformatted CDR Record Format (G3i)

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**
86-89	IXC**
90	BCC
91	MA-UUI
92	Resource Flag
93-96	Packet Count
97	TSC Flag
98-100	Reserved
101	Carriage Return
102	Line Feed
103-105	null

* Refer to Table 2-37.
** Data is right justified and padded with blanks.

Table 2-20. 24-Word ISDN Expanded CDR Record Format (G3i)

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*
12	Space
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

* Refer to Table 2-37.

** Data is right justified and padded with blanks.

Continued on next page

24-Word ISDN Expanded CDR Record Format (G3i) ***(Continued)***

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
104-107	IXC*
108	Space
109	BCC
110	Space
111	MA-UUI
112	Space
113	Resource Flag
114	Space
115-118	Packet Count
119	Space
120	TSC Flag
121	Space
122	Reserved
123	Space
124	Reserved
125	Space
126	Reserved
127	Space
128	Reserved
129	Space
130	Reserved
132	Space
133	Carriage Return
134	Line Feed
134-136	null

* Data is right justified and padded with blanks.

International CDR Enhancements for Periodic Pulse Metering

The CDR output interface has been modified to include two new CDR record formats. The formats are identical to the TELESEER® CDR Unit and printer formats used in International System 75 (IR1V2) and provide Periodic Pulse Metering (PPM) pulse counts in the output record.

The new CDR output formats are selected using Feature Related System Parameters. Table 2-21 shows the CDR output format when `int-process` is entered in an `CDR Output Layout` field on the screen. Table 2-22 shows the output format when `int-direct` is entered in an CDR Output Layout field.

PPM uses pulses transmitted over the trunk line from the serving CO at periodic intervals during the course of the outgoing call to determine call charges. Each pulse has an intrinsic value; at the end of the call, the sum of the pulses represents the charges for the call. The more expensive the call, the faster the sending rate of the pulses becomes.

The TN465 Loop Start CO Trunk circuit pack is capable of detecting 16-kHz PPM pulses. The pulses can occur at a maximum of once every 1.4 seconds during the call or, if the CO accumulates the total PPM count, they can occur up to three to five seconds after the call is dropped. In order to ensure that delayed PPM data is received, the Outgoing Glare Guard Timer on the trunk group should be adjusted accordingly. Once the trunk circuit has collected the PPM count, the CDR output may be either an International TELESEER or International Printer record. Each of the records has a format to include PPM pulse count information.

PPM pulse detection is applicable for outgoing calls on CO, DIOD, FX, PCOL, and WATS trunks. The PPM field on the trunk group form should be set to ‘‘y’’ (yes) to enable the PPM detection and reporting functions.

Table 2-21. CDR International Data Format — TELESEER CDR Unit

ASCII Character Position	Data Field Description
01	Format Code (first)
02	Format Code (second)
03	Time Hour - (tens)
04	Time Hour - (units)
05	Time Minute - (units)
06	Time Minute - (tens)
07	Duration Hour
08	Duration Minute - (tens)
09	Duration Minute - (units)
10	Duration Minute (tenths)
11	Space
12	Condition Code
13	Space
14	Access Code dld - (1st digit)
15	Access Code dld - (2nd digit)
16	Access Code dld - (3rd digit)
17	Access Code usd - (1st digit)
18	Access Code usd - (2nd digit)
19	Access Code usd - (3rd digit)
20	Space
21-38	Dialed Number (1st to 18th digit) * **
39-43	Calling Number (1st - 5th digit) **
44	Space
45-59	Account Code (1st - 15th digit) **
60	Space
61	IXC
62	FRL
63-65	Space
66-67	Incoming Circuit ID (1st - 2nd digit) ‡
68-70	Space
71-72	Outgoing Circuit ID (1st - 2nd digit) ‡
73	Space
74-78	PPM Count (1st - 5th digit) **
79	Carriage Return
80	Line Feed
81-83	Null

* 21-23 are blank.
** Data is right justified and padded with blanks.
‡ Data is right justified and padded with 0’s

Table 2-22. CDR International Data Format — Printer Record

ASCII Character Position	Data Field Description
01	Date of Month - (tens)
02	Date of Month - (units)
03	Month - (tens)
04	Month - (units)
05	Year - (tens)
06	Year - (units)
07	Space
08	Time Hour - (tens)
09	Time Hour - (units)
10	Time Minute - (tens)
11	Time Minute - (units)
12	Space
13	Duration Hour
14	Duration Minute - (tens)
15	Duration Minute - (units)
16	Duration Minute - (tenths)
17	Space
18	Condition Code
19	Space
20-22	Access Code # 1 (1st - 3rd digit) *
23-25	Access Code # 2 (1st - 3rd digit) *
26	Space
27-44	Dialed Number (1st - 18th digit) * §
45	Space
46-50	Calling Number (1st - 5th digit) *
51	Space
52-66	Account Code (1st - 15th digit) *
67	Space
68-72	PPM Count (1st - 5th digit) *
73	Space
74	Incoming Circuit ID (1st digit) **
75	Incoming Circuit ID (2nd digit) ‡
76	Space
77	Outgoing Circuit ID (1st digit) **
78	Outgoing Circuit ID (2nd digit) ‡
79	Carriage Return
80	Line Feed

* Data is right justified and padded with blanks.
** Data is right justified and padded with 0's
‡ Data is blank
§ 27-29 are blank

Date Record Format

Three formats are available for date records, one for the 94A LSU or 3B2 CDRU (Table 2-23), one for the printer (Table 2-24), and one for the TELESEER CDR unit (Table 2-25). The records sent to the TELESEER CDR and printer contain the date only while the records sent to the 94A LSU or 3B2 CDRU contain both date and time.

Table 2-23. Date Record Format to 94A LSU or 3B2 CDRU

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.

Table 2-24. Date Record Format to Printer

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.

Table 2-25. Date Record Format to TELESEER CDR Unit

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.

Set Time and Date

The system clock must be set for daylight savings time. Changing the time and date ensures that CDR records have the correct time and date for the records being kept. The time and date can be changed using the Manager I terminal (G1) or the G3 Management Terminal.

If the time is changed while calls are in progress, the actual call durations for these calls are not reflected in the CDR record.

CDR Output Devices

CDR data is collected by the system and is continually sent to an output device for processing. An output device could be a TELESEER CDR unit, printer, 94A LSU, host computer, or customer-provided equipment. A system can have two output devices.

A standard 232C interface is provided by the system's processor circuit pack. This allows for direct connection of the CDR output device to the system. If this port is not used, additional interface equipment is required as described in the Hardware and Software Requirements part of this feature description. The system can support two CDR output devices. One of these devices can use the direct EIA-232C connection; the other device will require the additional interface equipment.

When a system has two CDR output devices, one device is administered as the Primary CDR Output Device; the other device is administered as the Secondary CDR 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 for assessing network performance or helping to find network problems. The following information applies to the port used for the Secondary CDR output device:

- Data going to the secondary port should be the same as that going to the primary port. However, CDR 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 CDR Output Device, the system will discontinue sending records to the secondary port for two 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 CDR port, the secondary port is busied out for two minutes. This makes system resources available to send data to the primary CDR port before the data is lost. The system will continue to busy out the secondary port for two-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 (G1.1) or 250 (G3i) 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 (G1.1) or 250 (G3i) CDR records which are sent to the output devices.

A 1,200 bits/second rate may only be used over the cable for the TELESEER CDR unit, 94A LSU, or printer, if the system line size is less than 1,000. If the system line size is greater than 1,000, a rate of at least 2,400 must be used with the 18-word CDR records. If the system line size is greater than 600, a rate of at least 2,400 must be used with the 24-word CDR records.

The time stamp on calls recorded by CDR 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 CDR call detail record.

TELESEER CDR Unit

The TELESEER CDR 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 and extension
- Call type
- Account code
- Access code/trunk number/trunk group number
- Printed call categories
- Recorded call categories

The TELESEER CDR unit can store up to 28,000 call records, 500 extension numbers, and 2000 account codes. CDR records sent to the TELESEER CDR unit are 80 bytes or 640 characters long.

The TELESEER CDR 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 2-17.

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 2-18.

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 2-19.

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 CDR 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 2-20.

Printer

An 80- or 132-column (character) printer can be connected as an CDR output device. The printer prints CDR records in a two-line format. No data processing or reports are provided. The 18-word CDR records sent to the printer are 84 bytes or 672 bits long. The 24-word CDR records sent to the printer are 135 bytes or 1,080 bits long.

94A Local Storage Unit (LSU)

The 94A LSU collects and stores MDRs data from the system. The 94A LSU stores MDRs for ETN customers or multi-location customers served by other System 75s, DEFINITY Generic 1s, or DEFINITY Generic 3is. CDR 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

TIME REPORT STARTED PRINTING

FIRST DATE OF REPORT INTERVAL

SECOND DATE OF REPORT INTERVAL

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

TOTAL SUMMARY BY DEPTS.

TOTAL NUMBER OF CALLS MADE BY EACH DEPT.

23-MAR-84 - 23-MAR-84 SUMMARY REPORT
04:58P COMPANY NAME PAGE 2

DURATION OF CALLS MEASURED IN HOURS, MINUTES, AND SECONDS

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

COST FOR EACH DEPT.

23-MAR-84 - 23-MAR-84 SUMMARY REPORT
04:58P COMPANY NAME PAGE 3

BY DEPARTMENT

TOTAL SUMMARY BY EXT. NO. IN DEPT ACCOUNTING

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 2-6. Example of a TELESEER CDR 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 09:43AM      3:00    101     7           454-1362       FX 556432            1.05
03/23 01:01PM      4:00    530     9           473-1502      LOC 556432            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  1766     9  1-416-324-5012       IDD  88644              2.55
03/23 01:16PM     30:00  2388     9  1-206-888-4587       OST88644               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 11:29AM     11:00  2400     9  98720112223334131293124 59
                                                       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 41363
  DATE   TIME   DURATION EXT  ACC DIALED DIGITS          TYP ACCOUNT CODE COST $
  ----   ----   --------  ---  ---  ------ ------          ---  ------- ----      -------
03/23 11:01AM     12:00  3011     4  663-2828              FX 41364               5.01
03/23 01:15PM      5:00  3011     7  288-5454              FX 41364               1.77
03/23 02:13PM      3:00  2400     6  1-417-987-3498       WTS 41363               0.70


      NUMBER                  DURATION                 COST
        3                      0:20:00           $     7.48
```

Figure 2-7. Example of a TELESEER CDR Unit Account Code Detail Report

FIRST DATE OF REPORT INTERVAL

TIME REPORT STARTED TO PRINT

23-MAR-84 - 23-MAR-84 ACTIVITY REPORT
05:07P 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	6	1-206-324-5151	WST		2.86
03/23 03:12PM	5:00	4155	6	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	6	1-408-454-1362	WST	98267	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 4500

DATE TIME	DURATION	EXT	ACC	DIALED DIGITS	TYP	ACCOUNT CODE	COST $
03/23 06:05AM	11:00	4500	9	1-419-643-7474	0ST	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	98267	0.33
TOTALS	11:00						1.82

TOTAL CALLS 2

Figure 2-8.

Example of a TELESEER CDR 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		1.79
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	11:29AM	11:00	2400	9	9872011222333413129312459			
						OCC	95643	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	1-419-643-7474	OST	54321	1.49
03/23	06:05PM	11:00	4500	9	1-818-643-7474	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	88644	13.20
03/23	02:35PM	15:00	1122	9	9871011278565618198883200			
						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	9872011263457613153657219			
						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 2-9. Example of a TELESEER CDR Unit Selection Report

Considerations

CDR provides detailed call information on incoming and outgoing trunk calls, and intra-switch calls. This information can be used to facilitate cost allocation, traffic analysis, and detection of unauthorized calls.

The system can store up to 200 (G1.1) or 250 (G3i) CDR records for the primary port, which are sent to the output devices when the link comes back up.

The TELESEER CDR unit can store up to 28,000 call records, 500 extension numbers, and 2,000 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 7,200 calls per hour to a 93B CMDR.

When a voice terminal user wants an CDR record generated for a particular account number, the CDR access code (for example, **6** and the account number must be dialed before the ARS, AAR, or TAC and called number are dialed.

The originally dialed extension number or the trunk access code on an incoming call, or the originator's extension number on an outgoing call, is always recorded for CDR 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 CDR. 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 ARS or TAC.

If the system line size is greater than 1,000, the CDR device must support a baud rate of at least 2,400 bps when using 18-word CDR records. If the system line size is greater than 600, the CDR device must support a baud rate of at least 2,400 bps when using 24-word CDR records.

CDR records of DS-1 calls are only generated if the answer supervision timeout is exceeded or if the call is answered at the far end. Therefore, more accurate CDR 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 (250 seconds).

Interactions

The following interaction discussions assume CDR 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 CDR 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.

It is not possible to option the attendant for intra-switch calls. Intra-switch records are produced for an intra-switch optioned extension calling the attendant or for a call from the attendant to an intra-switch optioned extension. In the case of an attendant-assisted call involving an intra-switch extension, the calling number recorded is the extension of the party who called the attendant, and the dialed number recorded is the extension that the attendant extended the call to. The record will have a Condition Code 0.

- AUDIX

  For remote AUDIX over DCS, if station A on node 1 forwards its calls to AUDIX on node 2, the CDR record is produced on each switch. The record from node 1 contains A as the dialed number. The record form node 2 contains AUDIX as the dialed number. If the calling number is on a different switch within the DCS network, or the calls comes in over ISDN, the actual calling number will be recorded in the `Calling Number` field, and the TAC of the trunk bringing the call into the local switch will be recorded in the `Incoming Trunk Access Code` field of 24-word records. If the forwarded call is an incoming call, then, as in all cases (other than vectoring) in which an incoming call is forwarded, transferred, or conferenced using an outgoing trunk, two separate CDR records are produced, one for incoming and one for outgoing trunk usage. The outgoing trunk usage record lists AUDIX as the `Calling Number`.

- Authorization Codes

  Authorization codes will be recorded on 94A LSU and 3B2 CDRU CDR records if account codes do not exceed five digits. The authorization code is always recorded on the printer, TELESEER CDR, and 24-word CDR records. On the 59-character CDR records, the authorization code is never recorded.

- AAR and ARS

  CDR records the following information for Automatic Route Selection (ARS):

  — Fact that an ARS call was made

  — Calling extension number

— FRL 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)

— IXC code, if any

If CDR is suppressed for the trunk group actually used on an ARS call, an CDR 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 CDR is suppressed for that trunk group, the user hears reorder tone and the CDR 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 CDR 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 CDR 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.

- Automatic Callback

  When the Automatic Callback feature is used for an intra-switch call, no CDR record will be generated for the first call attempt or the ringback. If the caller or extension being called is optioned, however, a record of the actual call will be output.

- ACA

  ACA calls will generate intra-switch CDR if the terminating extension is monitored. The originating extension for ACA calls cannot be administered for intra-switch monitoring.

- Automatic Wakeup

  No CDR intra-switch records will be generated for wakeup calls.

- Bridged Call Appearance

  CDR 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 CDR record. The duration of the call is recorded when the last party drops off the call. This also applies for intra-switch calls.

- 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 CDR record. If a Call-By-Call

Service 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 or intra-switch 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.

  There will be only one record generated for a forwarded intra-switch call. In this record, the dialed number will be the same as the extension dialed by the originating party.

  If a T1 trunk calls a station and the station is forwarded to another T1 trunk, the outgoing record will show the station being forwarded to the second T1 trunk rather than the first T1 trunk being forwarded to the second T1 trunk.

- Call Park

  When a voice terminal user parks an incoming or intra-switch call, that user’s extension is recorded as the dialed number in the CDR record. Call duration in CDR reflects the entire time the incoming trunk is busy (incoming) or until the call ends (intra-switch).

- Call Pickup

  When an incoming or intra-switch 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 Vectoring (G3i)

  The Feature Related System Parameters form can be administered so that the VDN extension will be used in place of the Hunt Group or Member extension. If administered to do so, this overrides the ‘‘Call to Hunt Group - Record’’ option of CDR for incoming Call Vectoring calls.

  For incoming calls to a VDN, the duration of the call is recorded from the time answer supervision is returned.

  — If answer supervision is returned by the vector (via an announcement, collect, disconnect, or wait with music command), and the call nevers goes to another extension, then the VDN extension is recorded as the called number in the CDR record.

  — If the call terminates to a hunt group, then the VDN, hunt group, or agent extension is recorded as the called number as per the administration discussed above.

  — If the call terminates to a trunk, then the following two CDR records will be generated:

1. An incoming record with the VDN as the called number and the duration from the time answer supervision was provided to the incoming trunk.

2. An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.

Outgoing vector calls generate ordinary outgoing CDR records with the originating extension as the calling number.

No Ineffective Call Attempt records will be generated for Call Vectoring **route to** commands that are unsuccessful.

If a vector interacts with an extension or group that has Call Forwarding All Calls active, normal Call Forwarding/CDR interactions apply.

Some calls may originally look like intra-switch calls, but result in trunk calls (for example, a call from a station administered for intra-switch CDR to a VDN, which ends up an outgoing call on an outgoing trunk). Such calls will not generate intra-switch CDR records; the CDR record will have a condition code A - outgoing.

- Call Waiting Termination

  Call duration timing starts when the voice terminal answers an incoming call.

- CAS

  If a CAS attendant extends a call for a user, and CDR is not assigned to the RLT trunk group, the user's extension is recorded as the originator of the call. If the RLT trunk group does have CDR administered, the RLT trunk is recorded. If a CAS attendant answers a call but does not extend the call, no CDR records are made.

  Intra-switch CDR records are generated for calls that are extended within the invoking switch if the originator or dialed number is optioned for intra-switch CDR.

- CO Trunks

  All incoming and outgoing calls on a CO trunk group will be recorded.

- Conference

  For the purpose of CDR recording, a call is considered a conference call if it contains at least one trunk which is eligible for CDR recording plus two or more non-attendant parties, or if it contains at least one party optioned for intra-switch CDR. Condition Code C applies to each CDR record made for a conference call.

  For a conference call, a separate CDR 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 CDR reflects the entire time the trunk was on the conference call.

Trunk-to-trunk transfer calls are treated like conference calls for CDR purposes. A separate CDR record is produced for each trunk used in a trunk-to-trunk transfer.

If the originator of the conference call is optioned for intra-switch CDR, each time the originator dials a non-trunk party, a new CDR record is started. For example, Station A is optioned for intra-switch CDR and calls Station B. Station A conferences in Station C. Station A drops from the call. Station B or C drops from the call. Two CDR records are output with Condition Code C: one for the A to B call and one for the A to C call.

If the originator is not optioned for intra-switch CDR, but one or more parties brought into the conference are, one record with Condition Code C will be generated for each dialed intra-switch party. For example, Station A calls Station B, which is optioned for intra-switch CDR. Station A conferences Station C. Station A drops from the call. Station B or C drops from the call. One CDR record is output with condition code C for the A to B call.

Intra-switch conference call CDR records are output when both the calling number (originator) and dialed number (terminator) of the call drop. The duration of the call will be from the time the terminator answers until both the originator and terminator drop from the call.

If the attendant originates the conference, only the dialed numbers corresponding to intra-switch optioned extensions will stimulate the creation of CDR records.

- DCS

  If the calling number is on a different switch within the DCS network, the actual calling number will be recorded in the `Calling Number` field, and the TAC of the trunk bringing in the call will be recorded in the `Incoming Trunk Access Code` field. DCS signaling messages do not generate CDR records.

- DDC and UCD

  Either the hunt group extension number or individual hunt group member extension number (depending on administration) is recorded as the called number.

- DID

  All incoming calls on the DID trunk group will be recorded.

- Emergency Access to the Attendant

  No intra-switch CDR records will be generated for Emergency Access calls.

- FX Trunks

  All calls made on an FX trunk group will be recorded.

- Hot Line Service

  The stored number used on an outgoing or intra-switch Hot Line call is recorded by CDR 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

  Intercom calls can generate intra-switch CDR records.

- Intercom — Dial

  Intercom calls can generate intra-switch CDR records.

- Inter-PBX Attendant Calls

  If a user calls an Inter-PBX attendant and the trunk group used has CDR assigned, CDR 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

- ISDN

  When specific answer supervision is received from the network, an indication is sent to the CDR device to this effect. If an ISDN call has been interworked, the CDR record will not record the call as having answer supervision.

- Loudspeaker Paging

  When loudspeaker or chime paging is used, CDR may not correctly record the length of time a station was connected to an AUX trunk.

- Manual Originating Line Service

  If an attendant establishes an outgoing call for a voice terminal, designated as a Manual Originating Line, the CDR 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 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.

LDNs cannot be administered for intra-switch CDR. However, a call from an intra-switch optioned extension to a LDN will produce an intra-switch CDR.

- 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

  CDR data is recorded if the voice terminal is involved in an outgoing/incoming trunk call or it (or the other terminal involved in the call) has been optioned for intra-switch CDR.

- PCOL

  A PCOL call will show the dialed number in the dialed number field of the CDR record rather than a TAC.

- 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 CDR record is not generated for the call.

- SVN

  SVN calls will generate intra-switch CDR if the terminating extension is monitored. The originating extension for SVN calls cannot be administered for intra-switch monitoring.

- Service Observing

  No CDR records will be generated for Service Observing calls.

- Tandem Tie Trunk Switching

  The calling party on an incoming trunk can dial the CDR account code. The calling number field in CDR is the trunk access code for the incoming trunk group, the called number is the number dialed.

- Temporary Bridged Appearance

  An CDR record is not affected by any second or subsequent voice terminal bridging a call.

- Temporary Signaling Connections (TSC)

  TSCs and TSC requests associated with recorded calls will be recorded in the CDR record of the associated call, provided the switch is administered to use the ISDN version of CDR format layouts, which contain the necessary fields. Non-call-associated TSCs and TSC requests sent or received by the switch will generate their own CDR records if the switch is administered to record them. In either case, the `TSC Flag` field and the `Packet Count` field of the CDR record will be used to record TSC data.

- 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.

  If a voice terminal user receives an intra-switch call and then transfers it to another extension, the extension that originally received the call is recorded as the dialed number.

  Intra-switch CDR records are generated for each call to or from an intra-switch optioned extension. For example, Station A, which is intra-switch optioned, calls Station B. Station A then transfers the call to Station C. When either Station B or C drops, two CDR records with Condition Code 0 are output: one for the A to B call, and the second for the A to C call.

  Intra-switch CDR transfer records are output when both the calling number (originator) and dialed number (terminator) drop from the call. The duration of the call is from the time the terminator answers until both the originator and terminator have dropped from the call.

- Trunk-to-Trunk Transfer

  Although they are not really conference calls, Trunk-to-Trunk Transfer connections are treated as such for CDR purposes. A separate CDR 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.

- UDP

  If one user calls another user via a Uniform Dial Plan extension number, and the trunk group used has CDR assigned, CDR 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

- WATS and 800 Service

  Calls made on a WATS or 800 Service trunk group will be recorded, if CDR is administered for the trunk group.

Administration

CDR is administered by the System Manager. The following system parameters can be administered:

- Type of CDR output device to be used. 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. 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 PMS extension or a PCS extension.
- Whether standard or ISDN formats are used.
- CDR account code length (from one to 15), the system defaults to two digits.
- The speed at which the CDR device connected to the direct RS-232C interface on the processor circuit pack will operate (300, 1200, 2400, 4800, or 9600 baud rate).
- 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 CDR.
- Whether or not the called VDN is recorded instead of the hunt group

extension or hunt group member extension (G3i only).

- CDR can be suppressed for Ineffective Call Attempts or for All Calls Excluding Outgoing Calls; system defaults to no. Ineffective call attempts are calls 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. 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 NSF mismatch.
- The number of trailing digits in the CDR dialed number field to be blanked on output for an outgoing call originating from a station with CDR Privacy enabled.
- Whether intra-switch CDR records will be generated for internal calls.

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, CDR records will not be generated.

Trunks, Loudspeaker Paging, and Code Calling Access

CDR can be assigned to all trunk groups, Loudspeaker Paging Access trunks, and PCOL trunks. The system defaults to yes for CDR. The System Manager must determine which types of trunks will be assigned CDR.

COR

Specify if CDR account code entry is forced.

Feature Access Codes

Assign CDR account code access code. The system defaults to *6.

IXC Codes

- IXC access numbers
- Name of IXC (optional)

Modules and Modems

One or both of the CDR output devices can be connected to a PDM, Trunk Data Module, or a Modem. The following items must be administered:

- A netcon channel must be assigned using a data module form and entering data-channel or netcom channel for the type. This channel provides a path for CDR data from the Switch Processing Element to the time-division bus.

- If the CDR output device is connected to a PDM, administer a PDM form.
- If the CDR output device is connected to a Trunk Data Module, administer a Trunk Data Module form.
- If the CDR 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).

A TN726 Data Line circuit pack can be used in conjunction with an ADU to connect a 94A LSU, TELESEER CDR unit, or printer. If the CDR output device is connected to a TN726 Data Line circuit pack via an ADU, administer a Data Line Data Module form and a Netcon Data Module form.

If the EIA port on the Processor Interface circuit pack is used by the output device, the CDR output device extension should be administered as ‘‘eia’’.

Hardware and Software Requirements

Hardware requirements depend on the type of output device used for CDR. The CDR 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 CDR unit (Data Terminal Equipment), the interface equipment consists of either a PDM to a port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law) or a 212A-type modem to a port on a TN742 or TN746B (A-law) 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 Digital Line circuit pack (TN413, TN754B support A-law) or a 212A-type modem to a port on a TN742 or TN746B (A-law) Analog Line circuit pack. In the latter case, a modem pool facility is also required.
- If CDR is connected to a host computer, the computer must be connected over a private line terminated at the switch with a Trunk Data Module.
- A TN726 Data Line circuit pack can be used in conjunction with an ADU to connect a 94A LSU, TELESEER CDR unit, or printer.

Forced entry of account codes software is required.

CDR 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 CDR 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 CDR access code. The user then dials the desired account code, which can contain up to 15 digits. The user then dials the desired trunk access code, AAR access code, or ARS access code.

CDR Account Code Dialing can be optional or mandatory (forced). Forced entry of account codes can be assigned for any of the following:

- All Toll Calls (G1.1)

  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.

- Designated Toll Calls (G3i)

  In G3i, ''toll calls'' are defined by the administered toll analysis table. Each ''Dialed String'' entry in the toll analysis table that is designated as a toll call can also be administered to require forced entry of account codes. If the system is administered to require forced entry of account codes (on the Feature-Related System Parameters form), and a specific number or ''Dialed String'' is administered to require forced entry of account codes, any system user must dial an account code before dialing that number.

  This includes all calls made by AAR, ARS, or TAC.

- Toll Calls Made By Users With a Specific 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.

- Designated Toll Calls Made By Users With a Specific COR (G3i)

  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 that are administered to require forced entry of account codes.

- 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.

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)
- PCOL
- Remote Access Without Barrier Codes
- Trunk-to-Trunk Connections.

Considerations

CDR Account Code Dialing provides an easy method of allocating the costs of specific calls to the correct project, department, and so on. Call information is recorded by the CDR 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 CDR Account Code Dialing feature:

- Authorization Codes

  Authorization codes will be recorded on all CDR printouts except for the 59-character, LSU, and ISDN LSU formats, without regard to account code length. Authorization codes will be recorded on CDR printouts in the LSU and ISDN LSU formats if the account code length does not exceed five digits.

- AAR and ARS

  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 CDR 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.

- CDR

  CDR 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 CDR record shows only the first 2 digits (12) of the account code.

Administration

CDR Account Code Dialing is administered by the System Manager. The following items require administration for forced entry of account codes:

- Whether or not all toll calls require account code entry (per system)
- Whether or not each individual COR requires account code entry
- Whether or not each Dialed String in the Toll Analysis table requires account code entry (G3i)

Hardware and Software Requirements

No additional hardware is required. Optional CDR Account Code Dialing software is required.

Call Forwarding All Calls

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 can activate or deactivate the feature for a particular extension number, TEG, DDC, UCD group, or ACD split (but not vector-controlled splits; see Call Vectoring for more information).

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 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 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 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 will be temporarily 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 TEG, UCD groups, and DDC, Call Forwarding All Calls can only be activated by the attendant or voice terminal user with console permission.

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. 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.

- 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

  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 document.

- PCOL

  PCOL calls cannot be 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.

- CDR

  When a call is forwarded to an off-premises number, the call is recorded in CDR records as a call from the forwarding station.

- CDR 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 COS. 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 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 three-way conference. If no attendant (this includes CAS, 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 10 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 DDC group member
- A TECT member
- A Trunk Answer From Any Station answering user
- A UCD 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.

- Call Vectoring (G3i)

  A call cannot be parked on a VDN extension. Also, a call that is undergoing vector processing cannot be parked.

- 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

  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

  If a call involves only one party and was parked by one of the following three methods, the parked user will hear music-on-hold.

  - With a Call Park feature access code
  - By pressing the Transfer button, the Call Park button, and Transfer button again,
  - With the Call Park button

  If a call involves only one party but was parked by pressing the Conference button, the Call Park button, and then the Conference button again, the parked user will hear music-on-hold.

  If a call involves multiple parties (such as a conference call) and was parked using the Call Park button or a Call Park feature access code, none of the parties will hear music-on-hold.

  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 Time-out interval (from one 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).

- 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 Time-out 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 800 Call Pickup groups can be established. Each group can have up to 50 members. However, a voice terminal can be a member of only one Call Pickup group.

When a member of a call pickup group is away from his or her voice terminal and receives an incoming call, the other members of the call pickup group receive no indication that a call is available for pickup (that is, their voice terminals do not ring and the green LEDs associated with their Call Pickup buttons do not flash). Hearing an unattended voice terminal ring is the only way group members know that a call is available for pickup. As a result, members of a call pickup group must be located close enough to one another so each member can hear each other's voice terminal ring. Otherwise, the Call Pickup feature is useless.

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 Prompting (G3i)

Description

Uses specialized vector commands along with the Call Vectoring feature to provide flexible handling of incoming calls based on information collected from the calling party.

Call Prompting (along with Call Vectoring) may be used in various applications to achieve better and more flexible handling of incoming calls. This description describes four Call Prompting applications. A brief description of each of the four sample applications is given below (a more detailed description of each application is given later in this chapter):

- Automated Attendant — Allows the calling party to enter the extension of the party that he/she would like to reach. The call is then routed to that desired extension.
- Data In/Voice Answer (DIVA) Capability — Allows the calling party to hear an announcement based on the digits that he or she enters.
- Data Collection — Allows the calling party to enter data which can then be used by a host/adjunct to assist in call handling. This data, for example, may be the calling party's account number.
- Message Collection — Gives the calling party the option of leaving a message or waiting in queue for an agent.

Since the Call Vectoring feature is used with Call Prompting, it is recommended that the Call Vectoring feature described elsewhere in this chapter be read and understood in order to more easily comprehend this Call Prompting description.

As in the Call Vectoring feature, VDNs are used to access Call Prompting vectors. Each VDN routes to a single Call Prompting vector but several VDNs may route to the same Call Prompting vector.

Call Prompting Vector Commands

The following list shows the complete set of specialized vector commands that are used with the Call Prompting feature. Each of these commands is described in detail in the paragraphs which follow the list:

- Announcement extension [annc. ext.]
- [Collect no. of digits] digits after announcement extension [annc. ext.]
- Goto step [step no.] [if digits equal [digits]]
- Goto vector [vector no.] [if digits equal [digits]]
- Messaging Split [split no.] for extension [ext.]
- Route to number [number] [if digit equals [digit]]
- Route to digits with coverage [y/]

- Stop

The following paragraphs describe in detail the specialized vector commands used with the Call Prompting feature. The **announcement**, **goto step**, **goto vector**, **messaging**, **route to number** and **stop** commands are used by both the Call Vectoring and the Call Prompting features.

Announcement extension [EXT]

For details on this command, see the Call Vectoring feature description elsewhere in this manual.

Collect [NO. OF DIGITS] digits after announcement extension [EXT]

In this command, [NO. OF DIGITS] is an administered number of digits from one through 16, and [EXT] is an administered announcement extension or "none" (digits are collected without a prompt).

With this command, the switch can collect up to 16 touch-tone digits from the calling party. An optional announcement (prompt) may be used to request the calling party to enter these digits. In addition, the announcement can instruct the user to enter an asterisk (*) if incorrect data is entered. When the calling party enters an asterisk (*), the digits collected for the current **collect** command are deleted, digit collection is restarted, and the announcement is not replayed.

When programming this command, the maximum number of digits requested of the calling party must be specified in the administration of the command. If the calling party can enter less digits than the maximum specified, the announcement should instruct the calling party to terminate the entry with a pound (#) sign digit as an end-of-dialing indicator. If less digits than the maximum specified are entered and the calling party does not complete the entry with a pound (#) sign, a 10-second inter-digit time-out will occur. The time-out terminates the command and any digits collected prior to the time-out are available for subsequent vector processing.

When digit collection for the current **collect** command completes, vector processing continues at the next vector command. However, the switch will continue to collect any subsequently dialed digits. These "dialed-ahead" digits are to be saved for use by subsequent **collect** commands and provide the calling party with a means to bypass subsequent unwanted announcement prompts. For example, a frequent caller to a service may not want to listen to all of the announcements in full.

The sum of the digits collected for the current **collect** command plus the dial-ahead digits must not exceed the switch storage limit of 16. Any additional digits dialed are discarded until storage is freed up by a subsequent **collect** command. These additional digits are only available for use by subsequent **collect** commands and are never used by other vector commands that operate on digits (such as **route to digits**, **goto...if digits**, and so on). In addition, these digits will not be displayed as part of the Caller Information button operation.

This command functions as follows:

1. A touch-tone receiver (TTR) may already be connected to the call (due to the processing of a previous **collect** command). If a TTR is not connected to the call (this is the first **collect** command encountered during vector processing), Answer supervision is returned to the calling party (if not already returned), and the call (with inter-digit timing disabled), is connected to a TTR, if a TTR is required for the collection of digits. Incoming rotary trunks and internal hybrid and rotary station sets, for example, do not require connection of a TTR. Note that external users with rotary sets cannot use Call Prompting:
   - If a TTR is not available and the TTR queue is not full, the call is queued for the next available TTR. Processing of this command will not continue until a TTR becomes available and is connected.
   - If a TTR is not available and the TTR queue is full, then vector processing continues at the next vector command (the **collect** command is unsuccessful).
2. Once a TTR has been connected to a call, if dial-ahead digits are available, the system reacts as follows:
   - The connection of the announcement prompt (if administered) is skipped.
   - The 10-second inter-digit timer is started.
   - The dial-ahead digits are analyzed up to the maximum number specified for this **collect** command or up to the first pound (#) digit. If an asterisk (*) is found, the digits up to and including the * are deleted and any additional digits are again analyzed. Any remaining dial-ahead digits are to be saved for use by a subsequent **collect** command.
3. Once a TTR has been connected to a call, if dial-ahead digits are *not* available and a valid announcement extension has been administered for the **collect** command, the system reacts as follows:
   - If the announcement to be connected is busy (there are no available announcement ports) and the queue for the announcement is full or there is no queue, the calling party will continue to hear the current feedback. The system will then wait five seconds and try again to connect the call to the announcement. This process continues until the call is successfully queued or connected to the announcement, or the calling party disconnects from the call.
   - If the announcement to be connected is busy (there are no available announcement ports) and the queue for the announcement is not full, the call is queued for the announcement.
   - If an announcement port is available (either initially or after system retry), or if the queued request for the announcement has been fulfilled, any previous calling party feedback is removed from the call, and the calling party is connected to the announcement.

4. If no announcement extension is administered for the **collect** command or if the announcement extension is not administered within the system; the calling party receives silence (the system attempts to collect digits without the prompt), and the 10-second inter-digit timer is started and answer supervision is returned.

5. If an announcement was connected in one of the previous steps, the system reacts as follows:

   - If the calling party enters a touch-tone digit prior to the completion of the announcement, the call is disconnected from the announcement, the 10-second inter-digit timer starts, and the collected digit is analyzed as described in step 6.

   - If the announcement completes before the calling party enters a touch-tone digit, the system starts the 10-second inter-digit timer.

6. At this point the system is doing digit collection and analysis. In addition the inter-digit timer is activated. The system continues digit collection for this command until *one* of the following occurs:

   - The maximum number of digits specified has been collected
   - A pound (#) digit is collected (signifying end of dialing)
   - The inter-digit timer expires.

   The system then analyzes the collected digits and reacts as follows:

   - If the digit is an * (signifying an error was made while entering digits), the system deletes all digits collected for the current **collect** command and restarts the 10-second inter-digit timer. The announcement is not replayed.

   - If the calling party has entered the maximum number of digits specified, if the digit is a pound (#), or if the 10-second inter-digit timer expires, then the inter-digit timer is disabled, and vector processing continues at the next vector command (the **collect** command is successful). However, the switch will continue to collect any subsequent dialed digits (including # and * digits) to allow for the dial-ahead capability. These additional "dialed ahead" digits are saved for use by subsequent **collect** commands.

Goto step [STEP #] if digits equal [DIGITS]

In this command, [STEP #] is an assigned step number from one through 15. Digits (in the **if digits equal** part of the command) refers to the digits entered by the calling party (via the last **collect digits** command), and [DIGITS] is a digit string from one through 16 digits in length.

The **if digits equal [DIGITS]** parameter of this command is optional. The unconditional form of the command causes vector processing to continue at the specified step (command). With the **if digits equal [DIGITS]** parameter, this command causes vector processing to continue at the specified step (command), if the digits entered by the calling party for the last **collect digits** command are the same as the digits administered in the digit string of this command. This command is used for both skipping vector commands and looping through vector commands.

If the command is unconditional or if the digits entered by the calling party for the last **collect digits** command are exactly equal to the digits in the administered digit string, then vector processing continues at the specified step. If neither of these conditions are met, vector processing continues at the next vector command.

Goto vector [VECTOR #] if digits equal [DIGITS]

In this command, [VECTOR #] is an assigned vector number from 1 through 256. Digits (in the **if digits equal** part of the command) refers to the digits entered by the calling party (via the last **collect digits** command), and [DIGITS] is a digit string from 1 through 16 digits in length.

The **if digits equal [DIGITS]** parameter of this command is optional. The unconditional form of the command causes vector processing to continue at the specified vector. With the **if digits equal [DIGITS]** parameter, this command causes vector processing to continue at the specified vector, if the digits entered by the calling party for the last **collect digits** command are the same as the digits administered in the digit string of this command. This command is useful for applications that require more than 15 vector commands.

If the command is unconditional or if the digits entered by the calling party for the last **collect digits** command are exactly equal to the digits in the administered digit string, then vector processing continues at the first step in the specified vector. If neither of these conditions are met, vector processing continues at the next vector command within the original vector.

Messaging Split [SPLIT #] for extension [EXT]

For details on this command, see the Call Vectoring feature description elsewhere in this manual.

Route to number [NUMBER]

For details on this command, see the Call Vectoring feature description elsewhere in this manual.

Route to number [NUMBER] if digit equals [DIGIT]

In this command, [NUMBER] is the same as [NUMBER] in the previously described **route to number [NUMBER]** command, **digit** (in the **if digit equals** part of the command) refers to the digit entered by the calling party (via the last **collect digits** command), and [DIGIT] is a single administered digit from 0 through 9.

This command is simply a conditional **route to number** command. It allows a call to be conditionally routed to a specified destination based on a single digit entered by the calling party. In other words, if the digit entered by the calling party in the last **collect digits** command is the same as the administered digit, then the command will attempt to route the call to the specified destination. If more than one digit was collected when this command is encountered, it will fail and vector processing will continue at the next command. Everything that applies to the **route to number** command also applies to this command.

This command can only be used for routing based on single-digit comparisons. If a customer application requires comparisons based on more digits, then the appropriate combination of a **goto** and **route to number** command must be used.

Route to digits with coverage [y/n]

In this command, **with coverage [y/n]** refers to whether coverage should apply when routing, and **digits** refers to the digits entered by the calling party (via the last **collect digits** command). These digits represent a destination which may be any of the following:

- An internal extension (such as a split/hunt group, voice terminal, announcement, and so on)
- A VDN extension
- An attendant
- A remote extension
- An external number such as a TAC or AAR/ARS FAC followed by a public or private network number

This command allows a call to be routed to the destination specified by the digits entered by the calling party as a result of the last **collect digits** command. This command can be used to implement an automated attendant function. The optional coverage parameter determines whether coverage should apply when routing. If coverage applies and the digits are a system extension, vector processing terminates as soon as this command is encountered. If coverage doesn't apply, this command is identical to the **route to number** command with respect to coverage and success/failure of the vector command. In either case, the interactions with calling party restrictions are the same as in the **route to number** command.

This command functions as follows:

1. If the number of digits collected by the system is 0, vector processing continues at the next vector command.
2. If the coverage option is administered, and the extension is a system extension (not a VDN), the following happens:
    - The system provides ringback to the calling party.
    - Vector processing terminates.
    - Normal termination and coverage apply to this call. In other words, the call is treated as a normal calling party to extension call. All coverage criteria apply.

3. If coverage does not apply, this command functions the same as the previously described **route to number** command.

Stop

This command is used to terminate processing of any subsequent vector commands in the vector. In addition, if a TTR is allocated to the call when the **stop** command is encountered, the TTR is disconnected since there is no purpose in allowing digits to be entered when subsequent commands (such as **collect digits**) will no longer be processed. Current call processing, however, will continue (the call will not be dropped). The calling party will continue to hear the current feedback that was on the call when the stop was encountered. In addition, the stop command will not affect the queued status of a call (for example, if the call was in a split's queue when the stop was encountered, it will remain in the split's queue after the stop has been processed). Of course this is only the case if vectoring has been enabled. An implicit stop command is processed, if necessary, following the last administered command in a vector.

Vector Processing and Calling Party Feedback

Vector processing starts when a call routes to a VDN. The called VDN then passes control of the call to its assigned vector. Processing begins at step one in the vector and proceeds sequentially through the vector unless a **goto** command is encountered. Unadministered steps are skipped and a **stop** command is processed after the last administered command within a vector.

Calling party feedback is provided by the vector. However, incoming CO, FX, and WATS calls hear CO ringback until answer supervision is supplied (at which point vector feedback is applied). The initial calling party feedback for non-CO/FX/WATS (DID type)calls is silence. This silence continues until the vector processes a command which alters this feedback. Some commands that change feedback are **announcement** commands, which apply a recorded announcement, and **messaging** and **route to** commands which apply ringback. More details on calling party feedback for vectoring are found in the Call Vectoring feature described elsewhere in this chapter.

The Caller Information (CALLR-INFO) Button

The CALLR-INFO button displays information in the following format: "x=Info:1234567890" where x is a call appearance letter (a, b, c, and so on) and 1234567890 are the digits collected for the last **collect digits** command. Any digits that were "dialed ahead," and not asked for explicitly by the most recently executed **collect digits** command, will not be displayed as a result of a CALLR-INFO button depression.

When digits have been collected via Call Prompting, and an attendant or display-equipped voice terminal user presses the CALLR-INFO button, the display is updated with the information described in the previous paragraph. This information will remain displayed for 10 seconds, unless an incoming call is received, or the active call changes status (for example, another party is added to a conference). In this case, the display changes to show the new call identification information.

If the answering agent needs to display the collected digits again, the CALLR-INFO button can be depressed again to repeat the above operation (as long as the agent is active on the call or the call is still alerting). If the call is on hold, the CALLR-INFO button cannot be used to display information for an alerting call until the call is answered.

The CALLR-INFO button works when a call is alerting a voice terminal or attendant console and after a voice terminal user or attendant has answered a call. If no digits were collected, the attempt is denied.

Call Prompting Applications

Automated Attendant

The Automated Attendant application allows customers, particularly those who do not have DID trunks, to route incoming calls to a desired location without the use of an attendant. This allows the customer to reduce costs by reducing the need to employ live attendants. A sample prompting vector that implements the Automated Attendant application is shown in the following screen.

In the following example the calling party is prompted to enter the destination extension (up to five digits) of the party he/she would like to reach. For example, the announcement at extension 300 might say, "Enter the five-digit extension of the party you wish to reach." The illustrated vector will collect the digits and then route to the destination. If the **route to digits** command fails (for example, the calling party is an outside rotary user, the **route to number** command will execute and route the call to the attendant (default). On the other hand, as long as the destination is a valid system extension, the **route to digits with coverage y** will succeed, coverage will apply, and vector processing will terminate. (Even if the destination is busy, vector processing will terminate since call coverage processing will take effect.)

```
                                                          Page 1 of 1
                                   CALL VECTOR
      Number: 2              Name Auto Attendant

     ASAI Routing? y                    Basic? y         Prompting? y
   01 collect 5 digits after announcement extension 300
   02 route to digits with coverage y
   03 route to number 0
   04 stop
   05 ______________
   06 ______________
   07 ______________
   08 ______________
   09 ______________
   10 ______________
   11 ______________
   12 ______________
   13 ______________
   14 ______________
   15 ______________
```

DIVA

The Data In/Voice Answer (DIVA) application allows a calling party to receive information on a topic selected at a prompt. In addition, DIVA provides a means to partition different groups of users so that their calls may be handled more efficiently. A sample prompting vector that implements the DIVA application is shown in the following screen.

In the following example the calling party is prompted, by announcement extension 310, to enter a **1** or **2** to receive information on two different subjects. Then one of the following occurs:

- If the calling party fails to enter a **1** or **2** , an announcement (extension 320) is connected (with an explanation of options) and the calling party is returned to the initial prompt.
- If a **2** is entered, the calling party receives the chosen information (announcement ext. 313) and vector processing terminates.
- If a **1** is entered, the calling party is again prompted by announcement extension 315 to enter a **1** or **2** to further identify the type of information that is being requested. At this prompt, if either a **1** or **2** is entered, the calling party receives an announcement on the chosen topic. If anything else is entered, the calling party is routed to an agent to receive additional information or assistance. (Extension 50000 is a split extension.)

  In addition, the calling party may utilize the Dial-Ahead capability and enter the digit string **1 1** or **1 2** at the initial prompt. This will allow the calling party to bypass the secondary prompt and get to the information he/she desires more quickly.

```
                                                        Page 1 of 1
                              CALL VECTOR
  Number: 4             Name DIVA

 ASAI Routing? y                   Basic? y         Prompting? y
01  collect 1 digits after announcement extension 310
02  goto step 6 if digits 1
03  goto step 15 if digits 2
04  announcement extension 320
05  route to number 0
06  collect 1 digits after announcement extension 315
07  goto step 11 if digits 1
08  goto step 13 if digits 2
09  route to number 50000
10 stop
11 announcement extension 311
12 stop
13 announcement extension 312
14 stop
15 announcement extension 313
```

Data Collection

The Data Collection application provides the system with a method to collect digits from a calling party which can be used by an adjunct to assist in the handling of the call. For example, the calling party could be prompted to enter an account number which could be used by an adjunct to retrieve calling party account information. This information could then be displayed on a data terminal screen to be used by an agent. This capability can be fully automated using an integrated adjunct via the ICM feature. Alternatively, the Caller Information (CALLR-INFO) button is available, which allows the prompted information to be displayed on the answering party's voice terminal. The answering party can then use this information to manually retrieve calling party account information via an external adjunct or host computer. A sample prompting vector that implements the Data Collection application is shown in the following screen.

In the following example the calling party is prompted to enter a 10-digit account number representing his or her customer account. The vector then routes the call to a split to be answered by an agent. Using the CALLR-INFO button, the answering agent can display this account number and use it to retrieve calling party account information from the customer's host computer.

```
                                                          Page 1 of 1
                                CALL VECTOR
      Number: 10             Name_Data Collection____

     ASAI Routing? y                  Basic? y         Prompting? y
    01 collect 10 digits after announcement extension 300
    02 queue to main split 3 priority m
    03 stop
    04 ______________
    05 ______________
    06 ______________
    07 ______________
    08 ______________
    09 ______________
    10 ______________
    11 ______________
    12 ______________
    13 ______________
    14 ______________
    15 ______________
```

Message Collection

The Message Collection application gives the calling party the option of not waiting to be serviced by an agent, but instead leaving a message for the agent or the agent's associated split. For example, a customer with both Call Vectoring and Call Prompting enabled can let a calling party waiting in a split queue be given a choice (via a prompt) of remaining in queue or leaving a message for that split (or agent).

Message collection allows a calling party who does not have time to wait to be serviced to leave a message (name, phone number, reason for calling, and so on) so that an available agent may return the call at a later time. If the customer prompt requested that certain information be left in the message, a return call may not even be necessary. A sample prompting vector that implements the Message Collection application is shown in the following screen.

In the following example, the calling party is prompted to enter a **1** to be placed on the company mailing list or enter a **2** to speak with an agent for more information about the company. If anything other than a **1** or **2** is entered, an announcement is connected with a further explanation of the options, and the calling party is returned to the initial prompt. If a **1** is entered, the calling party is connected to AUDIX so that he or she may leave the pertinent information and be placed on the company mailing list. If the switch cannot connect to AUDIX (for example, if the AUDIX link is down), an announcement is played informing the caller to try and call back later. If a **2** is entered, the call is routed to a split to be answered by an agent to receive additional information about the company.

```
                                                    Page 1 of 1
                                  CALL VECTOR
   Number: 150                    Name Msg Collection
  ASAI Routing? y                        Basic? y         Prompting? y
 01  collect 1 digits after announcement extension 300
 02  goto step 6 if digits 1
 03  goto step 9 if digits 2
 04  announcement extension 301
 05  route to number 0
 06  messaging split 20 for extension 50000
 07  announcement extension 302
 08  stop
 09  queue to main split 3 priority m
 10  stop
 11 ______________
 12 ______________
 13 ______________
 14 ______________
 15 ______________
```

Considerations

With Call Prompting, a caller is prompted to enter information from his or her touch-tone phone. The system then uses this information to route the call (if necessary) to the right person or group of persons. The call may also be routed to an announcement, if desired. In addition, a caller may be asked to enter an account number or some other type of number to be used in handling the call. Since the system retrieves all of this type of information, and processes the call accordingly, time is not wasted trying to determine the type of call and to whom it is supposed to go. Also, agents spend less time gathering information and can handle more calls.

The system provides a maximum of 256 vectors, with a maximum of 15 steps (commands) per vector. These 256 vectors can be made up of any combination of prompting vectors and non-prompting vectors.

The maximum number of VDNs is 500.

The maximum number of integrated and analog announcements is 128.

Integrated announcements provide a maximum of eight and one half minutes recording time.

The maximum number of calls that can be connected simultaneously to one integrated announcement port is five.

A maximum of 80 Touch-Tone Receivers is allowed.

The **Route to number [number] if digit** command can only be used to conditionally route a call based on a single digit comparison.

Vector processing will execute a maximum of 1,000 vector commands for a given call. After executing the one-thousandth command, an implied **stop** command is executed.

Interactions

The following features and functions interact specifically with the Call Prompting feature. Interactions with Call Vectoring are found in the Call Vectoring feature description:

- Answer Detection

  Call Prompting competes with Answer Detection for ports on the TN744.

- Answer Supervision

  Answer supervision is only returned once during the life of a call. With respect to prompting commands, answer supervision is returned in response to a **collect digits** or an **announcement** command. In addition, if a call is answered and answer supervision hasn't previously been sent, it will then be sent.

- AUDIX

  If a **route to** command in a vector calls AUDIX, the call is treated as a direct call to AUDIX and the calling party may retrieve his or her messages.

- Authorization Codes

  Authorization codes are disabled with respect to routing via VDNs. In other words, if authorization codes are enabled, a **route to** command in a prompting vector accesses AAR or ARS and the VDN's FRL does not have the permission to utilize the chosen routing preference, then no authorization code is prompted for and the **route to** command will fail.

- AAR/ARS

  Any **route to** command in a vector can dial an AAR/ARS FAC followed by other digits.

- AAR/ARS Partitioning

  When a **route to** command in a vector dials an AAR or ARS FAC, the COR associated with the VDN is used to determine the AAR/ARS PGN. The PGN then determines the appropriate routing tables to use for the call.

- Automatic Incoming Call Identification

  When a call terminates to a display station via a **route to number** or **route to digits with coverage n** command, the station will display the following information: `Originator Name to VDN name`.

- Bridging

  If a principal extension with bridged appearances receives a call via a

route to command, that extension's bridged appearances will be updated to reflect the state of the call on the principal extension's call appearance.

- CallVisor ASAI

  Call Prompting competes with CallVisor ASAI switch-classified calls for ports on the TN744.

- Call Coverage

  When a **route to digits with coverage y** command terminates to a system extension (not a VDN), then vector processing terminates and normal routing, termination and coverage apply to this call. Coverage does not apply to all the other forms of the **route to** command. In other words, if a call terminates successfully at the destination, vector processing terminates and coverage is disabled. If the **route to** command fails, vector processing continues at the next vector command.

- Call Forwarding All Calls

  A **route to** command in a vector cannot dial the Call Forwarding All Calls FAC.

  If a **route to** command in a vector calls a system extension (split/hunt group, voice terminal extension, and so on) that has Call Forwarding All Calls active, the call is forwarded to the designated (forwarded to) destination.

  Calls that are forwarded to a VDN are considered successfully forwarded and call coverage is disabled. (For example, if a **route to digits with coverage y** command is performed within a vector that has been accessed via call forwarding, coverage does not apply and the command is treated as a **route to digits with coverage n** command.)

- Call Pickup

  If a **route to** command calls a voice terminal extension that is a member of a pickup group, that call can be picked up by another pickup group member.

- Call Waiting Termination

  When calls are routed to analog voice terminals via a **route to** command (except **route to digits with coverage y**), the call is considered successful if the analog extension is idle. If it is not idle, the **route to** is considered unsuccessful and call waiting does not apply. In other words, call waiting is disabled for these **route to** commands. On the other hand, for a **route to digits with coverage y** command, call waiting does apply if appropriate.

- Call Vectoring

  Call Prompting is administered through Call Vectoring administration. If only Call Vectoring is enabled, vectors can be administered using only Call Vectoring commands. If only Call Prompting is enabled, vectors can be administered using only Call Prompting commands (described elsewhere in this chapter). When both features are enabled, all the commands associated with vectoring and prompting are available.

Enabling both vectoring and prompting together provides the capability to prompt the caller to enter pertinent data while the caller is waiting in queue at an ACD split. For example, prompting and vectoring can be used together to enhance the message collection capability to provide a caller who is waiting in queue at an ACD split with the choice of remaining in queue or leaving a message for that split. In addition to queuing, vectoring provides the capability to change calling party feedback. In addition to silence, which both prompting and vectoring supply, vectoring can provide ringback or music feedback to the calling party.

The implications of having both features enabled are the following:

- When a Call Prompting command (such as a successful **route to** command) terminates vector processing, the call must be dequeued and dropped from all queues that it is currently residing in as part of the termination process.
- If a call is waiting in an announcement queue, waiting to be connected to an announcement or an announcement queue (announcement retry), or is currently connected to an announcement, and the call is dequeued from a split's queue and terminates to an agent's voice terminal extension, the announcement is disconnected and ringback is connected to the call.
- If a **collect digits** command is being processed for a call and the call is dequeued from a split's queue and terminates to an agent's voice terminal extension, the **collect digits** command is terminated and ringback is connected to the call. Digit Collection is not recommended while a call is in queue, because an answering agent could interrupt the process and hear the dialing.

- CAS

  If a **route to** command calls the attendant and CAS is enabled, the call completes to the CAS attendant if an RLT trunk can be seized.

- Coverage All Calls

  The **route to digits with coverage y** command works like any other call. All other types of **route to step** commands fail.

- Coverage Callback

  Coverage Callback only operates successfully for **route to digits with coverage y** commands that terminate successfully to a coverage point. The covering user is able to initiate coverage callback for the principal.

- Coverage Incoming Call Identification

  If a coverage call terminates at a covering user's display-equipped voice terminal, via a **route to digits with coverage y** command, the voice terminal user that the vector routed to is displayed as the called party instead of the VDN.

- DCS

A **route to** command in a prompting vector can route to a UDP extension and provide DCS transparency (Distinctive Alerting). In addition, DCS Call Forwarding, DCS Call Waiting, DCS Leave Word Calling, and DCS Conference/Transfer apply, where appropriate, to calls routed via a **route to** command.

- DOD

  DOD can be provided via a **route to** command within a vector. The COR of the VDN is used to determine calling party permissions/restrictions.

- FRL

  If a **route to** command dials an external number via AAR/ARS, the FRL associated with the VDN COR is used to determine the accessibility of a routing preference in an AAR/ARS pattern.

- Go To Cover

  Go To Cover operates correctly when a **route to digits with coverage y** command terminates successfully at an internal destination. At all other times, if the go to cover button is depressed, the feature is denied.

- Hold

  If a call is put on hold during the processing of a collect command, the collect command will be restarted, beginning with the announcement prompt, when the call is taken off hold. All dialed-ahead digits will be lost. Similarly, if a call to a vector is put on hold, vector processing will be suspended when a collect step is encountered. When the call becomes active, the collect step will resume.

- Hunting

  A **route to** command can call a hunt group.

- ICM

  The Call Prompting feature can be used to collect information from a calling party which may later be used by an adjunct to assist in the handling of the call.

- Individual Attendant Access

  A **route to** command can dial an individual attendant extension.

- Inter-PBX Attendant Calls

  If a **route to** command calls the attendant and this feature is enabled, the call will complete to the Inter-PBX attendant.

- ISDN-PRI

  A **route to** command in a prompting vector can route calls over ISDN-PRI trunks.

- LWC

  LWC will operate correctly when a **route to digits with coverage y** command terminates successfully at an internal destination. At all other times, if the LWC button is depressed, activation is denied.

- Night Service

  Route to commands that route to destinations with night service activated, redirect to the night service destinations.

- Priority Calling

  A **route to** command cannot dial the priority calling FAC.

- Queuing

  Queuing applies, where appropriate, to any calls that route to an attendant or hunt group via a **route to** command.

- Recorded Announcements

  Recorded Announcements can be accessed via a VDN through the use of the **announcement** command or the **route to** command, if the destination is an announcement extension. In addition, the **collect digits** command has the option to connect an announcement when prompting for digits.

- Redirect Notification

  Redirect notification applies when a call is about to redirect to coverage via a **route to digits with coverage y** command.

- Remote Access

  A **route to** command cannot dial the Remote Access extension.

- Ringback Queuing

  External call attempts made via **route to** commands (except for **route to digits with coverage y**)will not queue via Ringback Queuing when all trunks are busy.

- Rotary Dialing

  Outside users using rotary dialing will not be able to enter digits requested via a **collect digits** command. The 10-second inter-digit time-out takes effect and a **collect digits** command is skipped for these users. With this in mind, a default **route to** command (such as **route to attendant**) should always be provided.

- SAC

  When a **route to digits with coverage y** command terminates to a system extension with SAC active, the call is treated as a normal internal call to a station having SAC active (call coverage via SAC applies). When any other type of **route to** command terminates to an extension having SAC active, SAC is ignored. If the station has an idle appearance, the call terminates and the **route to** is successful. Otherwise, the **route to** command is considered unsuccessful and vector processing continues at the next vector command.

  If a **route to digits with coverage y** command terminates to a system extension and the station user then activates SAC, the system attempts to redirect the call to coverage (due to SAC coverage criteria). For any other **route to** commands that terminate to a system extension, activation of SAC by the station user (after termination) is ignored.

- CDR Account Codes

  A **route to** command cannot dial the CDR account code FAC.

- Subnet Trunking

  Subnet trunking applies to any AAR/ARS call dialed via a **route to** command.

- Temporary Bridged Appearance

  A Temporary Bridged Appearance is maintained at the principal's extension when a **route to digits with coverage y** command terminates to a system extension and redirects to coverage. However, if coverage is to AUDIX or the principal is an ACD agent, no Temporary Bridged Appearance is maintained at the principal.

- TEG

  A **route to** command can call a TEG.

- Transfer

  If a call to a VDN is transferred during a **collect** command, the **collect** command will be restarted when the transfer is complete, and all dialed-ahead digits will have been lost. Similarly, if a call to a vector is transferred, vector processing will be suspended when a **collect** step is encountered. When the transfer is complete, the **collect** step will resume. This also applies to attendant extended calls

- TCM

  A TCM is sent when a **route to** command dials a seven-digit ETN or 10-digit DDD number via AAR/ARS. This TCM is the FRL associated with the VDN COR.

- UDP

  A **route to** command can call a UDP extension.

Administration

Call Prompting is administered on a per-system basis by the System Manager. The Call Vectoring feature must be administered as described in the Call Vectoring feature described elsewhere in this chapter. In addition, the following items must be administered specifically for Call Prompting:

- The Call Prompting feature must be enabled on the System-Parameter Customer-Options form. This must be done by an authorized AT&T employee.
- Any display-equipped voice terminal or attendant can be administered with a Caller Information (CALLR-INFO) button.

Hardware and Software Requirements

Each Call Prompting announcement requires one port on a TN750 Integrated Announcement circuit pack or announcement equipment and one port on a TN742 or TN746B (A-law) Analog Line circuit pack.

Touch-tone receivers are required to accept the touch-tone digits entered by the Call Prompting users. The TN744 Call Classifier circuit pack (required for Call Prompting) provides eight touch-tone receivers. The system allows a maximum of 80 TN744 touch-tone receivers (ten circuit packs). TN744 touch-tone receivers are not used as general purpose touch-tone receivers in the system. Other touch-tone receivers such as those on the TN748C circuit pack (TN420C, TN744 support A-law) are still required for normal call processing and maintenance features.

Call Prompting software is required.

Call Vectoring (G3i)

Description

Provides processing of incoming and internal calls according to a programmed set of commands. The commands, called Vector commands, determine the type of processing that specific calls will receive. Vector commands may direct calls to on-premise or off-premise destinations, to any hunt group or split, or to a specific call treatment such as an announcement, forced disconnect, forced busy, or delay treatment.

It is possible for the system to collect digits from the user, route calls to a destination specified by those digits, and/or do conditional processing according to those digits (Call Prompting feature). The Call Prompting feature utilizes the Call Vectoring feature and a set of specialized vector commands. Also, the Look-ahead Interflow feature uses Call Vectoring for its operation.

Vector Directory Numbers and Vectors

Calls access vectors using VDNs. A VDN is a "soft" switch extension number that is not assigned to a physical equipment location. A VDN has several properties. These properties, administered by the System Manager VDN on the administration form, are listed below:

- Extension Number — This is the extension number used to identify the VDN.
- VDN Name — This is the name (up to 15 characters) that is associated with the VDN. The VDN Name is shown on the agent's display. This information is optional.
- COR — This is a one- or two-digit number that specifies the COR of the VDN.
- Display Override — This affects the operation of an agent's display when a call is routed through several VDNs. If administered as "no," the name of this VDN appears on the agent's display. If any subsequent VDNs are used to process this call, their names will not appear on the terminating display. If administered as "yes," the name of the VDN appearing on the terminating display will depend on the administration and chaining of the subsequent VDNs.
- Vector Number — This number (between 1 and 256) determines which vector is activated when a call comes in to a VDN. Several VDNs may send calls to the same vector.

Access to a VDN may occur in many ways. Since a VDN is an extension, it can be accessed in almost any way that an extension can be accessed. The primary ways that a VDN can be accessed are as follows:

- Internal call — The VDN extension can be dialed from another extension on the switch.

- CO trunk — A CO trunk may be mapped to a VDN (as an incoming destination or night service extension).
- Non-CO trunk — A call may come into a DID trunk and connect to a VDN extension.
- LDN — A VDN may be the night destination for an LDN.

A Call Vector is a set of up to 15 vector commands (described later in this chapter) to be performed for an incoming or internal call. As previously described, vectors are accessed by VDNs. Each VDN maps to one vector. However, several VDNs may map to the same vector. The answering user will see the information (such as the name) associated with the VDN on his or her display and can respond to the call with knowledge of the dialed number. This operation provides DNIS (described later in this description).

Applications

There are many different applications for the Call Vectoring feature. However, Call Vectoring is primarily used to handle the call activity of ACD splits. Call Vectoring can also manage a queue by keeping calls queued in up to three splits (with four different priority levels) while also providing a series of other processing options. Other common applications include:

- Special treatment for selected callers.

  For example, calls from preferred credit card customers may receive priority treatment, but they do not have to be handled by a separate split. Agents in the same split can handle both preferred customers and all other customers. A call can be queued into one of four priority levels, and calls to different VDNs (and vectors) could go to different levels, with preferred customers having top priority. This means that when all agents are busy in this split, calls from preferred customers would go to the top of the queue ahead of other callers already in the queue.

- Night treatment.

  During non-business hours, the Call Vector could route calls to a specified destination such as an announcement and a disconnect. During business hours, the vector could send calls to splits for connections with agents, or queue normally.

- Off-loading periodic excess calls resulting from promotions, seasonal trends, or regular daytime fluctuations in calls.

  A vector can test a split for the number of calls already in queue. If the number is above a certain threshold, the vector will bypass that split and route the call to someone else, such as an attendant. However, if the number of calls queued to the split is below the threshold, the vector would queue the call to that split.

- Information Announcements for the Calling Party

  The human intervention needed to distribute common messages can be minimized with information announcements. A group of people with a common interest can be instructed to call a specific number (VDN) that terminates to a specific announcement vector. The vector's announcement can be periodically updated to provide current information to the callers. Vectors providing information announcements are easily programmed by the System Manager.

What Happens When a Call is Processed by a Vector

General

When an incoming or internal call goes to a VDN, the VDN directs the call to a specific call vector. When the call goes to a Call Vector, the call's routing and treatment is determined by the commands in that vector. Processing starts at step one and proceeds sequentially through the vector unless a **goto** command is encountered. Any steps that have been left blank during administration will be skipped. The process automatically stops after the last step in the vector.

Call Vectoring allows the chaining of vector steps and vectors through the use of a **goto** command. In other words, one vector can direct the call to another vector or VDN, which can in turn direct the call to yet another vector, and so on.

The G3i Call Vectoring feature has an execution limit of 1,000 steps. Once a call enters vector processing, a loop counter will keep track of the number of vector steps executed. If the loop counter exceeds 1,000, a **stop** command is executed. The caller continues to hear the tone feedback currently in effect. Calls remain in any queues. The loop counter remains in effect across **goto vector** commands. This execution limit is provided as a means of system recovery from vector programmed infinite loops.

Three examples of how Call Vectoring functions are shown in the following figure. An outside call to a VDN is processed based on the various properties of the VDN form. Handling of the call is based on the commands associated with the vector. Displayed information is dependent on the setting of the `Display Override` field.

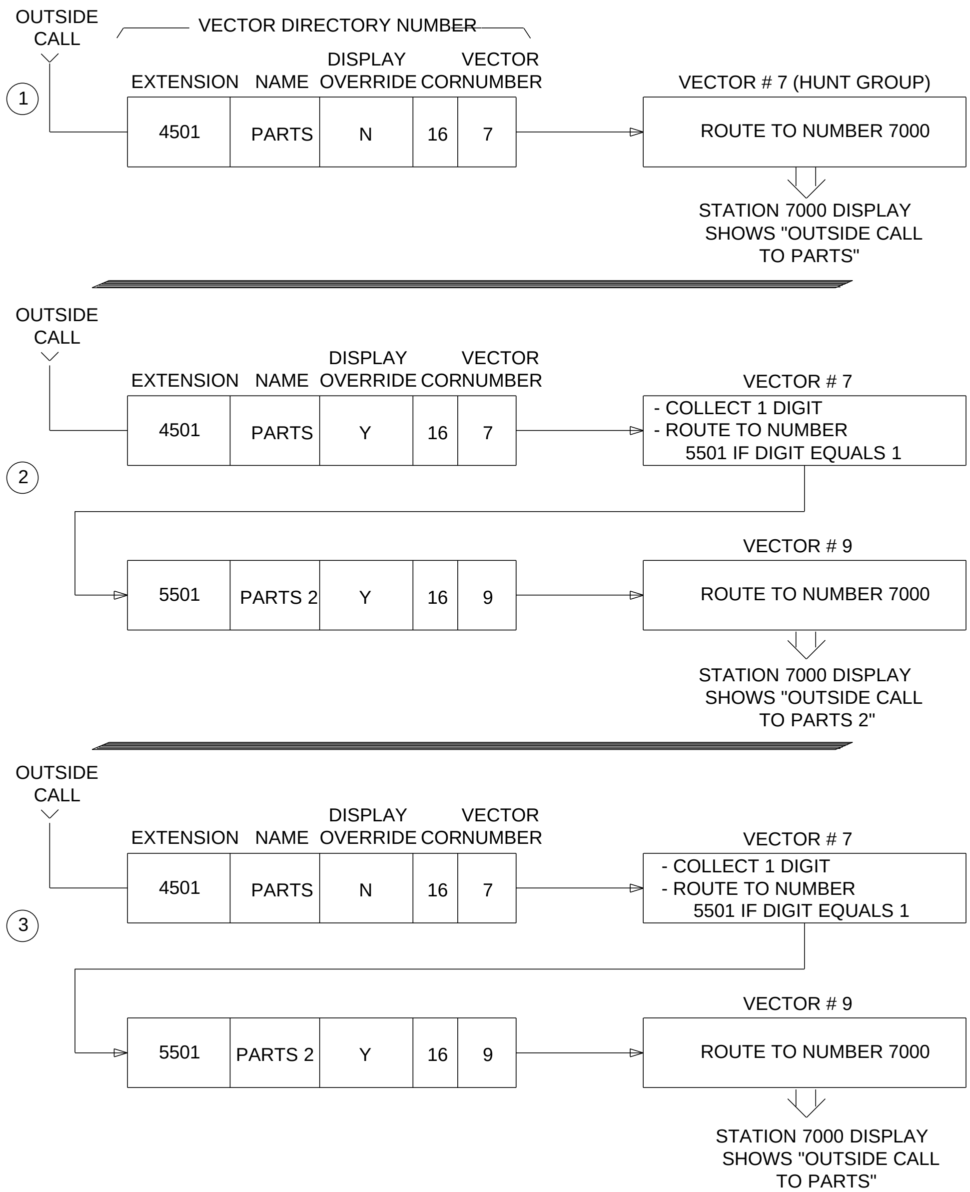

Figure 2-10. Typical VDN Call Processing Examples

Calling Party Feedback

The feedback that a caller will hear as a call is processed by a vector depends on how the call originates. The call may originate as one of the following:

- An internal call from another system user
- An incoming call over a non-CO (DID) trunk
- An incoming call over a CO trunk

If the call is an internal or non-CO call, the calling party will hear silence until one of the following occurs:

- An announcement is played
- A **wait** command with music or ringback is processed
- A **busy** command is processed
- The call is alerting at an station

If the call is a CO call, the calling party will hear CO ringback until one of the following occurs:

- An announcement is played
- A **wait** command with music is processed
- The call is answered

Additional Information

If the call is a CO call and answer supervision has been previously supplied (by processing of an announcement or wait with music command), the caller will hear the same thing as a caller on any other call:

- An announcement when any announcement command is processed
- Ringback, silence, or music when a **wait** command is processed
- Busy when a **busy** command is processed
- Ringback when the call is alerting an extension

After a **wait** command is processed, the calling party continues to hear that specified treatment unless an announcement is being played, another **wait** command with a different treatment is processed, the call is alerting at an extension, a **busy** command is processed, or vector processing terminates.

If a step cannot be executed, the calling party continues to receive the feedback to which they are currently listening and vector processing continues at the next step.

When a call terminates to an agent or extension, the calling party hears ringback. Any announcement or other treatment that is currently being heard is disconnected and ringback is supplied.

After an announcement completes, the calling party hears silence until another command that specifies treatment is processed (for example, wait, announcement, forced busy), or the call terminates at an extension.

If a call is routed off-premises via a trunk, the calling party hears the standard call progress tones and/or far-end feedback.

If the calling party disconnects at any time while the vector is being processed, vector processing is terminated, the call is removed from any queues, and the call is torn down.

Vector Commands

A Call Vector is a set of up to 15 vector commands. Vector commands determine the type of processing that specific calls will receive. The basic vector commands are listed below and described in the paragraphs that follow. Please note that most of these commands require one or more parameters (for example, an extension number is required with the Announcement command). For more information on these commands and how they should be administered, refer to *DEFINITY® Communications System Generic 3i — Implementation,* 555-230-650:

- Adjunct Routing (used only with the ICM feature)
- Announcement
- Busy
- Check Backup Split
- Collect (used for Call Prompting)
- Disconnect
- Goto Step
- Goto Vector
- Messaging
- Queue To Main Split
- Route To Number
- Route to Digits (used for Call Prompting)
- Stop
- Wait

Adjunct Routing [EXT]

In this command, [EXT] is the CallVisor ASAI link extension through which the switch will send a routing request to the associated adjunct.

The **adjunct routing** command is only available with ICM feature. This command causes a message to be sent to an adjunct that is requesting routing instructions. This command is described in detail in the Inbound Call

Management feature description, elsewhere in this chapter.

Announcement extension [EXT]

In this command, [EXT] is an assigned announcement extension.

The **announcement** command is used to let callers hear a given announcement. All calls will hear the announcement from the beginning. This may result in a delay because the call may have to wait in an announcement queue before the announcement is heard. While a caller is waiting to hear the announcement, the caller will continue to hear whatever treatment he or she is currently hearing.

The announcement is treated as a forced announcement. If the announcement's queue is currently full, the call retries the announcement step indefinitely before processing any new vector steps. If the requested announcement is not in-service, the current step is skipped and vector processing continues at the next vector step.

When a call is connected to the announcement, the previous treatment is discontinued and answer supervision is sent if it has not already been supplied. During an announcement, if the call is moved from waiting in a split's queue to alerting at an agent's extension, the announcement is disconnected and the caller hears ringback. When the announcement completes and is disconnected, the caller hears silence until a vector step with an alternate treatment is processed or the call reaches an agent's extension.

Busy

The **busy** command causes termination of vector processing and gives the caller a busy signal. The busy takes effect on non-CO trunk calls whether or not answer supervision has been sent; but if the call is a CO trunk call and answer supervision (via announcement command or wait command with music) has not been sent, the CO will ignore the busy and continue giving the caller CO ringback. Non-CO trunks will be dropped approximately 45 seconds after busy tone is applied.

Check Backup Split [SPLIT #] Priority [PRIORITY LEVEL]

The following variations of this command may be used:

- Check Backup Split [SPLIT #] Priority [PRIORITY LEVEL] if unconditionally.
- Check Backup Split [SPLIT #] Priority [PRIORITY LEVEL] if staffed agents > [# OF AGENTS].
- Check Backup Split [SPLIT #] Priority [PRIORITY LEVEL] if available agents > [# AVAIL].
- Check Backup Split [SPLIT #] Priority [PRIORITY LEVEL] if calls queued < [NQC MIN]
- Check Backup Split [SPLIT #] Priority [PRIORITY LEVEL] if oldest call waiting < [# OF SECONDS]

In the above listed command variations, [Split #] is a valid split number (1-99). [PRIORITY LEVEL] is either "l" (low), "m" (medium), "h" (high), or "t" (top). [# OF AGENTS] is the required number of logged-in agents (0-200). [# AVAIL] is the required number of available (idle) agents (0-200). [NQC MIN] is the minimum Number of Calls that must be in queue before a call is queued to the backup split. [# OF SECONDs] is the minimum number of seconds (2-998) that the oldest queued call will wait before queuing to the backup split.

The **check backup** command checks the status of a split for possible termination of the call to that split. When termination is not possible, queuing at the specified priority is attempted. Termination and/or queuing is attempted if the split meets certain conditions specified as part of the command. The possible conditions are shown below:

- Unconditionally (always attempts to queue)
- Number of available (idle) agents in the specified split
- Number of staffed agents in the specified split
- Number of calls queued at a given priority to the specified split.

  This condition tests for calls in queue at the specified priority level and higher.
- Age of the oldest call waiting in the specified split's queue

A call may be queued in up to three different splits at the same time. A call remains queued until either vector processing terminates (possible reasons include: **disconnect**, **busy**, or **route to** commands, the call drops, or the call is abandoned) or the call reaches an agent. When an agent becomes available in any split to which the call is queued the following actions are performed:

- The call begins alerting the agent
- The call is removed from any other queues
- Vector processing terminates

If the desired backup split is one of the splits to which the call is already queued, the call will be requeued at the new priority level provided the command conditions are met. This step is skipped and vector processing continues at the next step if any of the following are true:

- The command conditions are not met
- The desired split's queue is full
- The desired split is unstaffed
- All staffed agents are in the auxiliary work mode
- The desired split is not vector controlled
- The call is already queued in this split at the specified priority level
- The call has been previously queued to three different splits
- The desired split has no queue and no agents are available

Collect [NO OF DIGITS] digits after announcement extension [EXT]

In this command, [NO. OF DIGITS] is an administered number of digits from 1 through 16, and [EXT] is an administered announcement extension or "none" (digits are collected without a prompt).

The **collect** command lets the caller enter up to 16 digits from a touch-tone phone. An optional announcement may be played first. This step is part of the Call Prompting feature and is described in greater detail in the Call Prompting feature description elsewhere in this manual.

Disconnect after announcement extension [EXT]

In this command, [EXT] is an administered announcement extension or "none."

The **disconnect** command ends treatment of a call and removes the call from the switch. This command also allows the optional assignment of an announcement that will play immediately before the disconnect. If the switch has not yet sent answer supervision, it does so immediately before disconnecting a call so there will be a charge for calls that terminate with the disconnect command. The exception is on ISDN calls where disconnect can occur without returning answer supervision when an announcement is not played.

During playing of the announcement, if the call is moved from waiting in a split's queue to alerting at an agent's extension, the announcement is disconnected, the caller hears ringback, and the **disconnect** command is aborted.

Goto Step [STEP #]

The following variations of this command may be used:

- Goto Step [STEP #] unconditionally
- Goto Step [STEP #] if available agents in split [SPLIT #] is [# AVAIL]
- Goto Step [STEP #] if staffed agents in split [SPLIT #] is > [# OF AGENTS]
- Goto Step [STEP #] if calls queued in split [SPLIT #] is [> or <] [NQC] pri [PRIORITY LEVEL]
- Goto Step [STEP #] if oldest call waiting in split [SPLIT #] is [> or <] [# OF SECONDS]
- Goto Step [STEP #] if digits equal [DIGITS]
- Goto Step [STEP #] if time of day is [DAY] [HOUR] [MINUTE] to [DAY] [HOUR] [MINUTE]

In the above listed command variations, [STEP #] is an assigned step number from 1 through 15. [DIGITS] is a digit string from 1 through 16 digits in length. [SPLIT #] is a valid split number (1-99). [PRIORITY LEVEL] is either "l" (low), "m" (medium), "h" (high), or "t" (top). [# OF AGENTS] is the required number of logged-in agents (0-200). [# AVAIL] is the required number of available (idle) agents (0-200). [NQC] (Number Queued Calls) is the minimum or maximum

Number of Calls that must be in queue before the command will execute. [# OF SECONDS] is the minimum or maximum number of seconds (2-998) that the oldest queued call will wait before the command will execute. [> or <] indicates the "greater than" or "less than" function. [DAY], [HOUR], and [MINUTE] are self-explanatory.

The **goto step** command is a branching step that provides conditional or unconditional movement to later or earlier steps in the Call Vector.

Conditional branching may be used to compensate for heavy traffic or for night/weekend service. Calls in queue and/or any wait treatment in effect before a **goto** will remain in effect after the branch.

The conditions of branching are as follows:

- Unconditionally (always branches)
- Number of available (idle) agents in a specified split
- Number of staffed agents in a specified split
- Number of calls queued at a given priority to the specified split
- Age of the oldest call waiting in the specified split's queue
- Time of Day, start time, and end time specified
- Collected Digits from a collect step (Call Prompting feature only)

Goto Vector [VECTOR]

The following variations of this command may be used:

- Goto Vector [VECTOR #] unconditionally
- Goto Vector [VECTOR #] if available agents in split [SPLIT #] is [# AVAIL]
- Goto Vector [VECTOR #] if staffed agents in split [SPLIT #] is > [# OF AGENTS]
- Goto Vector [VECTOR #] if calls queued in split [SPLIT #] is [> or <] [NQC] pri [PRIORITY LEVEL]
- Goto Vector [VECTOR #] if oldest call waiting in split [SPLIT #] is [> or <] [# OF SECONDS]
- Goto Vector [VECTOR #] if digits equal [DIGITS]
- Goto Vector [VECTOR #] if time of day is [DAY] [HOUR] [MINUTE] to [DAY] [HOUR] [MINUTE]

In the above listed command variations, [VECTOR #] is an assigned vector number from 1 through 256. [DIGITS] is a digit string from 1 through 16 digits in length. [SPLIT #] is a valid split number (1-99). [PRIORITY LEVEL] is either "l" (low), "m" (medium), "h" (high), or "t" (top). [# OF AGENTS] is the required number of logged-in agents (0-200). [# AVAIL] is the required number of available (idle) agents (0-200). [NQC] is the minimum or maximum Number of Calls that must be in queue before the command will execute. [# OF SECONDS] is the minimum or maximum number of seconds (2-998) that the oldest queued call will wait before the command will execute. [> or <] indicates the "greater than" or "less than" function. [DAY], [HOUR], and [MINUTE] are self-explanatory.

The **goto vector** command is a branching step that provides conditional or unconditional movement to another vector.

The **goto vector** command is used for chaining more than one vector together. Calls in queue and/or any wait treatment in effect before a **goto** will remain in effect after the branch.

The conditions of branching are:

- Unconditionally (always branches)
- Number of available (idle) agents in a specified split
- Number of staffed agents in a specified split
- Number of calls queued at a given priority to the specified split
- Age of the oldest call waiting in the specified split's queue
- Time of Day, start time, and end time specified
- Collected Digits from a collect step (Call Prompting feature only)

Messaging Split [SPLIT #] for extension [EXT]

In this command, [SPLIT #] is the administered AUDIX or Message Center split/hunt group number and [EXT] is the extension (from one through five digits) of the party for whom the message is to be left (such as a split/hunt group, VDN, extension, and so on) or "none" (defaults to VDN).

This command causes the calling party to be connected to the AUDIX or Message Center split so that he or she may leave a message for the specified extension (call answering service). If the calling party is successfully connected to the AUDIX or a Message Center agent, then vector processing terminates and a message may be left for the specified extension. If the extension is administered as **none**, then the VDN that accessed the vector associated with this command is used as the default extension.

Priority can be given to the call by assigning a high priority to the messaging split extension via the COR.

This command functions as follows:

1. If the split number specified in the command is a valid message service split (such as AUDIX or Message Server Adjunct), and either the extension is a valid assigned extension or is administered as **none** (defaults to current VDN), then the system will attempt to terminate the call to the message service split for call answering service:
   - If the call is queued to the message service split or if the call terminates to an available message service agent or AUDIX voice

port, then the calling party is connected to ringback (successful termination), and vector processing terminates.

- If the split queue is full, the AUDIX link is down, all AUDIX voice ports are out of service, or the message service split is DCS-AUDIX and all DCS trunks are busy, then vector processing continues at the next vector command (unsuccessful termination).

2. If call termination was successful and the administered extension (or default VDN) is a message service subscriber, the calling party can leave a message for the specified extension.

   If the extension or VDN was not a subscriber of the message service, one of the following may occur:

   - If the message service split is AUDIX, the calling party will receive ringback until he or she disconnects.
   - If the message service split is a Message Server Adjunct, the calling party may be answered by an message service agent but no message is taken since the specified extension (default VDN) is not a Message Server Adjunct subscriber.

Queue To Main Split [SPLIT #] Priority [PRIORITY LEVEL]

In this command, [SPLIT #] is a valid split number (1-99). [PRIORITY LEVEL] is either "l" (low), "m" (medium), "h" (high), or "t" (top).

The **queue to main split** command sends a call to a split and assigns a queuing priority level to the call if all agents are busy. A call sent with this command either connects to an agent in the split or enters the split's queue.

A call may be queued in up to three different splits at the same time. A call remains queued until either vector processing terminates or the call reaches an agent. When an agent becomes available in any split to which the call is queued, the following actions are performed:

- The call begins alerting the agent
- The call is removed from any other queues
- Vector processing terminates

If the desired split is one of the splits to which the call is already queued, the call is requeued at the new priority level. This step is skipped and vector processing continues at the next step if any of the following are true:

- The desired split's queue is full
- The desired split is unstaffed
- All staffed agents are in the auxiliary work mode
- The desired split is not vector controlled
- The call is already queued in this split at the specified priority level
- The call has already been queued to three different splits

- The desired split has no queue and no agents are available

Route to number [NUMBER]

In this command [NUMBER] is an administered digit string from 1 through 16 digits in length. This digit string represents a destination number which may be any of the following:

- An internal extension (such as a split/hunt group, voice terminal, announcement, and so on)
- A VDN extension
- An attendant
- A remote extension
- An external number such as a TAC or AAR/ARS FAC followed by a public or private network number.

In addition to digits 0 through 9, the * and the # sign, four special characters can be administered in the digit string: ˜m (mark), ˜p (pause), ˜s (suppress), and ˜w (wait). Each special character counts as two digits towards the maximum of 16 digits for this command. The function of these special characters is the same as in the Abbreviated Dialing feature.

This command allows a call to be routed to a specified destination. The COR associated with the VDN is used to determine any calling party restrictions (such as lack of calling permission, FRL restriction, and so on). If this command is successful (that is, it successfully terminates to a station, seizes a trunk, and so on) then vector processing terminates. Otherwise, vector processing continues at the next vector command.

Call Coverage does not apply when this command is used to route a call to a system extension with Call Coverage assigned. In other words, call coverage is disabled with respect to the **route to number** command.

This command functions as follows:

1. If the number is a system extension or attendant group (not a VDN) the system will attempt to terminate the call to the endpoint if one of the following conditions occur:
   - The endpoint is alerted.
   - The endpoint has call forwarding or night service (hunt group) enabled, and the forwarded to (night service) destination is alerted (or in the case of off-premises call forwarding, a trunk is seized).

   The system then provides ringback to the calling party, and vector processing terminates. However, if the call cannot complete successfully (for example, no idle appearance is available), vector processing continues at the next vector command.
2. If the number is a VDN extension, then the following events occur:

- Vector processing terminates within the current vector.
- If the current VDN is administered with display override, then the new VDN overrides the current VDN as the VDN to be displayed at the terminating extension.
- Processing of the vector associated with the VDN extension begins.

3. If the number is an AAR/ARS FAC plus digits, or a remote (UDP) extension, standard AAR/ARS processing is performed to select the trunk group and outpulse the digits. If a trunk is seized, vector processing terminates, and the calling party hears feedback provided by the far end. Otherwise, the call cannot complete successfully (no trunks available, FRL/COR restricted, and so on), and vector processing continues at the next vector command.
4. If the number is a TAC plus digits, and a trunk is seized, then vector processing terminates and the calling party hears feedback provided by the far end. Otherwise, the call cannot complete successfully (no trunks available, COR restricted, and so on), and vector processing continues at the next vector command.
5. If the number is any other number (such as an FAC plus digits), vector processing continues at the next vector command (unsuccessful route to command).

A variation of the **route to number** command is the **route to number [NUMBER] if digit equals [DIGIT]** command which is used for call prompting. This command is discussed in detail in the Call Prompting feature description, elsewhere in this manual.

Route to digits with coverage [y/n]

The **route to digits** command is a Call Prompting command. It attempts to route the call like the basic **route to** command, except that the number routed to will be a set of digits collected from the caller (via a collect step). For more details see the Call Prompting feature description elsewhere in this chapter.

Stop

The **stop** command terminates the processing of any subsequent vector steps. After the **stop** command, any calls that are already queued remain queued, and any wait treatment such as silence, ringback, or music is continued.

Wait time [SECONDS] secs hearing [TREATMENT]

In this command, [SECONDS] is the number of seconds (0-998) to wait before processing the next vector step. The number of seconds must be an even number. [TREATMENT] indicates the treatment (silence, ringing, or music) to be given to the caller during the wait period.

The **wait** command is generally used to set a length of time for a call to wait in the main split's queue and to specify either silence, ringing, or music while the call advances in queue. A wait time of 0 seconds can be used to change the call treatment without executing a delay.

Success/Failure Criteria for Vector Commands

Table 2-26 shows the success and failure criteria for the various vector commands. Note that the **collect**, **messaging**, and **route to digits** commands are not included in this table, but are described in detail in the Call Prompting feature described elsewhere in this chapter. Also, the **adjunct routing** command is not included in this table, but is described in detail in the Inbound Call Management feature elsewhere in this chapter. It is very important that the information in this table be understood before assigning vectors.

Table 2-26. Criteria for Success/Failure of Vector Commands

Command	Succeed/Fail Criteria	Vector Processing Disposition
Announcement	Fails if specified announcement is unadministered or unavailable.	Continues vector processing with next sequential step.
	Otherwise, succeeds.	Pass control to announcement software.
Busy	Always succeeds (unanswered CO trunks will not hear busy tone).	Exit vector processing, and pass control to call processing.
Check Backup Split	Fails if step conditions are not met, split's queue is full, split is unstaffed, all staffed agents are in auxiliary work mode, split is not vector controlled, call is already queued at specified priority to specified split, or call is already queued to three different splits.	Continue vector processing with next sequential step.
	Succeeds, and call is terminated to agent.	Exit vector processing, and pass control to call processing.
	Succeeds, and call is queued or requeued at specified split.	Continue vector processing with next sequential step.
Disconnect	Always succeeds.	Conditionally under control of announcement software. Then, exit vector processing, and pass control to call processing.
Goto Step	Fails if step condition is not met.	Continues vector processing with next sequential step.
	Succeeds if step condition is met.	Continues vector processing with the destination step.

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Table 3-26. Criteria for Success/Failure of Vector Commands *(Continued)*

Command	Succeed/Fail Criteria	Vector Processing Disposition
Goto Vector	Fails if step condition is not met.	Continues vector processing with next sequential step.
	Succeeds if step condition is met.	Continues vector processing with step one of destination vector.
Queue to main split	Fails if split's queue is full, split is unstaffed, all staffed agents are in auxiliary work mode, split is not vector controlled, call is already queued at specified priority to specified split, or call is already queued to three different splits.	Continue vector processing with next sequential step.
	Succeeds, and call is terminated to agent.	Exit vector processing, and pass control to call processing.
	Succeeds, and call is queued or requeued at specified split.	Continue vector processing with next sequential step.
Route To*		
ATTENDANT	Always succeeds.	Exit vector processing. Pass control to processing for attendant queue.
EXTENSION	Fails if extension has no call appearance available for termination.	Continue vector processing with next sequential step.
	Otherwise, succeeds.	Exit vector processing. Pass control to call processing.
REMOTE EXTENSION	Fails due to a resource failure (for example, no trunks), or COR/FRL restricted.	Continue vector processing with next sequential step.
	Otherwise, succeeds.	Exit vector processing. Pass control to call processing.
VDN	Always Succeeds.	Exit previous vector processing. Pass control to new vector.
OTHER	Fails due to unknown destination or feature activation.	

		Continue vector processing with next sequential step.
Stop	Always succeeds.	Exit vector processing. Control is passed to normal call processing. Any queuing or treatment in affect remains in affect.
Wait	Always succeeds.	Connect specified treatment and pass control to delay timer.

* The **route to** command may have additional success criteria when Look-ahead Interflow and/or Call Prompting is optioned.

ACD Split/Hunt Group Operation with Call Vectoring

A split is considered Vector Controlled if `yes` has been selected in the `Vector` field on the Hunt Group Administration screen. Only vector controlled splits are accessible from the following call vectoring commands: **check backup** and **queue to main split**.

Overview

Throughout this description, **split** is used to specify a hunt group or ACD split. **ACD split** is used when the group has ACD optioned. **Hunt group** is used when the group does not have ACD optioned.

This section describes the interactions between the Hunting, ACD, and Call Vectoring features. Some key points to note are:

1. Hunt groups and ACD splits may coexist with one another as well as with vector-controlled splits.
2. Vector-controlled splits may be called directly. However, these calls will not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.
3. ACD is required to obtain any CMS and/or BCMS split measurements.

Vector Controlled Splits

Vector Controlled Splits have the following properties:

- Agents may be logged to as many as three vector controlled splits simultaneously.
- Agents' voice terminals are not locked while they are logged into a vector controlled split.
- A vector controlled split may also be adjunct controlled. If it is, the more restrictive properties of each controlling function apply.
- A vector controlled split is not mapped to an adjunct via system

administration, unless it is also adjunct controlled.

- Request Notification is allowed (as long as the split is not adjunct-controlled). The adjunct receiving notification is the same one requesting it.
- When a vector controlled split is removed via system administration, any active notification request is terminated.
- A vector controlled split cannot be assigned a coverage path.
- Call Forwarding cannot be activated for a vector controlled split.
- No announcements are associated with vector controlled splits

Split Queue Priority Levels

Each call in a split queue can have one of four possible priority levels: Top, High, Medium, and Low. Calls are processed initially by priority level. Within each priority level, calls are processed in first-in, first-out (FIFO) order. A vector can be administered to process queued calls using any of the four priority levels. Calls not controlled by a vector can use only two priority levels: High and Medium.

The default priority level for non-vector controlled calls queued to a split is Medium. An ACD split may assign certain calls to the high-priority level. The high-priority level can be assigned to a call by intraflowing from one ACD split to another ACD split. This is achieved using the `Call Coverage option` field and setting the priority on intraflow field on the Hunt group form of the principal ACD split. The high-priority level can also be assigned to a call from a voice terminal user whose associated COR has priority queuing optioned.

Direct Agent Calls (DACs) have the highest priority of any calls (including those mentioned in the previous paragraph).

Split Thresholds

The split thresholds, available-agents, and staffed-agents, used by the **check** and **goto** vector commands, have slightly different meanings for hunt groups and ACD splits. For ACD splits, staffed-agents is the number of agents logged-in. Available-agents is the number of agents logged-in and ready to receive an ACD call (for example, Auto-in or Manual-in work mode and not currently on a call). Since hunt groups have no log-in, log-out, or work modes, a staffed-agent is merely an administered agent. All that is checked is the number of administered agents. Similarly, available-agents is the number of agents ready to receive a hunt group call.

When there is no queue, the oldest call waiting is always 0 and the number of queued calls is always 0. When the system time is not set, the time of day and oldest call waiting threshold will always fail.

No matter what the threshold may be, a call is never allowed to queue to an ACD split if all staffed agents are in Auxiliary Work mode or there are no staffed agents.

Outbound Call Management (OCM). An ACD split may be administered with a `bx.25` in the `Controlling Adjunct field`. This means that the ACD split is controlled by the OCM adjunct. Only calls from the OCM adjunct will be accepted by this ACD split. If both Vector and Controlling Adjunct are optioned for a particular ACD split, OCM calls may be directed to this ACD split via a vector. If a non-OCM vector call attempts to queue a call to this split, the associated vector step will fail and be skipped.

Call Vectoring and the Call Management System (CMS)

The CMS provides the following capabilities to be used with Call Vectoring:

- The CMS can request full translations to initialize its database. The PBX supplies data for VDNs and measured trunk groups which map to VDNs.
- The CMS can retrieve and change vectors on the PBX. However, no information is sent to the CMS when vectors are administered via the Manager I terminal (G1) or the G3 Management Terminal.
- The CMS can change the vector number of VDNs.
- The PBX notifies the CMS when the incoming destination of a trunk group is changed. This happens when a trunk group maps to either a split or VDN.
- The PBX notifies the CMS when a VDN is called. State change messages are also sent as a vector is processed.
- The PBX notifies the CMS when an agent transfers a trunk to a VDN.

DNIS

Call Vectoring provides DNIS, which allows agents with display-equipped voice terminals to receive visual displays that specify the name of the called VDN.

In traditional ACD arrangements, groups of agents are organized into splits. With this type of arrangement, an agent is trained to answer calls for one specific purpose in an efficient and professional manner. However, ACD managers are recognizing the need to relax this concept of limiting each split to a single call-answering task.

The alternative is to provide splits where each group of agents is proficient with several types of calls. The desired gain is to provide adequate service for the several call types with fewer agents and with less administrative intervention by the ACD manager. Using this approach, the changing staff needs of the several call types are averaged in time, and enough agents are staffed to provide adequate service for the prevailing average load. Where five agents might be needed in each of three smaller splits (15-agent total) to handle three types of calls, only 11 or 12 agents might be needed in the single (more general) split.

The DNIS function provided by Call Vectoring allows each answering agent to know the purpose of each incoming call as the call terminates to the agent's voice terminal. As a result, the natural efficiencies of the single-split/single-call type arrangement are not compromised. With the display information provided by DNIS (for example, the name of the called VDN), agents are aware of each call's purpose, and can answer each incoming call with the appropriate greeting. Agents do not have to waste time trying to determine the purpose of calls.

Call Vectoring Examples

A. Limiting an ACD Queue:

1. go to Step 6 if calls queued in split 15 > 5
2. queue to main split priority medium
3. go to Step 6 if calls queued in split 15 < 1
4. wait two sec hearing ringing
5. stop
6. busy

   or

   disconnect after announcement 2315 ("We're sorry all lines are busy. Please call back later.")

B. Providing an ACD Split to Handle Emergency Calls:

1. queue to maint split 9 at low priority (split 9 has from one to four agents)
2. go to Step 7 if calls queued in split 9 < 1
3. go to Step 8 if calls queued in split 9 > 10
4. wait time 36 sec hearing ringing
5. go to Step 8 if unconditionally
6. stop
7. announcement 2921 ("We are aware of the widespread situation. Efforts are being taken to rectify the problem. If your call is not urgent please call back later. If it is urgent, please wait.")
8. wait two sec hearing music (or silence)

C. Providing a Specific Emergency Announcement:

1. queue to main split 9 priority low
2. go to Step 6 if calls queued split 9 < 1
3. announcement 2982 ("We are aware of the power outage in Plainfield. If you still need help, please wait.")
4. wait time two sec hearing music

5. stop
6. busy

D. Providing a Forced Announcement to Handle Emergency Calls:

1. queue to main split 9 priority low
2. go to Step six if calls queued < 1
3. announcement 2983 ("We are aware of the current situation and are working to rectify the problem. If your call is not urgent, call back later.")
4. wait time six sec hearing music
5. stop
6. busy

E. Providing Forced Announcement for Attendant Queue:

1. announcement 2983 (same as example D)
2. wait time eight sec hearing music
3. route to 0

F. Basing Delay Intervals on Number of Calls in an ACD Queue:

1. queue to main split 6 priority low
2. go to Step 11 if calls queued < 1
3. go to Step 6 if calls queued > 16
4. go to Step 7 if calls queued > 8
5. go to Step 8
6. wait time six sec hearing ringback
7. wait time six sec hearing ringback
8. wait time 16 sec hearing ringback
9. announcement 2952 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
10. stop
11. busy

G. Providing Conditional Night Service for Attendant Queue or ACD Queue:

1. go to Step 4 if time of day is mon 09:00 to fri 17:00
2. route to 3300 (VDN to access a common vector for the night announcement)
3. stop
4. Steps 4 through whatever represent vector processing during working hours

H. Providing the Night Service Announcement:

   1. disconnect after announcement 2692 ("Our business is now closed. Please call back between the hours of 9 a.m. and 5 p.m. Monday through Friday.")

I. Using AUDIX to Provide Night Service for Message Center:

   1. go to Step 7 if time of day is all 16:30 to all 7:30
   2. queue to main split 47 priority high (split 47 is a group of daytime agents)
   3. go to Step 6 if calls queued $<$ 1
   4. wait time two sec hearing ringback
   5. stop
   6. busy
   7. wait time two sec hearing ringing
   8. messaging split 18 for extension 2000 (split 18 is the Audix split; extension 2000 is the extension split 47; this command will allow a message to be left for split 47.
   9. stop

J. Providing an Information Announcement for Callers:

   1. disconnect after announcement extension 2918 ("Today has been declared a snow day. Please report for work tomorrow at 8 a.m.")

K. Providing a Repeated Delay Announcement:

   1. queue to main split 53 priority top
   2. go to Step 7 if calls queued $<$ 1
   3. wait time 30 sec with ringback
   4. announcement extension 2556 ("Our agents are busy. Please hold.")
   5. wait time 20 sec with music
   6. go to Step 4 if unconditionally
   7. busy

L. Providing Conditional Intraflow for an ACD Split:

   1. queue to main split 21 priority medium
   2. go to Step 5 if calls queued in split 21 $<$ 1
   3. go to Step 5 if oldest call waiting in split 21 $>$ 45 sec
   4. stop
   5. check backup split 22 priority high if calls queued $<$ 5
   6. go to Step 7 if calls queued in split 22 $<$ 1

7. busy

M. Providing Unconditional Intraflow for an ACD Split:

   1. go to Step 6 if staffed agents in queue 21 < 1
   2. queue to main split 21 priority medium
   3. go to Step 6 if calls queued in split 21 < 1
   4. wait time 30 sec hearing ringing
   5. stop
   6. route to 3500 (VDN of another local ACD split)

N. Scanning Multiple Backup Splits:

   1. queue to main split 37 priority low
   2. go to Step 6 if calls queued in split 37 < 1
   3. go to Step 6 if oldest call waiting in split 37 > 45 sec
   4. wait time 30 sec hearing ringing
   5. stop
   6. check backup split 11, priority low if available agents > ??
   7. check backup split 12, priority low if available agents > ??

   Provided that there are agents logged into split 37 and not in aux work, this vector could result in a call being queued to three splits (the maximum simultaneous queues in which a call can reside) and hearing ringing until answered.

O. Combining the Conditions of Check Backup Split Commands:

   1. queue to main split 56 priority medium
   2. go to Step 7 if calls queued in split 56 < 1
   3. go to Step 7 if staffed agents in split 56 < 2
   4. go to Step 7 if oldest call waiting in split 56 > 34
   5. wait time 10 sec hearing ringing
   6. stop
   7. check backup split 57 priority high if unconditionally
   8. go to Step 11 if calls queued in split 57 < 1
   9. wait time 10 sec hearing ringing
   10. stop
   11. busy

P. Gracefully Closing an ACD Split:

   1. go to Step 3 if time of day is monday 8:00 to saturday 16:00

2. disconnect after announcement extension 2344 ("Please call back during business hours: 8:00 to 4:00, Monday through Saturday.")
3. queue to main split 52 priority low
4. go to Step 9 if calls queued in split 52 < 1
5. go to Step 10 if time of day is all 15:56 to all 16:01
6. wait time 20 secs hearing ringing
7. announcement extension 2340 ("Our agents are busy. Please hold. Calls are being answered in their order of arrival.")
8. stop
9. busy
10. announcement extension 2341 ("It is nearly 4:00, closing time for this office. We are trying to serve your call. If we can't, please call back between 8:00 and 4:00, Monday through Saturday.")
11. wait time 6 sec hearing music
12. go to Step 14 if time of day is all 16:00 to all 16:15
13. go to Step 11 if unconditionally
14. disconnect after announcement extension 2343 ("We are sorry. This office has closed. To be assured of service please call back between 8:00 and 3:45, Monday through Saturday.")

Q. Example of a Chained Vector for ACD:

1. go to Step 3 if time of day is monday 7:30 to friday 4:30
2. go to vector 55 (pointing to the chained vector)
3. queue to main split 59 priority low
4. go to Step 11 if calls queued in split 59 < 1
5. go to Step 11 if calls queued in split 59 < 3
6. go to Step 11 if oldest call in split 59 > 36 sec
7. wait time 16 sec hearing ringing
8. announcement extension 2355 ("Our agents are busy. Please wait. Calls are being answered in their order of arrival.")
9. wait time 20 sec hearing music
10. go to Step 8 if unconditionally
11. check backup split 1 priority low if calls queued < 10
12. go to Step ?? if calls queued in split 1 < 1
13. go to Step ?? if calls queued in split 1 > 9
14. go to Step 7 if unconditionally
15. disconnect after announcement ext. 2356 ("Every line is busy.

Please call back later.")

R. Vector 55:

1. disconnect after announcement ext 2357 ("Please call back during business hours: 7:30 to 4:30 Monday through Friday.")

Considerations

Call Vectoring is a valuable feature that provides a highly flexible way of processing incoming calls to the switch. With Call Vectoring, one can define a separate set of call processing steps for different types of incoming calls. Maximum benefits are realized when Call Vectoring is combined with Call Prompting and ACD.

The system allows a maximum of 256 vectors, with a maximum of 15 steps (commands) per vector.

A call may be queued in a maximum of three splits simultaneously.

A maximum of 16 characters is allowed in a **route to** command. Each stored digit counts as one such character, and each Special Character counts as two such characters.

The maximum value for the seconds parameter on the **wait** step is 998.

The maximum value for the longest call waiting time threshold in the **check backup split** or **goto** steps is 998 seconds.

The maximum value for the number of calls waiting threshold in the **check backup split** or **goto** steps is 200 calls.

The maximum value for the number of staffed agents threshold in the **check backup split** or **goto** steps is 200 agents.

The maximum value for the number of available agents threshold in the **check backup split** is 200 agents.

A maximum of 128 analog and integrated announcements is allowed.

A maximum of five calls can simultaneously be connected to one integrated announcement port.

A maximum of 200 agents per split is allowed.

A maximum of 500 agents per switch is allowed.

A maximum of 1,000 vector steps will be processed per call.

A call can be queued to multiple splits and/or hunt groups at the same time to minimize the waiting time. The maximum number of queues that a call can be queued at is three.

It is recommended that you not change a vector while it is processing calls because the calls already in the vector might experience problems. Instead, add a new vector and change the VDN to go to the new vector.

Agents should not be used for hunt group calls and ACD split calls simultaneously. Otherwise, all of the calls from one split (either ACD or hunt group) will be answered first. For example, if the ACD calls are answered first, none of the hunt group calls will be answered until all of the ACD calls are answered.

The oldest call waiting termination is only supported for agents who are servicing ACD calls only.

Call Vectoring operations in DEFINITY Generic 3i systems are different from those in DEFINITY Generic 2 systems. For DEFINITY Generic 2 Call Vectoring information, see *DEFINITY® Communications System Generic 2 and System 85 — Feature Descriptions, 555-104-301*.

Interactions

The following features interact with the Call Vectoring feature:

- Abbreviated Dialing

  A VDN extension can be used in a personal, group, system, or enhanced number list.

- Administered Connections

  The PSC administration screen blocks entry of a VDN.

- Answer Supervision

  Answer Supervision is only returned once during the life of a call. With respect to Call Vectoring commands, answer supervision is returned in response to an announcement, waiting with music, and disconnect. In addition, if a call is answered, answer supervision is sent if it hasn't previously been sent.

- AP Demand Print

  A VDN cannot be used as an argument to the feature access code for AP Demand Print.

- Attendant Control of Trunk Group Access

  If a **route to** step in a vector dials a controlled trunk group, vector processing continues at the next step.

- Attendant DXS with Busy Lamp Field

  A DXS lamp for a VDN is always off. A DXS button can be used to place a call to a VDN.

- Attendant Display

A button may be assigned to the attendant to display an internal caller's COR. The restriction identifiers are: orig, otwd, toll, code, and none. This operation is available for calls originated by a **route to** step in a vector. The COR of the originator is displayed.

- Attendant Recall

  Attendant Recall to a VDN will be blocked.

- AUDIX Interface

  A **route to** step in a vector may call the AUDIX extension. If a voice port can be seized to that adjunct, vector processing is terminated. The system sends a message to AUDIX requesting retrieval of messages for the originating extension (not the VDN).

  AUDIX may also be accessed by the **queue to main split** and **check backup** commands. Also, the messaging step may use an AUDIX hunt group in its operation.

- Authorization Codes

  Authorization codes are disabled with respect to routing via VDNs. In other words, if authorization codes are enabled, a **route to** command in a prompting vector accesses AAR or ARS and the VDN's FRL does not have the permission to utilize the chosen routing preference, then no authorization code is prompted for and the **route to** command will fail.

- AAR/ARS

  Any **route to** command in a vector can dial an AAR/ARS FAC followed by other digits.

- Automatic Callback

  Automatic Callback cannot be used for calls placed to a VDN.

- Automatic Call Distribution

  Splits which are not translated as vector-controlled may use all of the existing ACD features. Splits which are translated as vector-controlled lose certain ACD properties.

- Automatic Incoming Call Display

  The information displayed for the current call is replaced by the identity of the incoming call. A **route to number** step in a vector can initiate a call to an extension. The name of the originator (or the name of the trunk group) and the name of the VDN are displayed for that extension.

- Automatic Message Waiting Notification

  The Message button can specify a VDN. The associated lamp will be lit when messages are left for that VDN.

- Automatic Wakeup

  A wakeup call cannot be programmed for a VDN.

- Bridged Call Appearance

  VDN extensions cannot be assigned to bridged appearance buttons. A **route to** command to an extension with bridged appearances will update bridged appearance button lamps.

- Busy Verification of Terminals and Trunks

  Busy verification of VDNs is denied and intercept tone is returned.

- Call Coverage

  A vector or VDN cannot be translated as a coverage point. The coverage answer group administration screen blocks entry of a vector or VDN.

  Vectors and VDNs do not have coverage paths associated with them.

- Call Forwarding

  Calls can be forwarded to a VDN. Calls placed by a **route to** command to an extension that has call forwarding activated will be forwarded.

  An attendant or voice terminal with console permission can activate/deactivate call forwarding for a particular extension number, TEG, or hunt group. However, activation/deactivation of call forwarding for a VDN is blocked.

  If call vectoring is enabled, activation/deactivation of call forwarding for a vector-controlled hunt group is blocked. The user receives intercept tone.

- Call Park

  Calls cannot be parked on a VDN. If the call park access code is dialed and a VDN is entered, the user receives intercept tone. If the call park answer back access code is dialed and a VDN is entered, the user receives intercept tone.

- CMS

  See the section in this feature description entitled "Call Vectoring and the CMS" for CMS interactions.

- Call Pickup

  A VDN cannot be administered as a member of a pickup group. However, a vector call that **routes** to an extension can be picked up, if that extension is part of a pickup group.

- Call Prompting

  Call Prompting is administered through Call Vectoring administration. If only Call Vectoring is enabled, vectors can be administered using only Call Vectoring commands. If only Call Prompting is enabled, vectors can be administered using only Call Prompting commands, described elsewhere in this chapter. When both features are enabled, all the commands associated with vectoring and prompting are available.

  Enabling both vectoring and prompting together provides the capability to prompt the caller to enter pertinent data while the caller is waiting in queue at an ACD split. For example, prompting and vectoring can be used together to enhance the message collection capability to provide a caller who is waiting in queue at an ACD split with the choice of remaining in queue or leaving a message for that split. In addition to queuing, vectoring provides the capability to change calling party feedback. In addition to silence, which both prompting and vectoring supply, vectoring can provide ringback or music feedback to the calling party.

The implications of having both features enabled are:

- When a Call Prompting command (such as a successful **route to** command) terminates vector processing, the call must be dequeued and dropped from all queues that it is currently residing in as part of the termination process.
- If a call is waiting in an announcement queue, waiting to be connected to an announcement or an announcement queue (announcement retry), or is currently connected to an announcement, and the call is dequeued from a split's queue and terminates to an agent's voice terminal extension, the announcement is disconnected and ringback is connected to the call.
- If a **collect digits** command is being processed for a call and the call is dequeued from a split's queue and terminates to an agent's voice terminal extension, the **collect digits** command is terminated and ringback is connected to the call.

■ Call Waiting Termination

A **route to number** command in a vector can dial a single-line voice terminal. If the extension is busy and has call waiting termination administered, the **route to number** operation is considered unsuccessful and vector processing continues at the next step.

■ CAS

A **route to number** command in a vector can dial CAS. If a release link trunk can be seized, the **route to number** operation is considered successful and vector processing is terminated.

■ COR

Each VDN in the system has a COR associated with it. This VDN COR is used to determine the calling permissions/restrictions, the AAR/ARS PGN, and the priority queuing associated with a given vector.

■ Code Calling Access

A VDN cannot be used as the argument to the code calling access feature access code.

If a **route to number** command in a vector specifies the code calling feature access code, vector processing continues at the next step.

■ Conference

A call to a VDN can be included as a party in a conference call only after vector processing terminates for that call (for example, after a successful **route to** command).

- Coverage Callback

  A vector call does not follow any coverage paths; therefore, coverage callback will not be available.

- DCS Attendant Control of Trunk Group Access

  This allows an attendant at any node in the DCS to exercise control over an outgoing trunk group at a different node in the cluster. If a **route to number** step in a vector dials a controlled trunk group, vector processing continues at the next step.

- DCS Attendant Display

  Calls to/from a system in a DCS environment have calling/called party identification transparency. The name of the originator is displayed on the called terminal's alphanumeric display. The name of the party to whom the call is directed is displayed on the originator's display.

  The same display operations occur even if the originator, VDNs, and terminator are on different nodes in a DCS network.

- DCS Automatic Callback

  Automatic Callback calls do not apply for VDNs.

- DCS Call Forwarding

  Calls can be forwarded to a VDN anywhere in the DCS network. An attendant cannot activate/deactivate call forwarding for a VDN.

- DCS LWC

  LWC messages cannot be generated for a VDN.

- DCS Voice Terminal Display

  This feature allows calling and called name information (plus miscellaneous identifiers) to be sent from a terminal on one node to a terminal on another node. The name of the originator is displayed on the called terminal's alphanumeric display. The name of the party to whom the call is directed is displayed on the originator's display.

  The same display operations occur even if the originator, VDNs, and terminator are on different nodes in a DCS network.

- DID

  DID trunks can dial a VDN and be subject to the treatment of its associated vector.

- DOD

  DOD can be provided via a **route to** command within a vector. The COR of the VDN is used to determine calling party permissions/restrictions.

- Do Not Disturb

Do Not Disturb cannot be activated for a VDN.

- Emergency Access to the Attendant

  When night service is in effect, emergency calls to the attendant route to the night destination. The night destination can be a VDN.

  The extension number where emergency queue overflow will redirect can be a VDN.

- Facility Busy Indication

  The facility busy lamp indication for a VDN is always off. A facility busy button may be used to call a VDN.

- FRL

  If a **route to** command dials an external number via AAR/ARS, the FRL associated with the VDN COR is used to determine the accessibility of a routing preference in an AAR/ARS pattern.

- Facility Test Calls

  If a **route to number** command in a vector specifies a Facility Test Call, vector processing continues at the next step.

- Forced Entry of Account Codes

  If a COR requiring entry of account codes is assigned to a VDN, the **route to number** commands executed by the associated vector will be unsuccessful and vector processing continues at the next step.

- Individual Attendant Access

  Each attendant console can be assigned an individual extension number. That extension number can be used as the argument to a **route to number** command in a vector.

  Each attendant has a queue that allows two incoming calls to wait. This individual attendant queue has priority over all other attendant-seeking calls. A call established by a **route to number** command in a vector can wait in this queue and is removed from vector processing.

- ISDN-PRI

  A VDN may be accessed by an ISDN-PRI trunk. ISDN-PRI calls may be disconnected without providing answer supervision.

  A vector may initiate calls over ISDN-PRI facilities via a **route to number** command.

- Integrated Directory

  VDN names and extensions are not available in the Integrated Directory feature.

- Intercept Treatment

  A VDN cannot be used for Intercept Treatment.

- Intercom — Automatic

A VDN cannot be included in an intercom group.

- Intercom — Dial

  A VDN cannot be included in an intercom group.

- Inter-PBX Attendant Calls

  A **route to number** command in a vector can dial the Inter-PBX Attendant. If the call attempts to access a controlled trunk group, vector processing continues at the next step.

- Intraflow and Interflow

  The functionality of intraflow and interflow may be obtained using the **check backup** and **goto** Call Vectoring commands.

  Calls may intraflow from an ACD split which is not vector-controlled into one that is vector-controlled.

- Last Number Dialed

  After a voice terminal user with a Last Number Dialed button dials a VDN extension, the VDN extension is stored as the Last Number Dialed and may be redialed by pressing the button.

- LWC

  LWC messages cannot be stored, canceled, or retrieved for a VDN.

- Manual Message Waiting

  Message Waiting buttons may point to a VDN.

- Manual Signaling

  A manual signaling button can point to a VDN. However, activation of a Manual Signaling button which points to a VDN is ignored.

- Multiple Listed Directory Numbers

  The incoming destination for each CO trunk group and each FX trunk group used for LDNs can be a VDN. The DID LDN night extension can be a VDN.

- Music-on-Hold Access

  The system-wide Music-on-Hold Access feature must be administered in order for music to be heard when a **wait** command with music is executed. If it is not administered, the calling party receives silence.

- Network Access — Private

  A **route to number** command in a vector can access private networks.

- Network Access — Public

  A **route to number** command in a vector can access public networks.

- Night Service

  A VDN can be administered as a night service destination.

Route to commands that route to destinations with night service activated redirect to the night service destinations.

- Priority Calling

  A VDN cannot be used with the priority calling access code. Intercept tone is supplied to the user. If a **route to number** in a vector specifies the priority calling access code, vector processing continues at the next step.

- Property Management System Interface

  VDNs cannot be used with the following features and functions: Message Waiting Notification, Check-In, Check-Out, Room Status, and Automatic Wakeup.

- Recorded Announcement

  The first announcement extension, second announcement extension, first announcement delay, second announcement delay, and recurring second announcement do not exist for a hunt group translated as vector-controlled. Recorded announcements may be accessed via the **announcement**, **collect**, **disconnect**, and **route to number** vector commands.

- Remote Access

  A remote access user can access a VDN.

  If a **route to number** command in a vector specifies the remote access extension, vector processing continues at the next step.

- Ringback Queuing

  External call attempts made via **route to** commands (except for **route to digits with coverage y**) will not queue via Ringback Queuing when all trunks are busy.

- Send All Calls

  If the destination of a **route to number** command has the Send All Calls feature active, the feature activation will be ignored. If there is an idle appearance, the call will terminate there and vector processing will terminate. If not, vector processing continues at the next step.

  If the Send All Calls button is pressed after a vector call is terminated, button activation will be denied.

- Service Observing

  Service observing cannot be used toward a VDN.

- CDR Account Code Dialing

  If a **route to number** command in a vector specifies an CDR account code, vector processing will continue at the next step.

- CDR

  The Feature Related System Parameters form can be administered so that the VDN extension will be used in place of the Hunt Group or Agent extension. If administered to do so, this overrides the "Call to Hunt Group - Record" option of CDR for Call Vectoring calls.

If a vector interacts with an extension or group that has Call Forwarding All Calls active, normal Call Forwarding/CDR interactions apply.

For incoming calls to a VDN, the duration of the call is recorded from the time answer supervision is returned.

— If answer supervision is returned by the vector (via an **announcement**, **collect**, **disconnect**, or **wait with music** command), and the call never goes to another extension, then the VDN extension is recorded as the called number in the CDR record.

— If the call terminates to a hunt group, then the VDN, hunt group, or agent extension is recorded as the called number as per the administration described above.

— If the call terminates to a trunk, then the following two CDR records will be generated:

1. An incoming record with the VDN as the called number and the duration from the time answer supervision was provided to the incoming trunk.

2. An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.

Outgoing vector calls generate ordinary outgoing CDR records with the originating extension as the calling number.

No Ineffective Call Attempt records will be generated for Call Vectoring **route to** commands that are unsuccessful.

- Subnet Trunking

  Subnet trunking will apply to AAR/ARS calls placed via a **route to number** command.

- System Measurements

  Hunt group measurements will need to be interpreted differently when Call Vectoring is used. For example, If a call queues to splits one, two, and three, but the caller abandons the call, the measurements will show only split 1 as having an abandoned call.

- TEG

  A VDN cannot be administered as a TEG member.

  A **route to number** command in a vector can specify a TEG extension.

- Timed Reminder

  The attendant Timed Reminder is not available for calls placed, transferred, or extended to a VDN.

- Time of Day Routing

Since a **route to number** command in a vector can specify the AAR or ARS access codes, the TOD routing algorithm can be used to route the call.

- Transfer

  Stable trunk or internal calls (such as those that are currently in a talking state) can be transferred to a VDN. Calls in which vector processing is active are treated the same as ringing calls for transfer purposes.

- TCM

  A TCM is sent when a **route to** command dials a seven-digit ETN or 10-digit DDD number via AAR/ARS. This TCM is the FRL associated with the VDN COR.

- Trunk Identification by Attendant

  This feature is available for incoming or outgoing calls routed through a VDN.

- UDP

  A **route to** command can call a UDP extension.

- Voice Message Retrieval

  If a **route to number** command in a vector specifies the Voice Message Retrieval access code, vector processing continues at the next step.

Administration

Call Vectoring is administered on a per-system basis by the System Manager. The following items require administration:

- Call Vector Form (one form per vector)
  - Vector Number
  - Vector Name (optional)
  - Vector Commands (up to 15)
- Vector Directory Number Form (one form per VDN)
  - VDN extension
  - Name used to identify the VDN
  - Whether or not the name of the VDN can be overridden (for display purposes) by the name of subsequent VDNs through which the call may be routed.
  - COR
  - Vector number accessed through the VDN.
- Hunt Group Form
  - Whether or not each hunt group is vector-controlled.

- System-Parameters Customer-Options Form
  - Basic vectoring, vectoring with prompting, or both must be enabled.

    Basic vectoring allows the vector to place calls in a queue for a split and to check the size of a queue for a particular split. The latter facility is useful for directing a call to the split where it will receive the quickest response.

    Vectoring with prompting allows the vector to collect digits from a caller and use these digits for routing the call. This facility can be used to program an automatic answering service to direct the caller to the required department.

    Only a subset of the vector commands are available to the customer if either option is omitted. The following table lists the vector commands and the vector options that must be selected to enable them. Though not included in this table, the **adjunct routing** command is available only if Basic Vectoring and CallVisor ASAI (used with the ICM feature) are enabled.

Table 2-27. Vector Commands Available With Selected Options

Command	Basic	Prompting
announcement extension A_EXT	X	X
busy	X	
check backup split X priority P if	X	
collect n digits after announcement extension A_EXT		X
disconnect after announcement extension A_EXT	X	
goto step S if unconditional	X	X
goto step S if ...	X	
goto step S if digits equal 0123...		X
goto vector V if unconditional	X	X
goto vector V if ...	X	
goto vector V if digits equal 0123...		X
queue to main split X priority P	X	
Messaging split X for EXT	X	X
route to digits with coverage Y/N		X
route to number D if digits equal 1CD		X
route to number D if unconditional	X	X
stop	X	X
Wait time T secs hearing FDBK	X	

Legend: A_EXT — Any valid announcement extension
X — Hunt group or split number (1-32)
P — Priority at which call is to be queued
S — Vector step number
V — Vector number
EXT — Any valid extension
D — Digit string (1-16 digits)
CD — Digit string (1-16 digits)
Used to compare to collected digits
Y/N — Yes or No
T — Number of seconds to delay (0-998)
FDBK — Type of feedback (silence, ringback, or music)

Administration software *does not* allow a VDN extension to be entered as data in the listed fields of the following screen forms:

- Recorded Announcements
- Call Coverage Answer Group
    - Group Member Assignments
- Call Coverage Paths
    - Coverage Point Assignments
- Console Parameters
    - CAS Back-up Extension
- Feature-Related System Parameters
    - ACA Long Holding Time Originating Extension
    - ACA Short Holding Time Originating Extension
    - Extensions With System wide Retrieval Permission
    - Controlled Outward Restriction Intercept Treatment
    - Controlled Termination Restriction (Do Not Disturb)
    - Controlled Station-to-Station Restriction
- Hospitality-Related System Parameters
    - Extension of PMS Log Printer
    - Extension of Journal/Schedule Printer
    - Extension of PMS
    - Extension to Receive Failed Wakeup LWC Messages
- Hunt Groups
    - Supervisor Extension
    - Member Extensions
- Intercom Group
    - Member Extensions
- Listed Directory Numbers
    - LDN Extensions
- Loudspeaker Paging and Code Calling Access
    - Extension Numbers Assigned to Codes
- Pickup Groups
    - Member Extensions
- Remote Access
    - Remote Access Extension

- Terminating Extension Group
  - Member Extensions

Administration software *does* allow a VDN extension to be entered as data in specific fields of the following screen forms:

- Abbreviated Dialing Lists
- Hunt Groups
  - Night Destination
- Listed Directory Numbers
  - Night Destination
- Trunk Groups
  - Night Destination
  - Incoming Destination

Administration software *does not* allow a VDN extension to be entered as auxiliary data for the following buttons:

- Bridged Appearance (brdg-app)
- Data Call Setup (data-ext)

Administration software *does* allow a VDN extension to be entered as auxiliary data for the following buttons:

- Remote Message Waiting Indicator (aut-msg-wt)
- Facility Busy Indication (busy-ind)
- Manual Message Waiting (man-msg-wt)
- Manual Signaling (signal)

For more detailed information on the administration of Call Vectoring, see *DEFINITY® Communications System Generic 3i — Implementation, 555-230-650*.

Hardware and Software Requirements

No additional hardware is required. Call Vectoring software is required.

CallVisor Adjunct/Switch Application Interface (ASAI) (G3i-U.S.)

Description

Provides an Integrated Services Digital Network (ISDN) based interface between the switch and one or more adjuncts. Through the services provided by the CallVisor ASAI, the adjunct(s) improves efficiency by initiating, receiving, and controlling calls on behalf of ACD agents or other system users. All agent call handling features are supported through CallVisor ASAI.

The CallVisor ASAI is an ISDN-BRI link that uses Q.932 based procedures and messages. A special information element in the ISDN messages, the Q.932 Facilities Information Element, carries CallVisor ASAI information across the interface. Typically, this interface is used with the Inbound Call Management (ICM) features, outbound call management (OCM) applications, and application programming interfaces for office messaging applications.

The system can have up to eight CallVisor ASAI BRI interfaces.

CallVisor ASAI Capabilities

The following capability groups are provided by CallVisor ASAI:

- Call Control Group
- Domain (Station/ACD Split) Call Control Group
- Notification Group
- Set Value Group
- Value Query Group
- Request Feature Group
- Routing Group
- Maintenance Group

The above capability groups are described in more detail in the following paragraphs.

NOTE:
The Domain (Station/ACD Split) Call Control Capability Group is a actually a subsection of the Call Control Capability Group. For the purpose of this document, the Domain (Station/ACD Split) Call Group will be treated as a separate capability group.

Call Control Capabilities Group

Call control capabilities are used by the adjunct to initiate, control, and terminate calls to which it is not a party. The following capabilities make up the Call

Control capabilities:

- Initiating Capabilities:
  - — Third Party Make Call — Used by the adjunct to set up a call. This capability supports the following types of adjunct-controlled calls:
    a. Switch-classified
    b. User-classified
    c. Direct-agent
    d. Supervisor-assist
  - — Third Party Take Control — Used to take control of a call that is already in progress if other third party call control functions (such as hold, drop, and so on) are desired.
- Control Capabilities:
  - — Third Party Selective Drop — Used by the adjunct to request that a party be dropped from the specified adjunct-controlled call.
  - — Third Party Selective Hold — Used by the adjunct to request that a party on a specified adjunct-controlled call be placed on hard hold.
  - — Third Party Reconnect — Used by the adjunct to request that a party be reconnected to a specified adjunct-controlled call.
  - — Third Party Merge — Used by the adjunct to request that two adjunct-controlled calls (already existing on the switch) be merged.
- Terminating Capabilities:
  - — Third Party Relinquish Control — Used by the adjunct to request that a specified adjunct-controlled call no longer send call events. The call must have been previously controlled by the same adjunct.
  - — Third Party Clear Call — Used by the adjunct to request that a specified adjunct-controlled call be dropped (all parties on that call are dropped and all resources are released from the call).
  - — Third Party Call Ended — Invoked by the switch as a result of an adjunct-controlled call being dropped. This capability lets the switch terminate the exchange of CallVisor ASAI-related messages between the switch and the adjunct.
- Event Reporting Capability:
  - — Event Report — Used by the switch to convey call feedback and event information in the form of event reports to the adjunct. Call events are sent to the adjunct for adjunct-monitored calls, which include controlled and active notification calls.

Call Types Supported by the Third Party Make Call Function

The Third Party Make Call function is used by the adjunct to set up the following types of calls:

- Switch-Classified Calls
- User-Classified Calls
- Direct-Agent Calls
- Supervisor Assist Calls

The above listed call types are described in the following paragraphs.

Switch-Classified Calls

This type of call is used primarily in OCM applications with AutoPace or Predictive dialing. An adjunct initiates Switch-Classified calls on behalf of the originator (extension number associated with a split, hunt group, or announcement) with the expectation that only the answered calls will be transferred to the originator; other calls are dropped by the switch.

With this type of call, the destination is alerted first and the call is then classified using Call Classification. (Call Classification is described in the OCM feature description, elsewhere in this chapter). The destination can be either a valid switch extension or an external number.

It is recommended that trunks with answer supervision be used for switch-classified outbound calls. Answer supervision from the network over the trunk facility provides an accurate and quick indication of when the far end answers.

User-Classified Calls

This type of call is set up by the adjunct on behalf of an originator to an internal or external destination. It is a call whose originator is alerted first. (If the originator has a station type that the switch can take off-hook remotely, it will do so without alerting the originator.) However, the call does not have the ‘‘Direct Agent’’ or ‘‘supervisor assist’’ call options. It is ‘‘user classified’’ because the user decides what to do with the call: allow continued ringing, drop the call, or speak to the answering party. The user listens to network-provided tone in order to determine call outcome.

This type of call is typically used in OCM applications with ViewFirst Dialing or Preview Dialing. It can also be used to place a new call for the purpose of merging with an existing call. The adjunct must be able to determine the originator’s extension and the destination address (either supply them directly or obtain them from the agent via the data terminal keyboard).

Valid originators for this kind of call are all voice terminal extensions. Valid destinations are either internal extensions (including Vector Directory Numbers (VDNs) or external numbers.

Direct-Agent Calls

This type of call is originated by the adjunct on behalf of any voice terminal user to an ACD agent logged into a specified split (as contrasted to an ACD call in which the switch selects the agent). This may be used with either ICM or OCM, whenever the user decides that the customer should talk to a specific ACD agent. This type of call is also used with call routing.

Valid originators for this kind of call are all voice terminal extensions (or VDN callers). Valid destinations are ACD agent extensions.

Supervisor Assist Calls

This type of call is set up between an ACD agent's extension and another voice terminal extension (typically a supervisor in the ACD split). This is used in either ICM or OCM applications when an agent (either while on the phone with a client or when idle) wants to consult with the supervisor.

Domain (Station/ACD Split) Call Control Capabilities Group

Domain (station/ACD split) call control capabilities allows an adjunct to receive event reports and control all calls beginning and ending at a specific station extension. Without these capabilities, similar control and monitoring functions would require the adjunct to have adjunct control of the call at a station. It allows the adjunct to control only the station extension associated with the domain-control association instead of allowing control of all parties (extensions) on a call. With these capabilities, the adjunct can:

- control all calls originating and ending at a station extension
- monitor select calls only at that specific station
- begin outbound calls from that station

NOTE:
The Domain (Station/ACD Split) Call Control Capability Group is a actually a subsection of the Call Control Capability Group. For the purpose of this document, the Domain (Station/ACD Split) Call Group will be treated as a separate capability group.

The following capabilities make up the Domain (Station/ACD Split) Call Control capabilities:

- Initiating Capabilities:
  - — Third Party Domain (Station) Control Request — Used by the adjunct to receive notification and control all calls at a specified extension.
  - — Third Party Domain Control Request for ACD Split Domain — Used by the adjunct to receive event reports at the specified split domain. Currently, only the Logout Event Report is available.
- Control Capabilities:
  - — Third Party Answer — Used by the adjunct to request on behalf of a station user to "answer" a ringing, bridged, or held call present at a station. Answering a ringing, bridged, or held call means to

connect a call by forcing the station off hook, if the user is on hook, or cutting through the call to the headset or handset, if the user is off hook (listening to dial tone or being in the off hook idle state).

- — Third Party Selective Drop — Used by the adjunct to drop a controlled extension from a call.
- — Third Party Selective Hold — Used by the adjunct to place a controlled extension on hold.
- — Third Party Reconnect — Used by the adjunct to reconnect a held call to a station.
- — Third Party Merge — Used by the adjunct to request that two adjunct-controlled calls (already existing on the switch) be merged.
- — Third Party Autodial — Used by the adjunct to set up a two-party call between the station and an internal or external destination. This capability can only be requested by an application having an active domain-controlled association.

- Terminating Capabilities:
  - — Third Party Relinquish Control — Used by the adjunct to end a domain-control association.
  - — Third Party Domain Control Ended — Used by the switch to inform an application that a domain-control association has been terminated because the domain was removed or changed to become an invalid domain by administration.
- Event Reporting Capability:
  - — Event Report — Used by the switch to convey call feedback and event information in the form of event reports to the adjunct. Call events are sent to the adjunct for adjunct-monitored calls, which include controlled and active notification calls.

Notification Capabilities Group

Notification capabilities are used by adjuncts to request and terminate event reporting on certain calls. The Notification capabilities are as follows:

- Event Notification Request
- Event Report
- Event Notification Cancel
- Event Notification Ended
- Stop Call Notification

An adjunct uses the Event Notification Request function to request notification of calls to VDNs and ACD splits. Such splits cannot be vector-controlled or adjunct-controlled.

Once the adjunct requests event notification on a VDN or split, the VDN or split is referred to as an ‘‘active-notification’’ VDN or split. From this point on, any call entering this active-notification VDN or split will cause the switch to start sending call events to the monitoring adjunct regarding the call, and the call becomes an active-notification call (and therefore an adjunct-monitored call). The switch will continue to send call events on the active-notification call until the call is disconnected, dropped, abandoned, or enters another active-notification VDN or split.

Call events are sent to the adjunct via the Event Report capability. Call events may be any of the following:

- Alerting
- Trunk Seized
- Cut-through/Progress
- Disconnect/Drop
- Queued
- Busy/Unavailable
- Answered
- Connected
- Reorder/Denial
- Call Initiated
- Call Offered to Domain
- Call Transferred
- Call Conferenced
- Call Ended
- Call Redirected
- Hold
- Logout
- Reconnected

The Stop Call Notification capability will stop an adjunct from receiving call events for a specific call. The switch will no longer send call events on that call to any adjunct, until the call enters another active-notification VDN or ACD split, or until another Third Party Take Control request on this call.

An adjunct can cancel a notification request with the Event Notification Cancel capability when it no longer wishes to receive call events for a particular VDN or ACD split.

If a switch administration change causes a VDN or ACD split to become invalid, the switch will inform the adjunct using the Event Notification Ended capability.

Set Value Capabilities Group

Set Value allows the following items to be set to a specified value:

- Message Waiting Lamp

  The adjunct uses this capability to set the state of a message waiting lamp.

Value Query Capabilities Group

The Value Query capability allows the adjunct to request information from the switch about the status or value of switch resources.

The following value queries are supported:

- **Time of Day Query** — The adjunct asks for the time. The switch response contains the year, month, day, hour, minute, and second.
- **ACD Split Query** — The adjunct sends a valid split extension in a value query message. The switch responds with the number of ACD agents available to receive calls in that split, the number of calls in queue, and number of ACD agents logged in.
- **Trunk Group Query** — The adjunct sends a trunk access code in a value query message. The switch responds with the number of idle trunks in the trunk group and the number of in-use trunks.
- **Call Classifier Query** — The adjunct sends a classifier request in a value query message. The switch responds with number of idle and in-use TN744 ports.
- **ACD Agent Status Query** — The adjunct sends an ACD split and an ACD agent extension in a value query message. The switch responds with the work mode and idle/busy state of the ACD agent.
- **Extension Query** — The adjunct sends a voice terminal extension identifier in a value query message. The switch responds with the talk state of the voice terminal.
- **ACD Agent Login** — The adjunct sends an ACD split extension and an ACD agent login audit flag in a value query message. The switch responds with a list that contains the extensions for each agent logged into the split.
- **Calls Query** — The adjunct passes an extension in the value query request. The switch responds with the call_id(s) for the calls present at the primary's extension, and also sends the party_id and party_state (for example, alerting, connected, and hold) of that station for each call. Additionally, information about a maximum of 10 calls present at the station is reported back to the adjunct. This query is valid for stations only.
- **Station Status Query** — The adjunct uses this query for status information on the extension number of the endpoint.
- **Party ID Query** — The adjunct sends a party_id in a value query

message. The switch responds with the party_id and the extension number for all the local parties on the call.

- **Station Feature Query** — The application passes an extension number and the feature to the switch. The extension need not be domain-controlled, but it must support the particular feature. The following features may be queried:
  - Message Waiting Indication
  - Send All Calls
  - Call Forwarding

Request Feature Capabilities Group

This capability allows the adjunct to request activation of the following switch features:

- **ACD Agent Login** — Log in agent to an ACD split. The following parameters are required for this function:
  - Login Identifier (password) — any string of length equal to or greater than that administered on the switch.
  - ACD Split Extension
  - Agent Extension
  - Mode: After Call Work, Auto-In, Manual-In, Auxiliary Work (optional, corresponds to initial work mode; if not specified, the default is Auxiliary Work)
- **ACD Agent Logout** — Logout agent from an ACD split. The following parameters are required for this function:
  - ACD Split Extension
  - Agent Extension
- **ACD Agent Change of Work Mode** — Change work mode of ACD agent to another mode. The following parameters are required for this function:
  - ACD Split Extension
  - Agent Extension
  - Mode: After Call Work, Auto-In, Manual-In, Auxiliary Work (corresponds to new work mode)
- **Send All Calls**
- **Call Forwarding**

The switch may acknowledge or deny the request.

Adjunct-Controlled Splits

The switch may have adjunct-controlled splits. An adjunct-controlled split is an ACD split designated as adjunct-controlled via switch administration. An adjunct-controlled split has the following properties:

- An adjunct-controlled split must have an administered controlling adjunct (CallVisor ASAI link extension) or a BX.25 interface.
- Agent login may only be done by the adjunct via Request Feature capabilities.
- An agent may be logged into only one such split at any given time.
- Change of work modes may only be done by the adjunct via Request Feature capabilities.
- Work mode changes take effect as soon as they are requested and processed.
- Agent logout may be done by the adjunct via Request Feature capabilities or by the agent. When logout is caused by action at the voice terminal, a logout event report is sent to the adjunct associated with the split (provided domain control has been requested for that split).
- Agents logged into an adjunct-controlled split have their voice terminal locked (all buttons and the touch tone pad) for the duration of the login (as long as the corresponding CallVisor ASAI link is up).
- A nonadjunct-monitored call directed to an adjunct-controlled split will receive busy tone.
- A nonadjunct-monitored call directed to an agent logged into an adjunct-controlled split will receive busy tone.
- Agents logged into an adjunct-controlled split may not make any calls from the voice terminal. All telephony functions must be done via third party call control.
- Adjunct-controlled splits may not have notification active. A request notification for an adjunct-controlled split will be denied.
- An adjunct-controlled split may be vector-controlled as well.

Although the Request Feature capability is mandatory in order to work with an adjunct-controlled split, by itself it only provides the ability to log in and log out agents. In order for agents to receive calls, ‘‘Call Control’’ must be assigned.

Routing Capabilities Group

The routing capabilities allow the switch to request (from the adjunct) routing instruction for a call. Incoming call information (such as CPN/BN and DNIS) could be used by the adjunct to route the call to an internal number, external number, a split, a VDN, an announcement extension, or a particular agent. The adjunct might also determine the best route for the incoming call based on the called number, the dialed digits from a call prompter, the customer database, or agent availability. Using this information, the adjunct can also provide priority ringing, priority queuing, or personalized handling of incoming calls by rerouting the call to a VDN or a specific agent. These capabilities require that Basic Call Vectoring be enabled. If digits will be collected from the caller, Call Vectoring

Prompting must also be enabled.

The following Routing Capabilities are provided:

- **Route** — Adjunct routing is initiated with a **Route** capability to request routing information based upon the incoming call information. A routing request is only administrable through the basic Call Vectoring feature. The **Route** capability is initiated by the switch when it encounters the ‘‘adjunct routing’’ command in a call vector.
- **Route Select** — The **Route Select** capability is sent from the adjunct to the switch in response to the **Route** capability. It provides the switch with the destination address where the call will be routed. In addition, the adjunct can request the switch to route the call as a Direct Agent call and/or a priority call.
- **Route End** — The **Route End** capability is sent by the switch to terminate routing. A call that is already routed will not be affected by this termination. It contains success/failure to indicate if the call was successfully routed.

Maintenance Capabilities Group

The maintenance capabilities are used to disable and enable switch-administered alarms for periodic link maintenance and to obtain information about the condition of the CallVisor ASAI and the CallVisor ASAI link. The following Maintenance Capabilities are provided:

- **Heartbeat** — This capability enables the adjunct or the switch to send an application to application message and receive a response in order to determine the sanity of the application on the remote endpoint.
- **Suspend Maintenance** — This capability enables the adjunct to disable switch alarms on an CallVisor ASAI link for maintenance functions.
- **Resume Maintenance** — This capability enable the adjunct to resume switch alarms on an CallVisor ASAI link.

CallVisor ASAI Applications

The following examples are a few of the many CallVisor ASAI applications:

- Data Screen Delivery

  The system can pass network information such as CPN/BN, DNIS and switch information such as digits collected via the integrated call prompter and agent extensions to a host/adjunct if the host/adjunct requests notification. The host/adjunct can use this information to automatically display the proper data screen.

- Data Screen Transfer

  The system can notify the host/adjunct of the transferred-to party, if the adjunct is monitoring the call. This allows the host/adjunct to transfer the data screen associated with the call.

- Adjunct Routing

  The system can request routing from the host/adjunct. The host/adjunct may respond with any valid internal/external number.

- Direct Agent Calling

  The host/adjunct may wish to route or transfer (using third party make call) a call to a particular agent, yet have this call treated as an ACD call and tracked as an ACD call by CMS. The Direct Agent Calling function provides this treatment.

- Supervisor Assist

  The host/adjunct may initiate a call on behalf of an agent to the agent's supervisor (as defined by the host/adjunct). This call is treated as a supervisor assist call by CMS.

- Voice Processor Integration

  The system can send information such as CPN/BN, DNIS, or agent selected to the speech processing adjunct(s). The speech processor can integrate this information with information the speech processor collects and send it on to a host for screen delivery or call routing.

- View First Dialing

  The system allows a host/adjunct to place a call on behalf of an agent. The agent may have previewed a data screen and used the data keyboard to tell the host/adjunct to begin this call.

- Auto Pace Dialing

  The system allows a host/adjunct to place a call on behalf of a group of agents. The system classifies the call, sending only answered calls to the agents. The system notifies the host/adjunct of each classification. Special Network Information Tones are detected and may be treated as answered or dropped.

- Agent Work Mode Control

  The system allows the host/adjunct to log agents into and out of various splits. The system also allows the host/adjunct to change agents' work modes. A supervisor might initiate this type of activity via the host/adjunct or the host/adjunct might automatically make a change due to work loads.

Considerations

A maximum of eight CallVisor ASAI links may be administered.

The switch does not recognize or address CallVisor ASAI messages to/from specific applications on an CallVisor ASAI link. Multiple applications may exist on the adjunct, but the CallVisor ASAI interface does not address (for example) event reports to a specific application; the adjunct will have to determine which of its applications gets the messages from the switch. The switch will, however, send multiple copies of the same event, one for each active association for a given call or user extension.

The maximum number of simultaneous notification requests is 170.

The maximum number of simultaneous CallVisor ASAI associations (excluding notification requests) is 300.

The maximum number of active adjunct-controlled calls is 300.

The maximum number of pending routing requests on a CallVisor ASAI link is 127.

The maximum number of splits is 99.

The maximum number of agents is 400.

The maximum number of VDNs is 500.

The maximum number of vectors is 256.

Interactions

The following features and functions interact with the CallVisor ASAI feature:

- Abbreviated Dialing

  When an agent is logged into an adjunct-controlled ACD split, programmed Abbreviated Dialing buttons or feature activation buttons are disabled.

  Abbreviated Dialing cannot be invoked through a Third Party Call Control request.

- Single-Line Voice Terminals

  If a single-line voice terminal user (with Automatic Answer) is logged into an adjunct-controlled split and the user goes on-hook, the user will be logged out (regardless of the work mode).

  For regular stations or ACD agents, third party call control may be used in conjunction with switch-hook flash operations. Appropriate events are reported. For simplicity, it is recommended that third party operations not be inter-mixed with manual ones.

  If a user-classified call is placed on behalf of a single-line voice terminal user, the user must either be off-hook (and not busy on a call), or go off-hook within five seconds of the call setup request. Otherwise, the call origination fails.

- Announcements

  An ACD split forced first or second announcements and vector announcements do not generate event reports for the adjunct. However, nonsplit announcements generate events which are sent to other parties on the

call.

Extensions assigned to integrated announcements may not be domain controlled. The Third Party Auto Dial capability may specify integrated announcement extensions as destination endpoints.

- Answer Detection

  CallVisor ASAI switch-classified calls compete with Answer Detection for ports on the TN744.

- Answer Supervision

  The Answer Supervision Time-out field determines how long the CO trunk circuit pack waits before sending the (simulated) answer message to the software. This is useful when true answer supervision is not available on a trunk. This message is used to send call information to CDR and to trigger the bridging of a service observer onto an outgoing trunk call. This timer is ignored if the trunk is expected to receive true answer supervision from the network (the switch uses the true answer supervision whenever available). Adjunct-monitored calls are treated like regular calls.

  With respect to switch-classified calls, when the Answer Supervision `y/n` field is set to `no`, the switch relies entirely on the call classifier to determine when the call was answered. When answer supervision on the trunk is set to `yes`, a switch-classified call is considered answered when the switch software receives the answer message from the trunk circuit pack. In reality, switch-classified calls may receive either an answer message from the trunk circuit pack or (if this never comes) an indication from the classifier that the far end answered. In this case, the switch will act on the first indication received and ignore any subsequent indications.

- Attendant Auto-Manual Splitting

  If an individual attendant receives a call with active domain-control associations, then activates the Attendant Auto-Manual Splitting feature, a Hold Event Report is returned to the associations controlling the extensions adjunct. The next event report sent depends on what button the attendant presses on the set (CANCEL = Reconnect, SPLIT = Conference, RELEASE = Transfer) .

- Attendant Call Waiting

  Calls that provide event reports over domain-controlled associations and are extended by an attendant to a local, busy, single-line voice terminal will generate the following event reports:

  - Hold - when the incoming call is **split away** by the attendant
  - Connect - when the attendant returns the call

  The following events are generated, if the busy station does not accept the extended call and its returns.

  - Alerting - when the call is returned to the attendant
  - Connect - when the attendant returns to the call

- Attendant Control of Trunk Group Access

  Trunks seized for Third Party Make Call attempts must not have attendant control activated. If they do, such calls will be denied. If the Route Select capability attempts to route to such a trunk, the step will fail and will be skipped.

- Attendant and Attendant Groups

  Individual attendants may be parties on adjunct-monitored calls and are supported like regular voice terminal users. The attendant group 0 is not supported.

- AUDIX

  Calls that cover to AUDIX do not maintain a simulated bridge appearance on the principal's station. The principal receives audible alerting followed by an interval of coverage response followed by the call dropping from the principal's set. When the principal receives alerting, the Alerting Event Report is sent. When the call is dropped from the principal's set, the Call Redirected Event Report is sent.

- Authorization Codes

  Authorization codes are not supported for the CallVisor ASAI Third Party Make Call capability. If the check for authorization codes fails, the call request is denied.

- Automatic Callback on Busy/Does Not Answer

  This feature cannot be activated by the adjunct over the CallVisor ASAI interface.

  This feature can be activated by a controlled station user, the callback will appear as an incoming call to the controlled-station association.

- Automatic Call Distribution

  When an agent is logged into an adjunct-controlled split, all display buttons are disabled. The display itself functions normally.

  Agents logged into adjunct-controlled splits have their voice terminal locked. Agents logged into adjunct-controlled splits must use Call Control and Request Feature capabilities to access voice terminal calling and ACD support features. Adjunct-controlled splits cannot receive any non-adjunct-monitored calls. Non-adjunct-monitored calls receive busy tone if they try to terminate at such splits.

  When the CallVisor ASAI link is down, adjunct-controlled splits behave like non-adjunct-controlled splits. Agents logged into such splits when the link is down have their voice terminals unlocked without being logged out. When the CallVisor ASAI link is restored, adjunct controlled splits return to being adjunct-controlled, and agent's voice terminals become locked again.

  Adjunct-controlled splits may also be vector-controlled. Splits that are both adjunct-controlled and vector-controlled have all the properties of both. Where there is a conflict, the more restrictive property applies. For example, an agent logged into such a split cannot log into any other split; the agent will have the voice terminal locked. Non-adjunct-monitored calls will not be allowed to terminate at such a split.

An event notification request will be denied if the split is an adjunct-controlled split or vector-controlled split.

In the case of a Direct Agent call joining the split queue, the destination agent's auto-in or manual-in button associated with the destination split will flash. Flashing will stop when no more Direct Agent calls are in the split queue waiting for this agent.

An agent cannot be logged into multiple splits if that agent is logged into an adjunct-controlled split.

When an agent is logged into multiple splits, then all Direct Agent calls destined for the agent will be serviced before all non-Direct Agent calls in all splits. When there is more than one split with Direct Agent calls waiting for the same agent, the Direct Agent call with the longest queue waiting time will be serviced first.

Direct Agent calls are not included in any of the existing measurements affecting queue status displays and buttons.

- Automatic Route Selection (ARS) and Automatic Alternate Routing (AAR)

  The ARS and AAR features are accessible by CallVisor ASAI adjuncts through Third Party Make Call requests. However, it is recommended that in situations where other applications use ARS trunks, ARS Routing Plans be administered using partitioning in order to guarantee use of certain trunks to outbound call applications. Each partition should be dedicated to a particular application/adjunct.

  When ARS/AAR is used, if the adjunct wants to obtain trunk availability information, it must query the switch about all trunk groups in the ARS partition dedicated for that application/adjunct. The adjunct may not use the ARS/AAR code in the query to obtain trunk availability information.

  When using ARS/AAR, the switch does not tell the adjunct which particular trunk group was selected for a given call.

  Care must be given to the proper administration of this feature, particularly the FRLs. If these aren't properly assigned, it is possible that calls will get denied regardless of trunk availability.

- Bridging

  Third party make calls delivered as non-ACD calls can alert bridging users. Appropriate events are sent for each bridging user and principal.

- Busy Verification of Terminals

  A domain-controlled station may be busy-verified. A Connected Event Report is provided when the verifying user is bridged in on a connection in which there is a domain-controlled station.

- Call Coverage

  All adjunct-monitored calls (except switch-classified calls) are allowed to go to coverage (either for the split or for the voice terminal) provided they are not priority calls and the coverage criteria are met.

  Direct Agent calls will follow the agent's coverage path rather than the split's.

  If an adjunct-monitored call goes to coverage and is answered or picked up by using the Call Pickup feature, the switch sends events to the adjunct.

  Switch-classified calls placed to destinations whose coverage criteria are met remain at the principal. Switch-classified calls attempting to be delivered to originators whose coverage criteria are met will follow the originator's coverage path.

- Call Forwarding

  All adjunct-monitored calls (except switch-classified) are allowed to be forwarded, even when the Priority Calling feature is enabled, provided the forwarding criteria are met.

  Direct Agent calls will forward if the destination split has call forwarding activated. After a Direct Agent call successfully terminates to the destination split, the call will forward if the destination agent has call forwarding activated.

  If any adjunct-monitored call is forwarded and is answered or picked up using the Call Pickup feature, then the switch sends an alerting and connected event to the adjunct with the actual party extension that was connected. If the call is forwarded off-premises via a non-PRI trunk, a ‘‘trunk seized’’ event is sent.

  Forwarded adjunct-monitored calls (including Direct Agent calls) are treated like regular ACD calls when offered to a split. They are treated like non-ACD calls when offered directly to a voice terminal user. When an adjunct-monitored call gets forwarded to an internal destination, a second alerting event is sent.

  Switch-classified calls offered to destinations with Call Forwarding active will remain at the destination. Switch-classified calls offered to originators with Call Forwarding active will be forwarded.

  Calls forwarded to an off-premises number on non-PRI trunks only get the trunk seized event, and, subsequently, the dropped event reports.

- Call Park

  A domain-controlled station can activate this feature. A call can also be parked on a domain-controlled station. When a call is parked on (or from) a domain-controlled station, the appropriate event report (Transfer or Conference) is sent. The ‘‘transferred-to’’ extension provided is the extension on which the call was parked.

- Call Pickup

A call alerting at a domain-controlled station may be picked up using this feature. The station picking up (either the principal or the pick up user or both) may be domain-controlled. A Connected Event Report is sent to all active association on the call when this feature is used. When a pickup user picks up the principal's call, the principal's set (if multi-function) retains a simulated bridge appearance and is able to connect to the call at any time. No event report is sent for the principal unless the principal connects in the call.

When a call has been queued first and then picked up by a pickup user, it is possible for an adjunct to see a connected event without having seen any prior alerting events.

- Call Prompting

  Up to 16 digits collected from the last collect digit vector command will be passed to the adjunct in the call offered Event Report and the Route capability.

  CallVisor ASAI switch-classified calls compete with Call Prompting for ports on the TN744.

- Call Vectoring

  Adjunct call routing is only administrable by the Call Vectoring feature.

  Third Party Make Call requests cannot have a VDN as the originator.

  Agents are not restricted to being logged into a single vector-controlled split (three is the limit).

  The agent's voice terminal is not locked while logged into a vector-controlled split.

  A vector-controlled split could be adjunct-controlled as well. If it is, the more restrictive properties of each will apply.

  A vector-controlled split is not mapped to an adjunct via administration, unless it is also adjunct-controlled.

  A vector-controlled split cannot be monitored.

- Call Waiting

  The Call Waiting tone is used to alert an analog user when a Direct Agent call is waiting.

- Class of Restriction (COR)

  Third Party Make Call attempts are placed using the originator's COR (voice terminal's or split's). The COR associated with the adjunct's link is not used at all.

  For switch-classified calls, if the destination's COR check fails, the call is dropped. COR checking is not done for the originator of a switch-classified call.

  A Direct Agent Calling field on the Class of Restriction form indicates whether the user can originate and receive Direct Agent calls. If either the originating or the destination party of a Direct Agent call does not have the proper COR, the call is denied.

In the case of adjunct routing, the COR of the associated VDN is used for calling party restriction checks.

- Conference/Transfer

  Switch-classified calls are not allowed to go to the attendant when transferred into a non-answering split.

  When an agent is logged into an adjunct-controlled split, Conference and Transfer can only be done via the agent's data terminal, since the voice terminal will be locked.

  A regular voice terminal user is allowed to transfer or conference an adjunct-monitored call. The adjunct is informed only when the transfer or conference operation is completed (second push of the transfer or conference button). The event reports contain a list of the parties remaining in the call as a result of the conference/transfer.

- Consult

  When the covering user presses the Conference or Transfer feature button and receives dial tone, a Hold Event Report is returned to all adjuncts active on the call. A Call Initiated Event Report is then returned to the covering user's adjuncts. After the Consult button is pressed by the covering user, Alerting and Connected Event Reports are returned to the principal's and covering user's adjuncts. The covering user can then conference or transfer the call.

- Distributed Communication Systems (DCS)

  Direct Agent calls cannot be made over a DCS link. The destination on a Direct Agent call must be an internal ACD agent extension.

  Third Party Make Call requests (excluding Direct Agent and switch-classified calls) can be placed over a DCS network. They are treated like off-switch calls.

- Do Not Disturb

  Activation of this feature by an ACD agent only blocks personal calls from terminating at the agent's voice terminal. ACD calls (including Direct Agent) are still delivered to the ACD agent when this feature is activated.

- Drop Button Operation

  The operation of this button is not changed with CallVisor ASAI.

  When **Drop** is pushed by one party in a two-party call, the Disconnect/Drop Event Report is sent with the extension of the party that pushed button. The originating party receives dial tone and the Call Initiated Event Report is reported on its domain-control associations.

  When **Drop** is pushed by the controlling party in a conference, the Disconnect/Drop Event Report is sent with the extension of the party who was dropped off the call. This might be a station extension or a group extension. A group extension is provided in situations when the last added party to a conference was a group (for example, TEG, split, and announcement) and **Drop** was used while the group extension was still alerting (or was busy). Since the controlling party does not receive dial tone (it's still connected to the conference), no Call Initiated Event Report is reported in this case.

- Extended Port Network (EPN)

  The Expansion Interface (EI) board (TN570) make it possible for CallVisor ASAIs to terminate on an EPN as well as the Primary Port Network (PPN).

  It is recommended that any CallVisor ASAIs that are critical to your company's business terminate on the PPN to enable the CallVisor ASAI to remain operational in the event of a fiber link or EI failure. Further, resources that are used by a critical CallVisor ASAI adjunct such as classifiers, trunks, announcements, and agent ports should also home on the PPN for the following reasons:

  — To keep these resources in service in the event of a fiber link or EI failure; and

  — To minimize the amount of cross carrier traffic that could degrade CallVisor ASAI response time and system performance.

- Forced Entry of Account Codes

  Third Party Make Call attempts to trunk groups with Forced Entry of Account Codes will be denied.

- Hold

  When an agent is logged into an adjunct-controlled split, Hold can be invoked only from the agent's data terminal. The adjunct must be able to invoke Third Party Selective Hold on behalf of a party. This party cannot be an attendant, trunk, announcement, vector, or split.

  When the voice terminal is not locked, and the user has an active adjunct-monitored call at the voice terminal, the user may place the call on hold and reconnect it to the voice terminal any number of times. Events are reported for hold when done either via switch-hook flash, hold button, or conference/transfer button.

- Hot Line

  A Third Party Make Call request made on behalf of a voice terminal that has this feature administered will be denied.

- Hunt Groups

  Event notification requests are not allowed for hunt groups (other than ACD splits). Therefore, the call offered event will never be sent for hunt groups. However, the queued event will be sent for hunt groups when an adjunct-monitored call queues at a hunt group.

- Interflow

  When an adjunct-monitored call interflows to another switch, adjunct notification will cease except for trunk events.

- Intraflow

  Direct Agent calls do not intraflow since they follow the agent's coverage path, rather than the split's.

- ISDN

  The Third Party Auto Dial calls will follow ISDN rules for the originator's name and number. The Call Initiated Event Report will not be sent for *en-bloc* BRI sets.

- ISDN-PRI Facilities

  ISDN-PRI Facilities may be used by either inbound or outbound adjunct calls.

  An incoming call over an ISDN-PRI facility (if so provisioned by the network) provides the calling and called party information which is passed on to the adjunct in the call offered Event Report and Route capabilities.

  An outgoing call over an ISDN-PRI facility provides call feedback events from the network.

  Switch-Classified calls always use a call classifier on ISDN-PRI facilities whether the call is interworked or not.

- Last Number Dialed

  Last Number Dialed is used for adjunct-monitored calls which are not switch-classified. A user pushing the last number dialed button will originate a call to the number last dialed on its behalf through a Third Party Make Call attempt. A call originated in this fashion will not be adjunct-monitored.

- Lookahead Interflow

  For the receiving PBX, the lookahead interflow information element passed in the ISDN message will be included in all subsequent call offered Event Report and Route capabilities for the call, when the information exists, and when the call is adjunct-monitored.

- Multiple Split Queuing

  When a call is queued in multiple splits/skill hunt groups, the party query will provide (in addition to the originator) only one of the split/skill hunt group extensions in the party list. When the call is de-queued, the Alerting Event Report will provide the split/skill hunt group extension of the alerting agent. There will be no other events provided for the splits from which the call was removed.

  A party query done while the call is queued on multiple splits/skill hunt groups will return only one of the split extensions.

- Music-On-Hold

  Music-on-hold (if administered and available) will be given to a party placed on hold from the other end either manually or via the adjunct.

- Night Service

  A controlled station can be a night service station.

- Outgoing Trunk Queuing

  Outgoing Trunk Queuing (ringback queuing) cannot be activated via CallVisor ASAI.

  Calls originated from a domain-controlled station (including Third Party Auto Dial calls) and placed to a trunk or a station supporting this feature will not be allowed to queue. It is recommended that no Automatic Call Back (ACB) buttons be assigned to stations involved in third party calls (including domain control).

  However, if this feature has been requested by the user before the station becomes domain-controlled the following events will be reported:

  — Alerting Event Report - when the call alerts the station

  — Connected Event Report - when the station connects to the call

  — Disconnect/Drop Event Report - if the station does not answer the call and the call drops

- Personal Central Office Line (PCOL)

  Members of a PCOL may be domain-controlled. PCOL behaves like bridging for the purpose of CallVisor ASAI event reporting. When a call is placed to a PCOL group, the Alerting Event Report is provided to each member's associations. The called number information passed in the alerting message will be the default station characters. When one of the members answers the incoming call, the Connected Event Report is provided with the station which answered the call. If another members connects to the call, another Connected Event Report is provided. When a member goes on hook but the PCOL itself does not drop from the call, no event is sent but the state of that party changes from the connected state to the bridged state. The Disconnect/Drop Event Report is not sent to each member's associations until the entire PCOL drops from the call as opposed to an individual member going on hook.

  Members that are not connected to the call while the call is connected to another PCOL member are in the bridged state. When the only connected member of the PCOL transitions to the held state, the state for all members of the PCOL changes to the held state even if they were perviously in bridged state. There is no event sent to the bridged user associations for this transition.

  All members of the PCOL may be individually domain-controlled. Each will receive appropriate events as applicable to the controlled station. Call Control requests are not recommended. However, Third Party Selective Hold, Third Party Merge, Third Party Reconnect, and Third Party Selective Drop are not permitted on parties in the bridged state and may also be more restrictive if the exclusion option is in effect from a station associated with the PCOL.

A Third Party Auto Dial or Third Party Make Call will originate at the primary extension number of a user. For a call to originate at the PCOL call appearance of a primary extension, that user must be off hook at that call appearance at the time the request is received.

If a party ID is requested while the PCOL is alerting or on hold, one party member will be reported for the group with the extension number specified as the default extension.

If a call query is requested on an extension while the PCOL call is active, only one call appearance will be associated with the particular call ID.

- Priority Calling

  CallVisor ASAI capabilities permit both Priority Calling and Direct Agent Calling for the same call. When a Direct Agent call is also a priority call, the call follows priority call rules for delivery to the destination. That is, calls will be delivered with three bursts of distinctive ringing, and will not go to the covering point for Call Coverage or Send All Calls.

  Calls originated as third party are also permitted to be priority. If they are, the priority Call Coverage and alerting rules apply.

- Privacy-Manual Exclusion

  The exclusion feature can be activated associated with bridges, TEGs, or PCOLs. An exclusion button must be defined for the station that wishes to utilize this feature. Analog stations cannot utilize this feature since they do not have feature buttons. Activation of this feature when the station is a member of a bridge, TEG, or PCOL causes all other connected members of the group to transition to the bridged state. In addition, other members receive denial when they attempt to manually connect to the call.

  Pressing the exclusion feature button toggles the feature from on to off as indicated by the green light associated with the button. Activation of this feature will affect all call control requests associated with other members of the bridge, TEG, or PCOL, and will affect the use of the analog principal's station if activated by a bridging user.

- Queuing

  Direct Agent calls have priority over all non-Direct Agent calls in the split queue.

- Ringback Queuing

  This feature is not supported for Third Party Make Calls. When there are no more trunks available, an adjunct placed call will receive reorder.

- Send All Calls

  If the destination agent has Send All Calls activated, then Direct Agent calls go to the agent's coverage path.

  If priority calling is used, then the Direct Agent call will not go to the agent's coverage path and will remain in the queue.

Send All Calls can be activated/deactivated either manually or via Request Feature capability.

- Service Observing

  Service Observing may only be initiated from the observer's voice terminal. Any type of adjunct-monitored call may be service observed as long as the service observing criteria are met.

  For a switch-classified call, the observer is bridged onto the connection when the call is given to the monitored party. The observer receives the warning tone after the bridging is complete (provided the warning tone option is administered).

  For a user-classified call, the observer is bridged onto the connection when the destination answers. When the destination is a trunk with answer supervision, the observer is bridged onto the call when actual far end answer occurs. When the destination is a trunk without answer supervision, the observer is bridged onto the call after the trunk answer supervision time-out event.

- Single-Digit Dialing And Mixed Station Numbering

  A call initiated using the Third Party Auto Dial capability is permitted to use single-digit dialing.

- CDR

  Calls originated by the adjunct via the Third Party Auto Dial capability or Third Party Make Call capability are marked with the condition code ‘‘B’’. Adjunct originated calls include calls originated by forcing the user off hook after a Third Party Auto Dial request or Third Party Make Call request; calls originated by the user going off hook and then requesting Third Party Auto Dial or Third Party Make Call; and calls originated by the user going off hook, dialing a few digits, and then requesting Third Party Auto Dial or Third Party Make Call.

  Calls originated manually from a domain-controlled station will not be marked with condition code ‘‘B’’.

- Subnet Trunking

  This feature is supported for Third Party Auto Dial

- Supervisor Assistance

  This feature can be accessed in the conventional way from the voice terminal if the voice terminal is not locked. In this case, the call is placed to the switch-administered split supervisor.

  If the voice terminal is locked (under adjunct control), this feature may only be accessed via the adjunct. This feature may also be accessed via the adjunct for voice terminals that are not locked.

- Temporary Bridged Appearances

  The operation of this feature has not changed with CallVisor ASAI. There is no event provided when a temporary bridged appearance is created at a multi-function set. If the user is connected to the call (becomes active on such an appearance), the Connected Event Report is provided. If a user goes on hook after having been connected on such an appearance, a Disconnect/Drop Event Report with cause CS0/16 (normal clearing) is generated for the disconnected extension (bridged appearance).

If the call is dropped from the temporary bridged appearance by someone else, a Disconnect/Drop Event Report is also provided.

Temporary bridged appearances are not supported with analog sets. Analog sets get the Call Redirected Event Report when such an appearance would normally be created for a multi-function set.

The call state provided to queries on extensions with temporary bridged appearances will be bridged if the extension is not active on the call or connected if the extension is active on the call.

The Third Party Selective Drop request is denied for a temporary bridged appearance which is not connected on the call.

- Terminating Extension Group

  Members of a TEG may be domain-controlled. A TEG behaves similarly to bridging for the purpose of CallVisor ASAI event reporting. If controlled stations are members of a terminating group, an incoming call to the group will cause an Alerting Event Report to be sent to all domain-control associations for the terminating group. For the member of the group that answers the call, a Connected Event Report is returned to the answering member domain-control association(s) with the station which answered the call. All the domain-control associations for the other group members (non-answering members without TEG buttons) receive a Call Redirected Event Report. When a button TEG member goes on hook but the TEG itself does not drop from the call, no event is sent but the state of that party changes from the connected state to the bridged state. The Disconnect/Drop Event Report is not sent to each member's associations until the entire TEG drops from the call as opposed to an individual member going on hook.

  Members that are not connected to the call while the call is connected to another TEG member are in the bridged state. When the only connected member of the TEG transitions to the held state, the state for all members of the TEG changes to the held state even if they were previously in the bridged state. There is not event report sent to the bridged user associations for this transition.

  All members of the TEG may be individually domain-controlled. Each will receive appropriate events as applicable to the controlled station. Call Control requests will work normally if invoked over the station domain. However, Third Party Selective Hold, Third Party Merge, Third Party Reconnect, and Third Party Selective Drop are not permitted on parties in the bridged state and may also be more restrictive if the exclusion option is in effect from a station associated with the TEG.

  Third Party Auto Dial or Third Party Make Call requests cannot specify the

TEG group extension. TEGs can only receive calls, not originate them.

If a party ID is requested while the TEG is alerting or on hold, one party member will be reported for the group with the extension number specified as the TEG group extension.

If a call's query is requested on an extension while the TEG call is active, only one call appearance will be associated with the particular call ID.

- Timed Reminder

  Third Party Auto Dial calls extended by an attendant and not answered will redirect back to the attendant when the timed reminder interval expires. See the Attendant Call Waiting feature for events returned to the adjunct.

- Transfer

  Manual transfer from domain-controlled station is allowed subject to the feature's restrictions. The Hold Event Report is provided as a result of the first button push. The Transfer Event Report is provided as a result of the second button push, and only if the transfer is successfully completed. The Transfer Event Report is sent to all active associations for the resultant call.

- Trunk-to-Trunk Transfer

  When this feature is enabled, adjunct-monitored calls transferred from trunk to trunk will be allowed, but there will be no further notification (except for dropped sent to the adjunct).

Administration

The CallVisor ASAI feature is administered on a per-system basis. The following items require administration:

- System Parameters/Customer Options Form

  On this form, the CallVisor ASAI Interface option must be enabled. In addition, the following CallVisor ASAI Capability Groups (previously described in this feature description) may be optioned on the form:

  — Adjunct Call Control Group — Allows the adjunct to invoke the Third Party Call Control capabilities.

  — Adjunct Routing Group — Allows the adjunct to provide adjunct routing information to the switch for incoming calls.

  — Domain Control Group — Allows an adjunct to control calls and receive event reports for station sets and receive logout event reports for adjuncts in a given ACD split.

  — Event Notification Group — Allows the adjunct to request incoming call notification and enables the switch to send event reports about such calls.

  — Request Feature Group — Allows the adjunct to request features such as change work modes, login/logout, Send All Calls, and Call Forwarding.

— Set Value Group — Allows the adjunct to request status changes for Message Waiting lamps (that is, control the on/off state of the lamps).

The following table shows which capabilities are automatically enabled when CallVisor ASAI is enabled on the System-Parameters Customer-Options form.

Table 2-28. Automatically Enabled CallVisor ASAI Capabilities

Capability	Adjunct Call Control Group	Adjunct Routing Group	Domain Control Group	Event Notif. Group	Request Feature Group	Set Value Group
Call Ended Capability	X					
Event Reports	X		X	X		
Logout			X			
Alerting	X		X	X		
Answered	X		X			
Busy/Unavailable	X		X	X		
Call Ended Event (FAC message)				X		
Call Initiated			X			
Call Offered to Domain				X		
Call Redirected			X	X		
Call Conferenced	X		X	X		
Connected	X		X	X		
Cut-Through	X		X	X		
Denial/Reorder	X		X	X		
Disconnect/Drop	X		X	X		
Hold	X		X	X		
Queued	X		X	X		
Reconnected	X		X	X		
Call Transferred	X		X	X		
Trunk Seized	X		X	X		
Request Feature					X	
Notification Request				X		
Notification Cancel				X		
Notification Ended				X		
Route End		X				
Route Request		X				
Route Select		X				
Set Value						X
Stop Call Notification				X		

Continued on next page

Table 3-28. Automatically Enabled CallVisor ASAI Capabilities *(Continued)*

Capability	Adjunct Call Control Group	Adjunct Routing Group	Domain Control Group	Event Notif. Group	Request Feature Group	Set Value Group
Third Party Answer			X			
Third Party Auto Dial			X			
Third Party Call Ended	X					
Third Party Clear Call	X					
Third Party Domain Control			X			
Third Party Domain Control Ended			X			
Third Party Make Call	X					
Third Party Make Call Proceed	X					
Third Party Merge	X		X			
Third Party Reconnect	X		X			
Third Party Relinquish Control	X		X	X		
Third Party Selective Hold	X		X			
Third Party Selective Drop	X		X			
Third Party Take Control	X			X		

NOTE:
By enabling the CallVisor ASAI option(s) on Page 1 of the form, the following capabilities are automatically enabled as well, regardless of the groups selected on Page 2: Value Query (includes Response Continued Capability), Abort, Heart Beat, Restart Procedure, Suspend/Resume Alarm).

The capabilities shown here correspond to the operation values given in Table 4-14 (Operation Value/Error Value Coding) of the *DEFINITY® CallVisor™ Adjunct Switch Applications Interface (ASAI) — Protocol Reference*, 555-230-221:

- Hunt Group Form

  If an ACD split is to be adjunct controlled, the Controlling Adjunct field of this form must be administered as either ‘‘bx.25’’ (for CPN/BN) or ‘‘asai’’ (if the CallVisor ASAI Request Feature Group option has been enabled on the System Parameters Customer Options form).

  If the `Controlling Adjunct` field is administered as `asai`, an `ASAI Link Extension` field appears. This field should contain the extension of the administered CallVisor ASAI BRI link extension (see Station Form administration).

- Call Vector Form

The CallVisor ASAI Link Extension must be administered for the adjunct routing step on the Call Vector form.

- Class of Restriction Form

  Direct Agent Calling is allowed only if both the originator and the destination agent have a Class of Restriction (COR) that allows Direct Agent Calling.

- Trunk Group Form

  Trunk Groups must be administered to pass or receive CPN/BN calling party information.

  For an OCM application using network answer supervision, those trunks which support network answer supervision should have the `Answer Supervision` field set to `yes`.

- Station Form

  A CallVisor ASAI endpoint must be administered as a BRI voice terminal extension. A maximum of eight CallVisor ASAI endpoints may be assigned.

- Special Information Tone

  Each Special Information Tone (SIT) field (there is a total of six different SITs) must be administered as either dropped or answered. When a particular SIT is detected by the TN744 Call Classifier circuit pack, the switch will act in accordance with the choices indicated on this form.

For more detailed information on the administration of CallVisor ASAI, see *DEFINITY® Communications System Generic 3i — Implementation, 555-230-650*.

Hardware and Software Requirements

The system supports two types of CallVisor ASAI interfaces: CallVisor ASAI messages over the ISDN protocol using BRI and CallVisor ASAI-like messages over the BX.25 protocol using an EIA interface. BRI is an open interface used by AT&T platform vendors (IBM, DEC, Stratus, and Tandem) and independent software vendors. The BX.25 interface is a proprietary interface used by the AT&T ISDN Gateway product.

The system supports both the TN748C and the TN744 circuit packs for use as tone detectors. With respect to CallVisor ASAI features, the TN744 is required for those customers who desire switch call classification. Each port on the eight port TN744 acts as a touch-tone receiver or call classifier. Each call classifier port is capable of detecting tones as well as Special Information Tones. TN744 is also used by on-switch call prompting.

This tone detection will work only if the public network provides similar tones to those used in the U.S.

Each CallVisor ASAI BRI Interface Link requires a port on a TN556 ISDN-BRI circuit pack. Up to eight ports (maximum) may be assigned.

The system must be equipped with the TN778 packet control board.

If EPNs are present, TN570 expansion interface boards are required.

CallVisor ASAI Interface 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.

Generally, 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. These defaults for the number of rings may, however, be changed by the system administrator.

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.

NOTE:
Special ring types are not supported over DID facilities.

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 LWC 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 DDC or UCD 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. Brief call waiting tones are optimal because, while the tone is sounding, speech cannot be heard on the call.

Hardware and Software Requirements

No additional hardware or software is required.

Centralized Attendant Service (CAS)

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 CAS has its own 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 RLT.

The CAS attendants are located at the main location. The main location can be a DEFINITY Communications System Generic 1, a DEFINITY Communications System Generic 3i, a DEFINITY Communications System Generic 2.1, System 85, a DIMENSION PBX, or a System 75 (V3).

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 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 (the tones may vary by country):

- 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. 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 terminal; 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, DEFINITY Generic 1s and Generic 3is can function as branches or as the main; 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 seven consoles (including a night console).

A network with CAS can also be a DCS, 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 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 reextend 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 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 dialtone.

- 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.

- Leave Word Calling

  If a nessage is left for a user on a branch switch and the attendant on the main switch tries to retrieve the message by using LWC message retrieval, permission will be denied.

- 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 Trunk Answer 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.

- CDR

  If the CAS main RLT trunk has the CDR option selected, CDR 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)
- 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 (TN760D supports A-law). The TN760B will serve all other tie trunk applications in addition to CAS. As an alternative, the TN722 DS1 Tie Trunk circuit pack (TN464B/C/D support A-law) can be used for the release link trunks of the CAS network.

CAS software is required.

Class of Restriction (COR)

Description

Defines up to 96 different classes of call origination and termination privileges. Systems may have only a single COR, one with no restrictions, or may have as many CORs (up to 96) as necessary to effect the desired restrictions.

A COR is assigned to each of the following:

- Attendant consoles (as a group)
- Authorization Code
- ACD split
- Code Calling Access zone
- Data module
- DDC group
- Individual Attendant Consoles
- Loudspeaker Paging Access zone
- Remote Access barrier code
- Terminating Extension Group
- Trunk group
- UCD group
- Voice terminal
- VDN G3i

Use of CORs can be categorized as follows:

- Calling party restrictions
- Called party restrictions
- Forced entry of account codes
- Partitioned Group Number
- Service Observing
- Priority Queuing
- Time of Day Plan Number
- Direct Agent Calling (G3i)
- Facility Access Trunk Test
- Fully Restricted Service
- Restriction Override

- Restricted Call List (G3i)
- Unrestricted Call List (G3i)
- Miscellaneous restriction groups
- Selective denial of public network calling through a CCSA or EPSCS network
- An ARS or AAR FRL for control of call routing

Features assignable as calling party restrictions are as follows:

- Code Restriction
- Origination Restriction
- Outward Restriction
- Toll Restriction
- All-Toll Restriction (G3i)
- TAC-Toll Restriction (G3i)

Features assignable as called party restrictions are as follows:

- Inward Restriction
- Manual Terminating Line Restriction
- Termination Restriction
- Public Restricted

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 discussed in the following paragraphs. As an aid to understanding CORs, the screen used to administer CORs is shown on the following page. The values shown on the form are the default values for a G3i system. However, these values can be changed to implement the desired restrictions. The `Restricted Call List`, `Unrestricted Call List`, `Direct Agent Calling`, and `Facility Access Trunk Test` fields only apply to G3i systems. Also, the `Unrestricted Call List` field is displayed only if the calling party restriction is administered as `TAC-Toll` or `All-Toll`.

```
                                                            Page 1 of 2
                              CLASS OF RESTRICTION

              COR Number: xx                                 FRL: _
                    APLT? _           Calling Party Restriction: ___________
  Partitioned Group Number: __         Called Party Restriction: ___________
       Service Observing? _       Forced Entry of Account Codes? _
        Priority Queuing? _
     Restricted Call List? _         Facility Access Trunk Test? _
   Unrestricted Call List: __ __ __ __ __ __ __ __ __ __
     Restriction Override: _________
                                        Fully Restricted Service? _
```

```
                                                           Page 2 of 2

    CALLING PERMISSION SET 1 (Enter "y" to grant permission to call
    specified COR)

      0? _     12? _     24? _     36? _     48? _     60? _     72? _     84? _
      1? _     13? _     25? _     37? _     49? _     61? _     73? _     85? _
      2? _     14? _     26? _     38? _     50? _     62? _     74? _     86? _
      3? _     15? _     27? _     39? _     51? _     63? _     75? _     87? _
      4? _     16? _     28? _     40? _     52? _     64? _     76? _     88? _
      5? _     17? _     29? _     41? _     53? _     65? _     77? _     89? _
      6? _     18? _     30? _     42? _     54? _     66? _     78? _     90? _
      7? _     19? _     31? _     43? _     55? _     67? _     79? _     91? _
      8? _     20? _     32? _     44? _     56? _     68? _     80? _     92? _
      9? _     21? _     33? _     45? _     57? _     69? _     81? _     93? _
     10? _     22? _     34? _     46? _     58? _     70? _     82? _     94? _
     11? _     23? _     35? _     47? _     59? _     71? _     83? _     95? _
```

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 Code Restriction, Origination Restriction, Outward Restriction, and Toll Restriction. These individual features are fully described elsewhere in this chapter. A brief description is given here:

- Code Restriction (G1.1) — 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 (G1.1) — 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.) This restriction applies to calls made using trunk access codes or CO or FX trunk groups. See the Restriction — Toll/Code feature description elsewhere in this manual for more details.

- TAC-Toll Restriction (G3i) — Prevents callers at specified voice terminals from making trunk access calls to certain toll areas as defined on the system's administered Toll Analysis form, unless the number is on an Unrestricted Call List associated with the caller's COR. This restriction applies to calls made using trunk access codes or CO or FX trunk groups. See the Restriction — Toll feature description elsewhere in this manual for more details.
- All-Toll Restriction (G3i) — This restriction is identical to the TAC-Toll Restriction described above, except this restriction also applies to ARS calls. See the Restriction — Toll feature description elsewhere in this manual for more details.

Called party restrictions prevent specified users from receiving certain calls. Features assignable as called party restrictions are Inward Restriction, Manual Terminating Line Restriction, Termination Restriction, and Public Restriction. These individual features are fully described 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.
- Public Restriction — Restricts users of specified voice terminals from receiving direct public network calls.

Looking at the screen form used to administer CORs (see Screen 2-24), the field labeled `Calling Party Restriction` and the field labeled `Called Party Restriction` are both administered as none. However, the field `Calling Party Restriction` could be administered as any of the other previously described calling party restrictions. Likewise, the field for `Called Party Restriction` could be administered as any of the other previously described called party restrictions. 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 groups, UCD groups, DDC groups, data modules, the attendant group, and individual attendant extensions.

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 Code Restriction or All-Toll Restriction (G3i), but there will be no restrictions on incoming calls.
  - Calling party restriction=Code (G1.1) or All-Toll (G3i)
  - 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 UCD group for answering business calls only. These terminals are not to be used individually.
  - Calling party restriction=Origination
  - Called party restriction=Termination.

Calling and called party restriction is checked at initial ‘‘termination’’ and at redirection points.

Each COR is established as needed and is arbitrarily identified by a number, 0 through 96. For example, if the COR for the storeroom is 12, the storeroom voice terminal(s) is assigned COR 12.

Forced Entry of Account Codes

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. Account code dialing can be optional or mandatory (forced) on a per-COR basis.

Looking at the screen used to administer CORs (screen 2-24), the field labeled `Forced Entry of Account Codes` 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 for users placing designated calls that have the CDR Forced Entry of Account Codes activated on the Toll Analysis form and have a COR with a y in the field assigned to that user.

If this field is `y`, and the COR is assigned to a trunk group, all calls made using the TAC of that trunk group require an account code, no matter what number is dialed.

Partitioned Group Number

When AAR and 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 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.

If the Time of Day Routing feature is assigned, this field is replaced with a `Time of Day Plan Number` field. This field is used to assign each COR one of eight Time of Day Routing Plans to be used when routing AAR and ARS calls. Time of Day Routing provides the most economical routing of AAR and ARS calls based on the time of day or week that each call is made. For more information on Time of Day Routing, see the Time of Day Routing feature elsewhere in this chapter.

Service Observing

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

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.

DAC (G3i)

The `Direct Agent Calling` field on the COR form indicates whether the user can originate and receive Direct Agent calls through an adjunct. If either the originating or the destination party of a Direct Agent call does not have the proper COR, the call is denied.

Direct Agent Calling allows an adjunct to transfer a call to a particular agent and have the call treated as a call to a split. For more information on DAC, see the Automatic Call Distribution and Inbound Call Management features elsewhere in this chapter.

Facility Access Trunk Test

This field on the COR form is used to grant a user permission to make Facility Test Calls to access trunks. A y in this field allows users with this COR to make these calls. An n in this field causes users with this COR to receive intercept treatment when they attempt to make these calls. For more information on Facility Test Calls, see the Facility Test Calls feature elsewhere in this chapter.

Fully Restricted Service

Denies the specified voice terminal access to public network trunks for either incoming or outgoing completion. Note that Restriction Override should be set to No.

Restriction Override

Specifies type of user (att, all, none) who can override redirection restrictions. If "none" is chosen, then the COR will be checked when transfer or conference is used.

Restricted Call List (G3i)

A Restricted Call List is assigned to the system by the System Manager. This call list is made up of specific digit strings which cannot be dialed from facilities that are restricted by the call list. The `Restricted Call List` field on the COR form is used to determine which facilities are restricted by this call list. If a COR has this field administered as y, facilities with that COR cannot be used to dial a number that matches one of the digit strings in the Restricted Call List.

Unrestricted Call List (G3i)

Ten Unrestricted Call Lists are assigned to the system by the System Manager. Each Unrestricted Call List is made up of specific digit strings which can be dialed from facilities associated with the call list. The `Unrestricted Call List` fields on the COR form are used to determine which facilities have access to each of the ten call lists. If a COR has any of these ten fields administered with an Unrestricted Call List number, facilities with that COR can call any numbers contained in the assigned list(s) (unless the number is included in the Restricted Call List, which overrides the Unrestricted Call List).

Selective Denial of Public Network Calling Through a CCSA or 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 used to administer CORs (Screen 2-24), the field labeled APLT 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 would indicate that the facility can access CCSA or EPSCS off-network capabilities.

ARS/AAR 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 unless the trunk group is the originator of the call. 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 (Screen 2-24) 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 six routes in each of the up to 254 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 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 AAR, ARS, and 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 96 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 used during implementation (see Screen 2-24). 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 Screen 2-24, no restrictions apply because all 96 CORs are specified as `y`.

The G3i screen form shown on the following page gives an example of Miscellaneous Restriction groups. This form is for COR 6 as is indicated in the `COR Number field`. The 96 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 restrictions are checked on the initial call termination and on redirection.

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 the following screen, 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.

```
                                                                      Page 1 of 1
                                    CLASS OF RESTRICTION
                COR Number:  6                                    FRL: 7
                      APLT? y          Calling Party Restriction: none
  Partitioned Group Number: 1           Called Party Restriction: none
         Service Observing? n      Forced Entry of Account Codes? n
          Priority Queuing? n                 Direct Agent Calling? _
      Restricted Call List? _             Facility Access Trunk Test? n
    Unrestricted Call List: __  __  __  __  __  __  __  __  __  __
  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
```

When a COR is administered, the allowance or denial of access from that COR to each of the 96 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.

COR 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

- 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 CO trunks) — No restrictions were specified for these trunks. The default values on the screen form (see Screen 2-24) 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 CO 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 (TEG) — 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'' (G1.1) or ''tac-toll'' (G3i) 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` field 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` field 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 96 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.

- AAR/ARS

  Originating FRLs are assigned via a COR. Termination Restrictions do not apply to ARS/AAR calls.

- Bridged Call Appearance

  The COR assigned to a voice terminal's primary extension also applies to calls originated from a bridged call appearance.

- Call Coverage

  Users who may normally be restricted from calls can still receive calls directed to them via Call Coverage only if the Restriction Override is set to ‘‘all’’.

  When a call goes to coverage, it is the called party's (not the covering party's) restrictions that will be used.

- Call Forwarding All Calls

  If a call would normally be restricted between the forwarding and forwarded-to extensions, Call Forwarding activation is denied. Since restrictions are always checked when calls are redirected, each forwarded call will be checked for restrictions. If restrictions are found, the call will be forwarded only if Restriction Override is set to ‘‘all’’. However, if restrictions are assigned after Call Forwarding is activated, any termination restrictions are ignored.

- Code Restriction (G1.1)

  This feature is assigned to an originating facility via a COR. (For Code

Restriction to take effect, Code Restriction must be 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

  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.) With G3i, TAC-toll restriction can be disabled for specific outgoing trunk groups on the trunk group form.

Administration

COR is administered by the System Manager. For each COR which is assigned, the following items must be administered:

- COR Number
- 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 CDR (yes or no)
- Partitioned Group Number
- Priority Queuing (yes or no)
- Service Observing (yes or no)
- Time of Day Plan Number
- Direct Agent Calling (G3i)
- Facility Access Trunk Test
- Restricted Call List (G3i)
- Unrestricted Call List (G3i)

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 UCD groups.

The main items of concern for individual voice terminals are calling party restrictions and called party restrictions (discussed previously under ‘‘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 Screen 2-25, access to trunk groups with a COR of 3, 7, or 10 are denied to facilities with a COR of 6.

Calling party and called party restrictions should be ‘‘none.’’ Whether or not a CO or FX trunk group is restricted is specified on the trunk group form used during implementation.

With G1.1, if Toll Restriction is to apply on a call, 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	none

With G3i, CO and FX trunk groups default to being toll restricted for TAC calls. Toll Restriction for TAC calls can be disabled for certain CO/FX trunk groups on the trunk form.

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 2-29), 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, 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 (COS)

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
- Console Permission
- Client Room

There are only two choices for each feature; a voice terminal user or individual attendant *can* or *cannot* access the feature.

There are 16 possible COSs. 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. Which COS numbers represent which combination of allowed/denied features are given in the *DEFINITY® Communications System Generic 3i — Implementation, 555-230-650*.

In addition to Automatic Callback, Call Forwarding All Calls, Data Privacy, and Priority Calling, DEFINITY Generic 1 and Generic 3i offer the following functions:

- Off-Hook Alert

  This function can be administered only if the optional Emergency Access to the Attendant 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 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 COR, not a COS.

Considerations

COS is used to assign as many as seven features. 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

A COS is assigned to each voice terminal extension by the System Manager. The parameters for each COS can be changed. The only other administration required is the assignment of a COS to each individual attendant and voice terminal.

A COS should be assigned on Data Module and Access Endpoint (G3i) forms. A separate COS should be used for data applications.

A COS can also be assigned to a remote access barrier code.

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, and so on.) 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 conferenced while accessing paging equipment. The attendant can, however, release the call after paging the called party.

- Conference — Terminal

  A call cannot be conferenced 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 two-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 COR for each of the nine individual paging zones and for the zone used to activate all zones simultaneously.
- Number of times (one to three) 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.
- Loudspeaker locations (name of zone).
- Three-Digit chime codes for extensions. The codes are combinations of the digits one through five.

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) 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.

Interactions

The following features interact with the Conference — Attendant feature:

- Bridged Call Appearance

  A Bridged Appearance button can be used to make conference calls.

- Call Vectoring (G3i)

  A call to a VDN can be included as a party in a conference call only after vector processing terminates for that call (for example, after a successful **route to** command).

- 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

On the System Parameters form there is a field where the administrator may specify lower limits on the number of parties that can be on a conference call. With central office trunks involved there can be up to five parties. Without CO trunks there can be from three to six.

Hardware and Software Requirements

No additional hardware or software is required.

Conference — Terminal

Description

Allows multi-appearance voice terminal users to set up six-party conference calls without attendant assistance. Single-line voice terminal users can set up three-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

The following features interact with the Conference — Terminal feature:

- Bridged Call Appearance

  A Bridged Appearance button can be used to make conference calls.

- Call Vectoring (G3i)

  A call to a VDN can be included as a party in a conference call only after vector processing terminates for that call (for example, after a successful **route to** command).

- Class of Restriction (COR)

  CORs are checked if Restriction Override is set to "none."

Administration

On the System Parameters form there is a field where the administrator may specify lower limits on the number of parties that can be on a conference call. With public network trunks (such as central office trunks) involved there can be up to five parties. Without CO trunks there can be from three to six. A conference tone is available through administration.

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 section.

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 section.

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 (ICI)

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 ICI is used in conjunction with Call Coverage are given in the Call Coverage feature description, elsewhere in this section.

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

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 elsewhere in this manual) 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 *DEFINITY® Communications System, Generic 1 and Generic 3i — Installation and Test, 555-230-104.*

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 features interact with the 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, and so on). 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 (which also includes Alphanumeric Dialing, Default Dialing, and Hotline 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 MPDM equipped with an ACU interface module. An analog modem interfaced with an ACU can also be used to provide dialing capability for a host computer.

The Administered Connections feature, described elsewhere in this chapter, may also be used to establish a data call.

This section describes the data call setup features for both DCP sets and ISDN BRI sets.

Data Call Setup for DCP Modules

Voice Terminal Dialing for DCP Data Modules

Allows voice terminal users to originate and control data calls from the voice terminal. DCP 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.

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 Data Extension button provides one-touch access to a data module.

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 discussed 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 accessed by a Data Extension button on a voice terminal. Only one Data Extension button can be administered for a single data module.

Voice terminal dialing has the advantage that the user may hear the different types of network tones.

For off-premises dialing, particularly for toll calls, the user may opt for voice terminal dialing, instead of keyboard dialing.

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 for DCP Data Modules

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. 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 2-29 lists the call progress messages.

To originate and disconnect a call using Data Terminal Dialing, the user presses ***BREAK*** 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 ***Originated/Disconnect*** 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.
- + character (wait) may be used to interrupt or suspend dialing until dial tone is received from the distant switch.
- , (pause) character may be used to place a 1.5-second pause in dialing (multiple , can be used).
- % (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.
- ***UNDERLINE*** or ***BACKSPACE*** may be used to correct previously typed characters on the same line.
- *@* 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. 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

Alphanumeric Dialing (G3i). Alphanumeric Dialing enhances Data Terminal Dialing by allowing a data terminal user to place a data call by entering an alphanumeric name. This capability makes Data Terminal Dialing both convenient and user-friendly. Instead of dialing a long string of numbers, the user can enter a simple alphanumeric name. For more detailed information, see the Alphanumeric Dialing feature description, elsewhere in this chapter.

Default Dialing (G3i). Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination by simply entering a carriage return at the `DIAL:` prompt. The data terminal user can still place calls to other destinations by entering the complete address after the `DIAL:` prompt (normal Data Terminal Dialing or Alphanumeric Dialing). For more detailed information, see the Default Dialing feature description elsewhere in this chapter.

Hotline Dialing (G3i). Hotline Dialing is discussed in detail in the Data Hot Line feature description elsewhere in this manual.

Table 2-29. Call Progress Messages for Keyboard Dialing for DCP

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.
ANSWERED - NOT DATA	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	Receiving a call	Notifies called user that the calling user abandoned the call.
NO TONE	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	Call in queue	Current position of the user in queue. XX-indicates position.
PROCESSING*	Call in queue	Notifies user when out of queue. Facility is available.
TIMEOUT*	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.

Continued on next page

Table 3-29. Call Progress Messages for Keyboard Dialing for DCP *(Continued)*

Displayed Message	Application	Meaning
INCOMING CALL-*	Receiving a call	Equivalent to ringing.
INVALID ADDRESS	Placing a call	The entered name is not in the Alphanumeric Dialing Table.
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	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	Placing a call	Notifies user that call entered a local hunt group queue. XX indicates position.

* Bell sounds when message is displayed.

Data Call Setup for ISDN-BRI Modules

Voice Terminal Dialing for ISDN BRI Data Modules

Allows ISDN BRI voice terminal users to directly originate a data call. To set up a data call, the user just presses the Data button on the ISDN BRI voice terminal, enters the desired number on the dial pad, and then presses the Data button again.

The following data functions are not supported by BRI terminals:

- One button transfer to data
- Return-to-voice
- Data call pre-indication
- Voice call transfer to data and data call transfer to voice

Data Terminal (Keyboard) Dialing for ISDN-BRI Data Modules

Allows a user to set up and disconnect data calls directly from a data terminal without using a voice terminal. The voice terminal functions of switchhook and the audible call progress tones are replace with keyboard dialing and text known as call progress messages. Unlike DCP, BRI is terminal dependent, meaning that the BRI data module, not the switch, prompts the user to enter information.

NOTE:
The 7500B Data Module also allows users to set up calls via its front panel. For more information on this feature, consult the *7500B Data Module — User's Manual 555-021-717*.

Before the user can make a data call using Data Terminal Dialing, `CMD:` must appear on the screen of the terminal. To access the `CMD:` prompt before placing a call, the user must press ***Enter*** on the keyboard a few times. If the `CMD:` prompt does not appear, the user must press ***Break A*** and ***T*** at the same time, and then press ***Enter***. To make a data call, the user types `dial`, enters a space, types the desired telephone number, and presses ***Enter*** at the `CMD:` prompt (For example, `dial 1234567`).

To disconnect a data call using Data Terminal Dialing, the user must first enter **+++** access the `CMD:` prompt. At the `CMD:` prompt, the user types `end` and presses ***ENTER***.

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.
- + character (wait) may be used to interrupt or suspend dialing until dial tone is received from the distant switch.
- , (pause) character may be used to place a 1.5-second pause in dialing. (multiple , can be used).
- % (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.’’
- ***UNDERLINE*** or ***BACKSPACE*** characters may be used to correct previously typed characters on the same line.
- *@* may be used to delete the entire line and start over with a new `CMD:` prompt.

Each line of dialing information may contain up to 36 characters.

Examples of dialing are as follows:

- CMD: d 3478
- CMD: d 9+(201) 555-1212
- CMD: d 8, 555-2368
- CMD: d 9+555-2368+%9999+123 (remote access)

Basic Digit Dialing. Regular digit dialing is provided through the Asynchronous Data Module (ADM) or 7500B Data Module. Digits from 0 to 9, "*", and "#" can be entered. This feature can be used by the user either from the associated 7500 Series voice terminal keypad or from the EIA terminal interface.

Alphanumeric Dialing. Alphanumeric Dialing enhances Data Terminal Dialing by allowing a data terminal user to place a data call by entering an alphanumeric name. This capability makes Data Terminal Dialing both convenient and user-friendly. Instead of dialing a long string of numbers, the user can enter a simple alphanumeric name. For more detailed information, consult the Alphanumeric Dialing feature description elsewhere in this chapter.

Default Dialing. Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination by either typing a ***d*** and entering ***Return*** at the `CMD:` prompt, or by pressing the data button twice. If no default dialing has been administered, the call will be disconnected in less than one second. The data terminal user can still place calls to other destinations by typing a ***d*** and the complete address at the `CMD:` prompt, and then entering ***Return*** (normal Data Terminal Dialing or Alphanumeric Dialing). This feature is mutually exclusive with the Data Hotline feature. For more detailed information, see the Default Dialing feature description elsewhere in this chapter.

Call Forwarding All Calls. Call Forwarding All Calls allows incoming data calls to be redirected to another extension that is designated by the user. Activation/deactivation of the feature is done either by the attendant or by the forwarding party itself through the dialing of a feature access code.

Data Hotline. Data Hotline is a security feature. The switch will terminate the call to a preadministered hotline destination. If a user enters an address either intentionally or unintentionally, the call processing will discard the address string received for the hotline endpoint. The call processing will automatically route the call just as if the hotline destination address had been entered by the user. This service does not impose any restriction on incoming calls received at the endpoint. This feature is mutually exclusive with the Default Dialing feature.

Administered Connections. An Administered Connection is an end-to-end connection between two access endpoints or data endpoints that is automatically established by the system whenever the system is restarted or the Administered Connection is administered and due to be active. The attributes of these connections are user-defined. To administer Administered Connections, use the Administered Connection form via the SAT.

Once the ADM has been administered as one endpoint of an administered connection, the system waits for the scheduled time to set up the connection. At the scheduled time, the system establishes the connection and maintains it for the length of time specified. Once the call is accepted, the set will enter into the continuous mode for the length of time specified. If the switch is rebooted during the continuous connection, the connection will reinitiate the call setup. At any time that the connection drops (for example, disconnected cabling), the switch will reinitiate the call setup.

Call Request. DEFINITY Generic 3i call processing will handle all various BRI Bearer data call requests that are presently defined. Some capabilities that are not supported by AT&T terminals may be provided by a non-AT&T terminal. The switch will complete most call requests. For those capabilities that the switch doesn't support, a proper cause value will return back to the terminal.

Cause Value. BRI stations will receive a cause or reason code that identifies why the call is being cleared. The BRI data modules will convert certain cause values to text messages and display them for the user.

Endpoint Initialization. BRI endpoints have to successfully complete endpoint initialization procedures in order to be fully operative. It is usually carried out at the time of installation, or as part of reconfiguration.

Multipoint Configurations on BRI ports. In a passive bus multipoint configuration, the system supports two BRI endpoints per port, thus doubling the capacity of the BRI circuit pack. When changing the configuration of a BRI from point-to-point to multipoint, the original endpoint need not be reinitialized. However, only endpoints that support SPID initialization can be administered in a multipoint configuration.

Exchange of User Information. The BRI protocol provides the users the capability of exchanging up to 128 octets of user information end-to-end. The information is passed in the User Information IEs to the receiving endpoint without being interpreted by the switch. However, there are some limitations to the exchange of User Information IEs.

Table 2-30. Call Progress Messages for Keyboard Dialing for BRI

Displayed Message	Application	Meaning
CMD:	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.
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	Receiving a call	Notifies called user that the calling user abandoned the call.
NO TONE	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	Call in queue	Current position of the user in queue. XX-indicates position.
PROCESSING*	Call in queue	Notifies user when out of queue. Facility is available.
TIMEOUT*	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.

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Table 3-30. Call Progress Messages for Keyboard Dialing for BRI *(Continued)*

Displayed Message	Application	Meaning
INCOMING CALL-*	Receiving a call	Equivalent to ringing.
WRONG ADDRESS	Placing a call	The entered name is not in the Alphanumeric Dialing Table.
PLEASE ANS-	Receiving a call	Originating voice terminal user has transferred call to data module using One-Button Transfer to Data.
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	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.

* 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. DEFINITY Generic 3i offers the enhancement of off-premises Multiple Line Dialing and additional call progress messages.

BRI has a voice to data restriction. A voice terminal cannot call a data terminal, and a data terminal cannot call a voice terminal.

BRI voice terminals cannot have Data Extension buttons. Although DCP sets have Data Extension buttons, these sets cannot have Data Extension buttons for BRI data extensions.

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 one voice terminal can be assigned buttons that access the same data module.

When multiple Data Extension buttons access a single data module, the access 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, users with access to the data module could 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. Only 22 of the 24 available digits in an abbreviated dialing number can be used for keyboard dialing. The remaining two digits must contain the ‘‘wait’’ indicator for tone detection.

- Call Coverage

  A hunt group made up of digital data endpoints should not be assigned a coverage path.

- 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.

- Default Dialing

  Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination.

- 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 or Data Origination can also be used to indicate that a conversion resource is needed.

- CDR

  With G3i, Data Call CDR records the use of modem pools on trunk calls.

- 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.

- 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 DLC board. The DLC provides eight ports for connection with asynchronous 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.
- Default Dialing — See the Default Dialing feature, elsewhere in this chapter, for administration of this feature.
- Alphanumeric Dialing — See the Alphanumeric Dialing feature, elsewhere in this chapter, for administration of this feature.

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 DCP data module requires one port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law). A DTDM shares the port with its associated voice terminal.

  Each BRI data module requires one port on a TN7556 BRI circuit port pack. Each BRI port may be shared by two endpoints, with each endpoint providing either voice or data capability. To support ISDN-BRI, the switch requires the TN778 Packet Control circuit pack. An ADM shares the port with its associated BRI voice terminal.

- ***7400A Data Module:*** The 7400A Data Module may be used instead of an MTDM when supporting the combined Modem Pooling feature. The 7400A Data Module supports asynchronous operation and provides a DCP interface to the switch and a RS-232C interface to the associated modem.

- ***7400B Data Module:*** The 7400B Data Module supports asynchronous data communications and can operate in the stand-alone mode for data only service or in the linked mode which provides simultaneous voice and data service (acts like a DTDM). The 7400B provides voice and data communications to 7400D series voice terminals and the 602A1 CallMaster voice terminal that have a connection to a data terminal or personal computer. The 7400B integrates data and voice into the DCP protocol required to interface the switch via a port on a Digital Line circuit pack. The 7400B may be used instead of an MPDM when asynchronous operation at speeds of 19.2 Kbps or less is required to provide a DCP interface to the switch for data terminals, printers, and so on. The 7400B does not support synchronous operation and keyboard dialing.

- ***7500B Data Module:*** The 7500B Data Module is a stand-alone unit that supports asynchronous or synchronous DCE and asynchronous DTE on the Basic Rate ISDN (BRI) switch interface. In asynchronous mode, the 7500B supports packet or circuit-switched data communications, and can be controlled via the front panel or the keyboard of a connected terminal. The following optional enhancements are available for the 7500B in an asynchronous DCE configuration: an RS-366 ACU interface and a second asynchronous EIA-232D interface. In synchronous mode, the 7500B supports circuit-switched or nailed-up data communications, requires either the Multi-purpose Enhancement Board or the High-Speed Synchronous Enhancement Board, and can only be controlled via the front panel. In order to be configured as a synchronous DCE, the 7500B must have either the Multi-purpose Enhancement Board or the High-Speed Synchronous Enhancement Board.

  When configured as an asynchronous DTE, the 7500B provides an EIA-232D interface and supports full-duplex data transmission at rates of up to 19200 bps. This configuration is most commonly used for modem pooling applications. Regardless of the configuration, the 7500B provides no voice functions and is not used with voice terminals.

- ***510D or 515 BCT:*** Each 510D or 515 BCT requires one port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law) for shared use of voice and data.

- ***7400D Series or CALLMASTER Terminals:*** Each Voice Terminal requires one port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law) 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 DSU for connection to associated data terminals. In addition, a 7400B Data Module may be used to provide all 7400D series voice terminals a connection to data terminals and a common DCP interface to the switch.

- ***7500 Series ISDN Voice Terminals:*** Each 7500 Series ISDN Voice Terminal requires one port on the TN556 BRI port circuit pack. Each voice terminal requires an optional ISDN ADM to support asynchronous DTE. Consisting of a board located inside the BRI voice terminal, the ISDN ADM allows the transmission of integrated voice and data through one voice terminal. The ISDN ADM shares the port with its associated voice terminal and supports the Hayes command set for compatibility with PC communications packages.
- ***Modems:*** Each modem requires one port on a TN742 or TN746B (A-law) 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:*** DEFINITY Generic 1 and Generic 3i require 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. A 7400A Data Module may be used in place of the TDM or MTDM.
- ***Data Line Data Module:*** Each port is connected to an ADU that converts to an RS-232 interface for connection to a Data Module.

Keyboard Dialing to off-premises data endpoints requires the use of a TN748C Tone Detector circuit pack (TN420C, TN744 support A-law). 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

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 DCE, DTE connected to the system by a data module, or other devices such as DTDM, PDM, 515 BCT, or a port of a 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 Call Hot Line feature instead.

- Data Terminal (Keyboard) Dialing

  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 ISN data endpoint from a system digital data endpoint requires a two-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 — The preadministered Hot Line Destination must be stored in an abbreviated dialing list. This list can be a system list, group list, personal list, or enhanced list. Therefore, before this feature can be activated, users' terminals need to be administered with an abbreviated dialing list that stores the Hot Line Destination. Then, each data module needs to specify the index number of the entry which stores the destination. The following fields from the data module form are needed to activate the Data Hotline feature:
  - Special Dialing Option (default/hotline) — For Data Hotline, ''hotline'' must be selected. Default Dialing and the Data Hotline feature cannot both be assigned to an extension.
  - An Abbreviated Dialing List — One entry in the data module's abbreviated dialing list needs to store the default destination.
  - Dial Code (0-999) — An index to the entry of the abbreviated dialing list that stores the default destination.

Hardware and Software Requirements

No additional hardware or software is required.

Data-Only Off-Premises Extensions

Description

Allows users to establish data calls involving DCE or DTE that is located remotely from the System site using DATAPHONE digital service or other private line data facilities. A Data-Only Off-Premises Extension uses an MTDM located on-premises. Communication with the remote data equipment is accomplished through the private line facility linking the on-premises MTDM 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

The 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 Module 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 Circuit 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 Digital Line circuit pack (TN413, TN754B support A-law). 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 from 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.

- 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 on analog voice terminals. However, Priority Calls appear on the next available line appearance on multi-appearance voice terminals.

- 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.

- 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 on analog voice terminals. However, Priority Calls appear on the next available line appearance on multi-appearance voice terminals.

- 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

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 DCS nodes.

Calls to and from a DEFINITY Generic 1 or Generic 3i in a DCS network have Calling/Called Name Display transparency under the following conditions:

- The other party is at another DEFINITY Generic 1 or Generic 3i and the tandem node is a System 75 Version 3 or later, DEFINITY Generic 1, DEFINITY Generic 3i, 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 DEFINITY Generic 1 systems, DEFINITY Generic 3i systems, and System 75s.

Within the same DEFINITY Generic 1 or Generic 3i 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 DEFINITY Generic 1s or Generic 3is with a System 75 Version 3 or later, DEFINITY Generic 2, System 85 Release 2 Version 2 or later, or a DEFINITY Generic 2.1 as an intermediate node
- A DEFINITY Generic 1 connected to a System 85 Release 2 Version 2 or later, or a DEFINITY Generic 2.1

Configurations in which 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 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 AT&T *DEFINITY® Communications System Generic 1 — Voice Terminal 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.

- CAS

  When a user dials the extension for CAS, a 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.

- DDC/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

DCS interface hardware and DCS software are required.

DCS Attendant Control of Trunk Group Access

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. 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 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 four 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 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 three-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 three-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.

- UDP

  DCS tie trunks should not be attendant controlled. This would result in all UDP 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

DCS interface hardware and DCS software are required.

DCS Attendant Direct Trunk Group Selection

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 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.

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

DCS interface hardware and DCS software are required.

DCS Attendant Display

Description

Provides some transparency with respect to the display of call-related information.

Calls to and from a DEFINITY Generic 1 in a DCS environment have Calling/Called Party Identification transparency under the following conditions:

- The other party is at another DEFINITY Generic 1 or System 75 and the intermediate node is a DEFINITY Generic 1, System 75 Version 3 or later, 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.

DEFINITY Generic 1 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 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 DEFINITY Generic 1s or System 75s.

DCS tie trunks between nodes must be administered with the Outgoing Display enabled. This enables the called party's name to be displayed at the calling attendant's display.

Interactions

When both 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. This information is given elsewhere in this chapter.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Automatic Callback

Description

Allows a user at one node to make an automatic callback call to a user at another node in the 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 DEFINITY Generic 1 or System 75 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 six 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 DEFINITY Generic 1 or System 75 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.

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 six 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 DEFINITY Generic 1 or System 75 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

DCS interface and DCS software are required.

DCS Automatic Circuit Assurance (ACA)

Description

Allows a voice terminal user or attendant at a G1 or G3i node to activate or deactivate ACA referral calls for the entire 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 node for one or more remote nodes, the remote nodes are notified when ACA referral is activated or deactivated.

If referral calls are generated at a remote node for a DEFINITY Generic 1 or Generic 3i node, the DEFINITY Generic 1 or Generic 3i 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 DEFINITY Generic 1 or Generic 3i 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 DEFINITY Generic 1 or Generic 3i referral calls.

A detailed description of the ACA 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 Generic 1 or Generic 3i node for the Generic 1 or Generic 3i node.
- If administered as remote, referral calls are generated at a remote node for the Generic 1 or Generic 3i node. In this case, the remote node identification must also be entered.
- If administered as primary, referral calls are made at the Generic 1 or Generic 3i node for a remote node.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Busy Verification of Terminals and Trunks

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 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 UDP 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 fully described elsewhere in this chapter.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Call Forwarding All Calls

Description

Allows all calls to an extension number to be forwarded to a selected extension number within the 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, LWC 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, which is fully described elsewhere in this chapter.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Call Waiting

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 DEFINITY Generic 1 or Generic 3i feature)
- Call Waiting — Termination
- Priority Calling

Attendant Call Waiting, Call Waiting Termination, and Priority Calling function the same between DEFINITY Generic 1, Generic 3i, or System 75 nodes in a DCS as they do from a single DEFINITY Generic 1, Generic 3i, or System 75. These features are fully described elsewhere in this chapter.

Call Waiting — Origination is not a feature of Generic 1 or Generic 3i, but is supported in a DCS for calls into Generic 1 or Generic 3i from nodes that do provide it. When activated before a call, this feature rings an idle single-line terminal with three-burst priority ringing. If the called party is busy, the call waits and the busy party hears three-burst call waiting tone through the handset. The waiting party hears Call Waiting ringback tone. (The number of bursts is administrable.)

Call Waiting — Origination can also be activated *after* the caller receives busy tone. After activation, the call waits, the busy party hears three-burst call waiting tone through the handset, and busy tone changes to Call Waiting ringback tone. (The number of bursts is administrable.)

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 Generic 1 or Generic 3i not 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

DCS interface hardware and DCS software are required.

DCS Distinctive Ringing

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 in a DCS environment as it does within a System.

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 two-burst ringing instead of the usual one-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 one-burst ringing. Calls from external tie trunk groups are treated as externally originated calls and receive two-burst ringing.

Administration

None required.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Leave Word Calling

Description

Enables terminal users to leave preprogrammed ‘‘call me’’ messages at other terminals within the DCS network. Messages can be left by calling, called, or covering users.

LWC transparency in a DCS configuration allows messages from a Generic 1 or Generic 3i 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.

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 DEFINITY Generic 1, Generic 3i, or System 75 to System 85 Release 2 Version 2 (or later) or DEFINITY Generic 2.1
- From DEFINITY Generic 1, Generic 3i, or System 75 through any intermediate node to another G1, G3i, System 75, System 85 Release 2 Version 2 (or later) or DEFINITY Generic 2.1
- To DEFINITY Generic 1 or Generic 3i from any other node

Retrieval of LWC messages is permitted only from a terminal at the node where the messages are stored.

DCS LWC cannot be activated from an attendant console.

Interactions

The following features interact with the DCS LWC feature:

- DCS Multi-appearance Conference/Transfer

  Activation of LWC is denied after a DCS call has been conferenced or transferred.

- DCS Call Forwarding All Calls

  If the forwarding extension and the designated extension are at different nodes, LWC 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 fully described elsewhere in this chapter.

Hardware and Software Requirements

DCS interface hardware and DCS software are required.

DCS Multi-Appearance Conference/Transfer

Description

Provides transparency of conference calls and the transfer of calls within a 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 UDP extension number.

A detailed description of the Conference — Attendant, Conference — Terminal, and Transfer features is 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 features interact 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

DCS interface hardware and DCS software are required.

DCS Over ISDN-PRI D-Channel

Description

Enhances the DCS feature by allowing access to the public network for DCS feature connectivity between DCS switch nodes. With this feature, DCS features are no longer restricted to private facilities. The ISDN-PRI B-Channel is used for voice communications, and the ISDN-PRI D-Channel is used to transport DCS control information. The only difference between DCS networks that do not utilize the DCS Over ISDN-PRI feature and those that do is in the method of signaling. The DCS Over ISDN-PRI D-Channel feature uses Temporary Signaling Connections (TSCs) to transport certain DCS control information.

A TSC provides a temporary signaling path through ISDN switches for exchanging user-user information. There are two types of temporary signaling connections: Call Associated Temporary Signaling Connections (CA-TSC) and Non-Call Associated Temporary Signaling Connections (NCA-TSC).

A CA-TSC refers to a service for exchanging USER INFORMATION messages that are associated with an ISDN B-Channel connection by the call reference value of the call control data packets.

An NCA-TSC is a connection not related with any ISDN B-channel connections. It is an administered virtual connection established for exchanging USER INFORMATION messages on the ISDN D-Channel for the DCS over ISDN-PRI D-Channel application as well as for the DCS AUDIX application. Once an NCA-TSC has been administered and enabled, it will be active for an extended period of time. There are two types of administered NCA-TSCs depending on their setup mechanism: *Permanent* and *As-needed*.

Once enabled, a permanent NCA-TSC will remain established while the system is running. If the permanent NCA-TSC is dropped for any reason, the system will attempt to reestablish the connection. An as-needed administered NCA-TSC is established based on user request and the availability of TSC facilities. The connection will be dropped after an administered period of inactivity.

The system can transport DCS or DCS Audix messages over an ISDN-PRI D-Channel and over BX.25 data links when functioning as a gateway between a PBX equipped with the DCS Over ISDN-PRI D-Channel feature and a PBX equipped with traditional DCS using BX.25 data links. In this situation, the messages travel from the gateway through the NCA-TSCs or CA-TSCs to TSC-capable PBXs and from the gateway to PBXs that support only traditional DCS via a BX.25 logical channel.

At least one PBX must be configured as an ISDN DCS Gateway node in a DCS network that consists of PBXs that support DCS Over ISDN-PRI D-Channel and PBXs that do not support the feature.

For examples of various DCS network configurations and how they are administered, see *DEFINITY® Communications System Generic 3i — Implementation*, 555-230-650.

Considerations

A DCS cluster of TSC-compatible systems can consist of up to 64 nodes (X.25 and ISDN-PRI).

The maximum number of administered NCA-TSCs per signaling group is 64.

The maximum number of administered NCA-TSCs per system is 256.

The maximum number of TSC gateway channels per system is 64.

The maximum number of TSCs (CA-TSCs and NCA-TSCs) per signaling group is 256 (network limitation).

The maximum number of CA-TSCs per system is 400.

The maximum number of TSCs (CA-TSCs and NCA-TSCs) per system is 656 (400 CA-TSCs and 256 NCA-TSCs).

The maximum number of signaling groups per system is 8.

The maximum number of processor channels per system is 64.

System users should notice no difference between DCS features over ISDN-PRI and traditional DCS features.

Interactions

Most feature interactions with DCS Over ISDN-PRI D-Channel are the same as those with traditional DCS features. However, some interactions are unique to the DCS Over ISDN-PRI D-Channel feature. These unique interactions are listed below:

- Attendant DXS with Busy Lamp Field

  An attempt by the attendant to directly select an extension that has been previously administered as belonging to a administered NCA-TSC will result in intercept tone being received.

- D-Channel Backup

  In the event of a D-channel switchover (primary to secondary or secondary back to primary) in a private network, administered NCA-TSCs that were active are assumed to have remained active. Any unacknowledged user-user service requests are assumed to be rejected, and administered NCA-TSCs which were in the process of being established at the time of the switchover are dropped when the switchover occurs. Those administered NCA-TSCs that were dropped will be reattempted again.

If a D-channel switchover occurs on a D-channel going to the public network then all TSCs will be dropped. A maintenance-provided ‘‘heartbeat’’ message will periodically be sent over each permanent administered NCA-TSC to ensure that such a situation is detected and recovered from.

- Distributed Communications System AUDIX (DCS AUDIX)

  The DCS over ISDN-PRI D-Channel feature can be used to support DCS Audix. (The connection between G1 or G3i and AUDIX should be BX.25.)

- GRS

  GRS will select TSC compatible facilities when routing NCA-TSCs. In other words, a NCA-TSC request can only select a routing preference that supports TSCs.

  In a tandem node, GRS will first select facilities that support TSCs if the call falls into any one of the following two conditions:

  — It requests a CA-TSC explicitly

  — It contains a DCS information element in the SETUP message

  Once a trunk group with available members is selected, the call will proceed even if all the TSCs belonging to the associated signaling group are active. In other words, the completion of a call is given priority over DCS transparency.

- ISDN-PRI

  This feature uses ISDN-PRI call control protocol and messages.

- SDN

  The DCS over ISDN-PRI D-Channel feature allows the system to access public networks such as SDN. SDN supports all DCS features except for the following:

  — DCS Attendant Control of Trunk Group Access

  — DCS Attendant Direct Trunk Group Selection

  — DCS Busy Verification of Terminals and Trunks

- CDR

  CDR will record both the status and the utilization of TSCs. Both CA-TSCs and NCA-TSCs can be recorded. For more information, consult the CDR description in this manual or the CDR manual.

- Voice Terminals

  An attempt to dial an extension that has been previously administered as belonging to a administered NCA-TSC will result in intercept tone being received.

Administration

The ISDN-PRI option must be enabled on the System-Parameters Customer-Options form before associated forms and fields on forms can be administered.

To implement this feature, the following form(s) or sections of the form(s) must be completed.

- Signaling Group Form — Complete the following fields:
  - `Max number of NCA TSC` — Specifies the maximum number of NCA-TSCs that will be supported in the Signaling Group. Allowable entries are 0 through 256. Default is 0.
  - `Max number of CA TSC` — Specifies the maximum number of CA-TSCs that will be supported in the Signaling Group. Allowable entries are 0 through 400. Default is 0.
  - `Trunk Group for NCA TSC` — Specifies the ISDN-PRI trunk group whose Incoming Call Handling Table (ICHT) will be used to handle incoming NCA TSCs through this signaling group.

    Complete Pages 2 through 5 of the form as required to assign administered NCA-TSCs. A maximum of 64 administered NCA-TSCs may be administered as part of the Signaling Group form.
  - `Service/Feature` — Specifies the service type for all the Administered NCA-TSCs in this signaling group. Allowable entries are `accunet`, `i800`, `inwats`, `lds`, `mega800`, `megacom`, `multiquest`, `operator`, `sdn`, `sub-operator`, `wats-max-bnd`. Default is blank.
  - As-needed Inactivity Time-out(min) — Specifies the inactivity time-out for as-needed NCA TSCs assigned in this signaling group. When the timer expires, an as-needed administered TSC will be torn down. Enter a number between 10 and 90. Default is blank.
- ISDN TSC Gateway Channel Assignment Form — Used to map administered NCA-TSCs to (BX.25) processor channels in an Integrated DCS Network. (Form only required for PBX location serving as an ISDN Gateway in an Integrated DCS Network.) Complete all fields as required.
- ISDN Trunk Group Form — Complete the following fields:
  - `Used for DCS?` — Appears for `access`, `rlt`, `tandem`, `tie`, and `isdn-pri` trunk group types. This field determines if a trunk will send and receive messages via a DCS signaling link. (Note: This field does not appear for an ‘‘isdn-pri’’ trunk group with a service type of `dmi-bos` or `ivbs`. For all other `isdn-pri` trunk group types, if `y` is entered in this field, then the dependent fields `PBX ID` and `DCS Signaling` appear.
  - `PBX ID` (dependent field) — Only displayed when the `Used for DCS?` field on the ISDN PRI trunk group form is `y`. Identifies the remote PBX within the network with which the trunk will

communicate on a DCS signaling link. Allowable entries are one through 63. Default is 1.

— `DCS Signaling` (dependent field) — Only displayed when the `Used For DCS`? field on the ISDN PRI trunk group form is `y`. The option `bx.25` is used to support the traditional DCS feature. The option `d-chan` is used to support the DCS Over ISDN-PRI D-Channel feature. Enter `d-chan`.

— `NCA-TSC` Signaling Group — Specifies on which signaling group NCA-TSC calls routed to this trunk group should be established. Allowable entries are 1 through 8 or blank. Number entered must match at least one of the entries in the Sig Grp field on the Trunk Group Member Assignments portion of the Trunk group form. Default is blank.

NOTE:
If there is no ambiguity in the Sig Grp field (that is, only one number has been entered for one or more group members), then, if this field is left blank, the system will fill in the field based on the information that was entered in the `Trunk Brd` field on Page 1 of the Signaling Group form. If there is ambiguity, then you must fill in this field.

- Routing Pattern Form — Complete the following fields:

  — TSC — Allowable entries are `y` or `n`. When a `y` is entered in this field, the `CA-TSC` field becomes writable. (See ‘‘CA-TSC Request’’ below.) Default is `n`. For an NCA-TSC, the `n` option means that the NCA-TSC will be blocked if it is routed through this pattern in a tandem switch.

  — CA-TSC Request — This field becomes writable when a `y` is entered in the `TSC` field. (See ‘‘TSC’’ above.) You must specify when a CA-TSC will be established if the call is routed through a particular preference of a routing pattern. Allowable entries are then `as-needed`, `at-setup`, and `none`. The first two entries, ‘‘as-needed’’ and ‘‘at-setup’’ are meaningful only for CA-TSCs. If the ‘‘as-needed’’ option is entered, a CA-TSC will be established only when it is requested during a call. If the ‘‘at-setup’’ option is requested, then the CA-TSC will be established when the call is set up. Selection of the ‘‘none’’ option means that there will be no CA-TSC associated with this call. Default is ‘‘none’’.

- Communications-Links Processor Channels Form — The `Appl` field specifies to which application channel this processor channel is connected. Enter `gateway` in the associated `Appl` field if the processor channel is used as one end of the gateway channel assigned in the ISDN TSC Gateway Channel Assignments form. The gateway node serves as the terminating node to the D-Channel DCS network as well as the terminating node to the traditional DCS network.

A PBX serving as an ISDN DCS Gateway node introduces some interesting situations when administering processor channels in an associated traditional DCS PBX. In a traditional DCS network, (bx.25 processor channel links) the `Remote Proc Chan` field in the Processor Channel Assignments form refers to the processor channel of the destination PBX. In an Integrated DCS network, the `Remote Proc Chan` field in the Processor Channel Assignments form refers to the processor channel of the Gateway PBX (if the destination PBX is an ISDN DCS PBX), **not** the destination PBX.

On the contrary, the `Machine-ID` field in the Processor Channel Assignments form refers to the destination PBX, either an ISDN DCS PBX or a traditional DCS PBX. The Gateway PBX number **must** not be used in this field if the destination PBX is an ISDN DCS PBX.

Hardware and Software Requirements

The DCS over ISDN-PRI D-Channel feature requires the same hardware as other ISDN-PRI features.

ISDN-PRI software is required.

DCS Trunk Group Busy/Warning Indication

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 three-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 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 1,500 (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 Generic 1 or Generic 3i 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.

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

DCS interface hardware and DCS software are required.

Default Dialing (G3i)

Description

Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination by either entering ***Return*** at the `DIAL:` prompt (for data terminals using DCP data modules) or typing ***d*** and entering ***Return*** at the `CMD:` prompt (for data terminals using ISDN-BRI data modules). The data terminal user with a DCP data module can still place calls to other destinations by entering the complete address after the `DIAL:` prompt (normal Data Terminal Dialing or Alphanumeric Dialing). The data terminal user with an ISDN-BRI data module can still place calls to other destinations by typing ***d***, entering a space, typing the complete address, and entering ***Return*** after the `CMD:` prompt.

Default Dialing provides data terminal users who dial a specific number the majority of the time a very simple method of dialing that number. Normal Data Terminal Dialing and Alphanumeric Dialing are unaffected.

For the AT command interface supported by the 7400A data module, the **ATD** command may be entered instead of a carriage return to dial the default destination.

The default destination is administered by the System Manager. Default Dialing can be assigned to any data endpoint that has the Data Terminal Dialing capability.

Considerations

Default Dialing provides data terminal users who dial a specific number the majority of the time a very simple method of dialing that number.

Interactions

The following features interact with the Alphanumeric Dialing feature:

- Data Call Setup

  Default Dialing enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination by either ***Return*** at the `DIAL:` prompt (for DCP) or typing ***d*** and entering ***Return*** at the `CMD:` prompt (for ISDN-BRI).

Administration

Alphanumeric Dialing is administered by the System Manager. In addition to those items listed in the Data Call Setup feature description, elsewhere in this chapter, the following items must be administered:

- The preadministered default destination must be stored in an abbreviated dialing list. This list can be a system list, group list, personal list, or enhanced list. Therefore, before this feature can be activated, users' terminals need to be administered with an abbreviated dialing list that stores the default destination Then, each data module needs to specify the index number of the entry which stores the default destination. The following fields from the data module form are needed to activate the Default Dialing feature.
  - Special Dialing Option (default/hotline) — For Default Dialing, ''default'' must be selected. Default Dialing and the Data Hotline feature cannot both be assigned to an extension.
  - An Abbreviated Dialing List — One entry in the data module's abbreviated dialing list needs to store the default destination.
  - Dial Code (0-999) — An index to the entry of the abbreviated dialing list that stores the default destination.

Hardware and Software Requirements

No additional hardware or software is required.

Dial Access to Attendant

Description

Allows voice terminal users to access an attendant by dialing ***0*** (typically, in the U.S. Other countries may use different digits to reach the attendant). 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*** or an administered extension number with G3 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 COR) prohibits placing any calls, including Dial Access to Attendant calls.

Administration

The administrator may select an extension number other than zero for this feature. However, consideration should be given to the fact that prior to the emergence of 911 systems, the dialing of ***0*** was customary in the United States for emergencies and other urgent assistance requests. Using an extension number other than ***0*** for dial access to the attendant could create confusion and delay response to an emergency.

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:

- Extension Numbers

  Flexible numbering allows one, two, three, four, or five-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 three-digit extension number is administered and the first digit is a 4, the extension numbers can range from 400 to 499. Also, if a four-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 may be by the single digit ''0.'' It is also possible to setthe dial plan so that some other digit is used to reach the attendant (for example, 9 in Italy). In addition, Generic 1 and Generic 3i provide for Individual Attendant Access by assigning each attendant an individual extension number.

- Trunk Access Codes

  A minimum of one digit and a maximum of three (G3i-U.S) digits or four (G3i-Global) 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 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 part 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, ***3 2*** could be used to activate Call Forwarding All Calls and ***3 3*** used to deactivate Call Forwarding All Calls.

A UDP 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 UDP is to be established, all extension numbers must be the same length (four to five digits). A UDP also requires the following information, so that calls will route to the desired switch:

- A PBX Code, which represents the first one to four digits of a four- or five-digit extension number and can range from one 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 AAR pattern for the call — this information is required for each PBX code. With G1.1, this RNX can be of the form NNX. With G3i, the RNX is flexible (no fixed form).
- 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 DCS.
- A Local PBX ID (G3i only), which represents the PBX ID of the local switch.

Considerations

The entire Dial Plan is dependent on the first and second 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 UDP is to be established
- The type of code and the number of digits in the code for each first digit.
- Optionally, a second digit table.

If a UDP 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

Description

Supports two signaling techniques: Bit Oriented Signaling and Message Oriented Signaling for direct connection to host computers.

Message Oriented Signaling is used with 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 Channel
- Mode 1 — 56 kbps Channel
- Mode 2 — 0-19.2 kbps Synchronous/Asynchronous
- Mode 3 — Multiple Virtual Channels

ISDN-PRI can be assigned as the signaling mode. In this case, the TN767 DS1 circuit pack (TN464B/C/D support A-law) 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 PDM since the DMI protocol is identical to the 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.

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 (TN464B/C/D support A-law).
- DMI Trunk Group — Associate the trunks to the groups.

Hardware and Software Requirements

One TN722B or TN767 DS1 circuit pack (TN464B/C/D support A-law) is required for every 23 DMI trunks required. If ISDN-PRI signaling is used, a TN765 Processor Interface circuit pack is also required.

No additional software is required.

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

Description

Allows direct inward access to an answering group other than the attendant even if the system does not have the DID feature.

A DDC or UCD answering group can consist of voice terminals and individual attendants (described in the Individual Attendant Access feature elsewhere in this document). 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, 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, G1.1, or G3i) 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, Call Forwarding All Calls, or the Hunt Group Busy Function discussed later), 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 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 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 can have an assigned status lamp that identifies an incoming DDC or UCD call. However, the voice terminal or individual attendant must be idle (not active on any call appearance) before a group call will be directed to the terminal or console. 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 (one to 99 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 then 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 one to 200 calls if queuing is provided. If queuing is not provided, the queue length must be set to zero. 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 (discussed 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 one to 200; however, it cannot exceed the queue length.

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 five 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 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 99 groups with up to 200 members per group can be provided. The system maximum, however, is 500 group members.

Each system can contain up to 64 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. 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 dials the Hunt Group Busy activation code followed by the DDC or UCD group number, or presses the Auxiliary Work button, the terminal or console 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 Auxiliary 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 DID, tie, or DS1 tie trunk. A Caller to a busy hunt group hears ringing if the caller is incoming on a 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, the status lamp associated with the Auxiliary 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.)

LWC 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 system-wide 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 individual UCD member is assigned a 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 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.

Agents should not be used for hunt group calls and ACD split calls simultaneously.

Oldest call waiting termination is only supported on ACD calls, not on hunt group calls.

Interactions

The following features interact with the DDC and UCD 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 Auxiliary 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 Auxiliary 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.

- 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.

- 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

  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

  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 three-burst ringing is heard.

- CDR

  The system can be administered to record the called number on incoming calls as the particular hunt group extension number or hunt group member extension number.

- 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 (not applicable for vector-controlled hunt groups [G3i])
- Delay announcement interval (not applicable for vector-controlled hunt groups [G3i])
- Group extension number, name, and type (DDC and UCD)
- Coverage Path (not applicable for vector-controlled hunt groups [G3i])
- COR
- Four-digit security code
- Whether or not the group is served by a queue
- Queue length (one to 200 calls)
- Queue Warning Threshold (one to 200 calls)
- Port Number assigned to queue warning level lamp
- Group Members (extension numbers)

Also, the system can be administered to record (via CDR) 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, TN746B (A-law), or TN769 Analog Line circuit pack. A 21C-49 indicator lamp may be used as a queue warning level lamp. This lamp is approximately two 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 or TN746B (A-law) 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 (TN763D supports A-law) is required. Announcement equipment and music sources are not provided by the system.

No additional software is required. However, with G3i, if a hunt group is to be vector controlled, Call Vectoring software is required.

Direct Inward Dialing (DID)

Description

Connects calls from the public network directly to the dialed extension number without attendant assistance.

DID reduces the attendant's work load and provides the calling party immediate contact with the called party.

Considerations

DID trunk group(s) from the local telephone company CO is required.

Interactions

The Inward Restriction, Manual Terminating Line Restriction, Public Restriction, and Termination Restriction features (administered by the COR) prevent receiving DID calls at the restricted voice terminal. DID calls that reach a restricted voice terminal or that are transferred to a restricted voice terminal can be sent either to an attendant or recorded announcement.

When a DID trunk is accessed via a 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.

Incoming DID calls that reach a busy destination can be either given busy tone or routed to an attendant.

Administration

DID is administered by the System Manager. The only administration required is the administration of a DID trunk. The DID number assigned in the CO should match the extension numbers of associated system extensions. An unanswered DID calling timeout interval can be administered to have special treatment applied; use the system parameters miscellaneous form.

If MFC is to be used by DID, additional administration is required to set the country code on the country options form.

Hardware and Software Requirements

Requires one port on a TN753 DID Trunk circuit pack for each DID trunk. No additional software is required. If MFC signalling is to be used for DID, a TN744B call classifier is also required.

Direct Inward and Outward Dialing (DIOD) — International (Japan)

Description

Provides a two-way service, with both inward and outward dialing features, allowing calls from a Japanese public exchange to be made directly to a subscriber on a G1.2SE. The Dialing Inward and Outward Dialing (DIOD) feature is a combination of the DID feature and the Direct Outward Dialing (DOD) feature. The Japanese version of DID, however, implies a two-way service, with both inward and outward dialing features.

Considerations

The TN429 circuit pack is an 8-port loop start trunk circuit pack required for DIOD. The TN429 has the capability of being used in any of the following ways:

- DID trunk
- CO trunk — two-way, one-way incoming, or one-way outgoing CO
- DIOD trunk with DID service and one-way outgoing CO trunk capability.

The TN429 supports several administerable timers and different trunk group types (DID, CO, DIOD, FX, and WATS). When the TN429 is administered, only loop-start is allowed to be entered as its trunk type.

Interactions

When the DIOD trunk is being used as a DID trunk, the following interactions apply:

- The Inward Restriction, Manual Terminating Line Restriction, and Termination Restriction features (administered by the COR) prevent receiving DID calls at the restricted voice terminal.
- When a DID trunk is accessed via a 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.

When the DIOD trunk is being used as a DOD trunk, the calling party restrictions (assigned by the COR) prevent placing DOD calls from the restricted voice terminal.

Administration

The System Manager must administer a DIOD trunk group form.

Hardware and Software Requirements

Requires the TN429 DIOD trunk circuit pack. No additional software is required.

Direct Outward Dialing (DOD)

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

DOD 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 CO, a WATS serving office, or a 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 COR) prevent placing DOD calls from the restricted voice terminal.

Administration

DOD 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. If MFC is to be used by DOD, additional administration is required to set the country code on the country options form.

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. Remember that the ringing cycles are administrable.

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	one-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	two-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	three-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	three-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

Distinctive 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 two- and three-burst ringing is optional only on single-line voice terminals. If Distinctive Ringing is disabled, the user will hear a one-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 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 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 two- or three-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 or TN746B (A-law) Analog Line circuit pack. No additional software is required.

Do Not Disturb

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, 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 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 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 work load 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 10 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 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 1,600.

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.

- 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

Description

Provides a digital interface for the following:

- Voice-Grade DS1 Tie Trunks
- AVD DS1 Tie Trunks
- Robbed-Bit Alternate Voice/Data (RBAVD) DS1 Tie Trunks
- DMI Tie Trunks
- ISDN-PRI Trunks
- CO Trunks
- FX Trunks
- Remote Access Trunks
- WATS Trunks
- DID Trunks
- Off-Premises Stations
- Access Endpoints (G3i)

Voice Grade DS1 Tie Trunks

The Voice-Grade DS1 tie trunks are an alternative to four-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:

- ETN or TTTN tie trunks
- Main/Satellite tie trunks
- Tie trunks used to interface with EPSCS and CCSA networks
- Release link trunks for CAS
- Access Trunks.

The TN722B, TN767, or TN464C/D 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 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.

AVD Tie Trunks

AVD DS1 tie trunks permit alternate voice and data calling between a DEFINITY Generic 1 or Generic 3i and a System 75, 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 (TN464B/C/D support A-law) 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.

Tie Trunks

Robbed-Bit Alternate Voice/Data DS1 tie trunks permit alternate voice and data calling between a Generic 1 or Generic 3i and a System 75, System 85, or DEFINITY Generic 2.1.

Robbed-Bit Alternate Voice/Data DS1 tie trunks can be used to connect the system with other digital switches.

Robbed-Bit Alternate Voice/Data DS1 tie trunks can be used to dynamically access voice/data SDN Services.

For normal AVD facilities, modem pool resources are not automatically inserted. With Robbed-Bit Alternate Voice/Data facilities, a data origination access code can be used to force modem pool insertion on the call.

The TN722B or TN767 DS1 circuit pack (TN464B/C/D support A-law) is used to support Robbed-Bit Alternate Voice/Data 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

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 (TN464B/C/D support A-law) 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 section.

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

When the TN767 DS1 circuit pack interface (TN464B/C/D support A-law) 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

Signaling for remote access trunks is depends on the trunk group and incoming/outgoing dial types.

DID Trunks

When the DS1 interface is used to provide DID trunk group service, Robbed Bit Signaling must be used.

Off-Premises Stations

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.

Access Endpoints (G3i)

Individual channels of a DS1 can be assigned extensions with the Access Endpoint function. Access Endpoints are discussed in the Administered Connections feature description elsewhere in this manual.

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 transmission 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. A TN464C can support up to 32 trunks.

The system can support a maximum of 30 DS1 circuit packs.

Interactions

The following features interact with the DS1 Trunk Service feature:

- CAS

  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.

- DCS

  AVD DS1 tie trunks can only be used in a DCS between two DEFINITY Generic 1s or between Generic 1 or Generic 3i and System 75, System 85, or DEFINITY Generic 2.1.

- ETN

  AVD DS1 tie trunks can only be used in an ETN between two Generic 1s or between Generic 1 or Generic 3i and System 75, 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 EPSCS or 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 DS1 circuit pack (TN464B/C/D support A-law) 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. A TN464C circuit pack is required to support 32 trunks.

No additional software is required.

E1 Trunk Service

Description

Provides the same service as DS1 Trunk Service at 2.048 mbps rate and with 32 channels. See the DS1 Trunk Service feature.

End-to-end signalling

End-to-end signalling allows a rotary or DTMF station to sent DTMF digits over a trunk. This allows a rotary station user to access equipment such as AUDIX that is controlled by DTMF digits. The trunk can be either rotary or tone signalling type. For a rotary signalling type, addressing digits are sent over the trunk as rotary. After addressing is complete and the call is connected, any additional digits will be sent as DTMF. See the Last Number Dialed or Abbreviated Dialing features for examples of using end-to-end signalling. is a switch feature that converts a station's rotary or touch tone dialing digits into touch tone digits that can then be sent over a

Interactions

None.

Considerations

Flashhooks from rotary (500 type) station sets must be longer than 120ms to be distinguished from the rotary digit "one" because the rotary digit "one" is converted and sent as a DTMF "one."

Administration

Hardware and Software Requirements

EIA Interface

Description

Provides an alternative for host connections and analog voice terminal users who use simple data terminals or Personal Computers (PCs) which emulate simple data terminals, and do not use PCs with PC communications packages, within the system hardware, for interconnection between EIA-232 compatible DTE and the system. The EIA Interface consists of a Data Line circuit pack port and an ADU.

The EIA Interface supports speeds of LOW, 300, 1200, 2400, 9600, and 19200 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, and so on, via a menu-driven selection mode at the DTE. Also, the Manager I terminal (G1) or the G3 Management Terminal 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 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.

DCE may also be connected to a data line port by use of a null modem.

Even though the EIA Interface is an excellent alternative for analog voice terminal users with simple data terminals, the 7400B data module is preferred when connecting digital voice terminals.

Considerations

The system's EIA Interface support offers a convenient and lower-cost alternative for host connections and for analog users with simple data terminals. DTEs can connect directly to a Data Line circuit pack which 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-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 two-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 two seconds or two breaks within one 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 Characteristics — 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 four 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

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 COS, 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:

- 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.

- COR

  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.

- Priority Queue

  The priority queue administration can change the priority of an Emergency Access to the Attendant call to an equal or lower priority than other types of calls.

- 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 the emergency queue overflow will redirect the call
- 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.

End-to-end Signalling

Description

End-to-end signalling is a switch feature that converts a station's rotary or touch-tone dialing digits into touchtone digits than can then be sent over a trunk to any equipment that requires a DTMF input. When sent on a rotary trunk, addressing digits are sent as rotary dial pulse. After the addressing digits are sent, any additional digits are converted to touchtone digits. See the Last Number Dialed or Abbreviated Dialing features for examples using end-to-end signalling.

Interactions

None.

Considerations

None.

Administration

None.

Hardware and Software Requirements

None.

Enhanced DCS (EDCS)

Description

Enhanced DCS is a new feature for G3i-Global. Enhanced DCS adds features to the existing DCS capabilities. Additional features include exchanging information to provide class of restriction checking between PBXs in the EDCS network, providing call progress information for the attendant, allowing attendant intrusion between a main and a satellite, and allowing a main PBX to provide DID/CO intercept treatment rather than the satellite PBX.

Considerations

If the DCS link fails, the administrator can choose to allow calls to continue without class of restriction checking or to block all DCS calls to inward-restricted stations.

Interactions

- Class of Restriction

  When a call goes to coverage, it is the called party's (not the covering party's) restrictions that will be used.

Administration

This feature is enabled with the system parameters miscellaneous form.

Hardware and Software Requirements

This feature requires the same hardware as DCS.

Facility and Non-Facility Associated Signaling (G3i)

Description

Provides signaling for ISDN-PRI.

Facility Associated Signaling

Facility Associated Signaling (FAS) allows an ISDN-PRI DS1 Interface D-Channel to carry signaling information for only those B-Channels located on the same DS1 facility (circuit pack) as the D-Channel.

Non-Facility Associated Signaling

Non-Facility Associated Signaling (NFAS) allows an ISDN-PRI DS1 Interface D-Channel (signaling channel) to convey signaling information for B-Channels (voice and data channels) on ISDN-PRI DS1 facilities other than the one containing the D-Channel. As a result, a D-Channel can carry signaling information for numerous B-Channels located on different DS1 facilities.

D-Channel Backup

To improve reliability in the event of a signaling link failure, a D-Channel Backup may be administered. If a signaling link failure does occur, a switch to a backup D-Channel can be implemented.

D-Channel Backup requires that one D-Channel be administered as the Primary D-Channel and that a second D-Channel be administered as the Secondary D-Channel. These assignments insure that at certain times during D-Channel Backup procedures that both D-Channels are in the same state. This avoids the occurrence of both switches at each end of the DS1 interface selecting the same D-Channel to be put into service. In these cases, the Primary D-channel is given precedence over the Secondary D-Channel.

The following figure shows a possible configuration involving three ISDN-PRIs between a G1.1 switch and the public network. Two of the ISDN-PRIs contain a D-Channel and 23 B-Channels, while the other ISDN-PRI contains 24 B-Channels. One of the D-Channels is the Primary D-Channel, and the other is the Secondary D-Channel. Together, this pair of D-Channels will signal for all 70 (23+24+23) of the B-Channels that are part of the three PRIs.

Since the D-Channels are signaling for more than one DS1 facility, the D-Channel Backup feature requires the use of the NFAS feature. At any given time, one of the two D-Channels will be carrying Layer 3 signaling messages, while the other D-Channel will be active at Layer 2, but in a standby mode only. Any Layer 3 messages received over the standby D-Channel will be ignored. Since only one of the D-Channels can be active at a time, load sharing between the two D-Channels is not possible. The two D-Channels can provide signaling for only a predefined set of B-Channels and cannot dynamically back up other D-Channels on other interfaces.

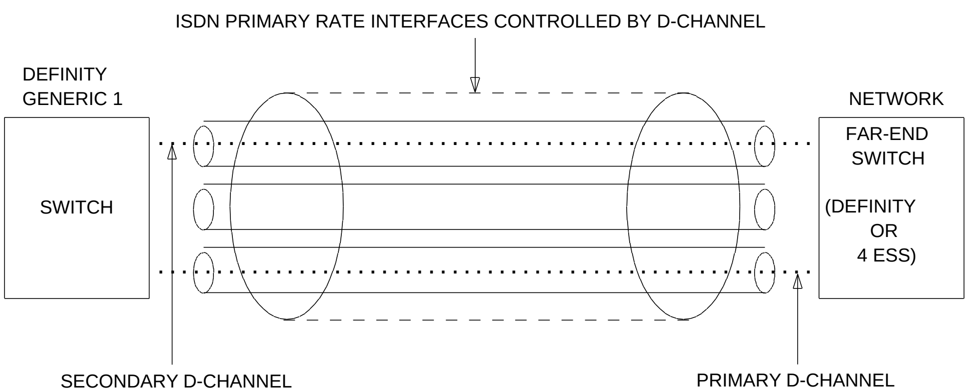


Figure 2-11. Example D-Channel Backup With Three ISDN-PRIs

D-Channel Backup Activation

D-Channel Backup can be invoked in response to the following events:

- D-Channel Failure

  If the signaling link fails on the active D-Channel (D1) or the hardware carrying D1 fails, then the system will send a message over the standby D-Channel (D2), which requests that D2 become the active D-Channel. D2 then becomes the active D-Channel and will carry all subsequent signaling messages. When the signaling link or hardware on D1 recovers from the failure, D1 becomes the standby D-Channel.

- System Technician Commands

  If a system technician command requests that a D-Channel switch-over take place, the first action taken by the system will be to tear down the signaling link on D1. After this has been completed, a message is sent on D2 to request that D2 become the active D-Channel. D2 then becomes the active D-Channel and the switch-over will be complete.

Only those ISDN-PRI facilities that use the NFAS feature will be capable of providing the D-Channel Backup feature. The reason for this limitation is that the two D-Channels must be located on different PRI DS1 facilities. As a result, the D-Channels must support NFAS so that they can signal for B-Channels on different ISDN-PRI DS1 facilities.

When a transition from one D-Channel to another occurs (D-Channel Backup is activated, all stable calls (calls that have been answered already) will be preserved. Some messages may be lost, resulting in a loss of call-related information, but the calls themselves will be maintained. The effect of the transition on unstable calls (those that have not been answered yet) is unpredictable since the results depend on which messages were lost and the contents of those messages.

Interactions

None.

Administration

The following provisioning and administration must be considered when implementing FAS and NFAS. Coordinate the following with the far-end switch for the DS1 facilities to be used:

- Decide which DS1 facilities will use FAS.
- Decide which of the remaining DS1 facilities will carry D-Channel signaling information on the 24th or 32nd channel. For those channels that have a D-Channel Backup, D-Channel pairs must be allocated.
- Define Signaling Groups (1 through 8). A Signaling Group is a group of B-Channels for which a given D-Channel (or D-Channel pair) will carry the signaling information. Each Signaling Group must be designated as either a FAS or NFAS Signaling Group.
  - A FAS Signaling Group must contain all the ISDN B-Channels on the DS1 interface associated with the group's D-Channel, and cannot contain B-Channels from any other DS1 circuit pack. Some of the DS1 ports may use in-band (robbed-bit) signaling and be members in a tie trunk group rather than an ISDN trunk group. These tie trunks cannot be members of a Signaling Group.
  - There is no restriction on which DS1 ports can belong to an NFAS Signaling Group. Normally, an NFAS Signaling Group consists of one or two D-Channels and several complete DS1 interfaces.

    If a Signaling Group contains only a subset of a DS1's B-Channels (ports 1 through 12, for example), it is considered an NFAS Signaling Group, not a FAS Signaling Group. The remaining B-Channels on the DS1 will then be assigned as members of another NFAS Signaling Group.
- An Interface ID must be assigned to each DS1 facility in an NFAS Signaling Group. For example, if the B-Channels in a Signaling Group span three DS1 facilities, a unique Interface ID must be assigned to each of the three DS1 facilities.
- D-Channel Backup involves two or more ISDN-PRI DS1 facilities that interconnect the switch to another PBX or to the network. Two D-Channels must be present on the facilities. One of the D-Channels is designated as Primary and the other as Secondary. This designation must be agreed upon by both sides of the interface and administered prior to initialization.

Hardware and Software Requirements

See the Integrated Services Digital Network — Primary Rate Interface feature description, elsewhere in this chapter, for hardware and software requirements.

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 (DDC or UCD 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.

See the System Capacities table in Chapter 4 for the number of Facility Busy Indication buttons that are allowed in the system. 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 five 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 (FRLs) and Traveling Class Marks (TCMs)

Description

Provides up to eight levels of restriction for users of the AAR and/or ARS features.

FRLs and 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) or by the administered dial string (G3i). 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. A blank entry 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 (G1.1). No specific order is required with G3i.

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 zero through seven. 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 (G1.1) or continues (G3i). With G1.1, 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 or first compatible trunk group (G3i). (See Ring-back Queuing, elsewhere in this section, 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 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 for extended calls. If Individual Attendant Access is assigned, the individual attendant's COR FRL is used. Data terminals use the FRL contained within the COR assigned to the associated data module.

The Remote Access feature can be accessed via a 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 RLTs, but including CCSA and EPSCS Access trunks
- WATS
- CO
- 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 CDR 15-digit account codes are used, the FRL field in the CDR record is overwritten.

Authorizations Codes can be used to raise a user’s FRL.

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 G1.1 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 zero through seven, 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 zero through five in one pattern and a range of two through seven 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 10 Barrier Codes, each with its own COR (and FRL) can be assigned. Although the COR defines other restrictions, 10 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 Private Network Access or ARS software is required.

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 CO trunks. DID trunks cannot be accessed.

  With G3i, a user's Class of Restriction must be administered with the Facility Access Trunk Test option in order for the the user to make trunk test calls.

- 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 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 Facility Test Calls access code must be assigned. Also, with G3i, a user's Class of Restriction must be administered with the Facility Access Trunk Test option in order for the the user to make trunk test calls.

Hardware and Software Requirements

No additional hardware or software is required.

Forced Entry of Account Codes

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 (G1.1)

  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), and directory assistance calls.

- Toll Calls Made By Users With a COR (G1.1)

  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 TAC Calls Made Over a Trunk Group With a Specific COR

  Any TAC call using a trunk group that is assigned a COR with forced entry of account codes cannot be made 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.

- Designated Calls (G3i)

  In G3i, each Dialed String entry in the toll analysis table can be administered to require forced entry of account codes. If the system is administered to require forced entry of account codes (on the Feature-Related System Parameters screen), and a specific number or Dialed String is administered to require forced entry of account codes, any system user must dial an account code before dialing that number.

  This includes all calls made by ARS or TAC.

- Designated Calls Made By Users With a Specific COR (G3i)

  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 that are administered to require forced entry of account codes.

For details on how account codes are used, see the CDR Account Code Dialing feature description, elsewhere in this manual.

Considerations

Forced Entry of Account Codes, by requiring account codes to be dialed on specific outgoing calls, provides an easy method of allocating the costs of specific calls to the correct project, department, and so on. Call information is recorded by the CDR feature for this purpose.

Account Code length can be up to 15 digits. This maximum decreases if an authorization code is dialed.

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:

- ARS

  If a trunk group is accessed via 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.

- 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 CDR Account Code access code and the 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.

- CDR

  CDR 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 CDR 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 individual COR requires account code entry
- Designated calls requiring Forced Entry of Account Codes (Toll Analysis form) (G3i)

Hardware and Software Requirements

No additional hardware is required. Optional CDR Account Code Dialing software is required.

Generalized Route Selection

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.

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. By providing additional parameters in the routing decision, GRS enhances AAR and ARS and 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. For data calls, GRS enables G3i to distinguish between restricted and unrestricted digital transmissions, allowing the switch to route data calls onto the appropriate facilities. GRS also provides the opportunity to integrate voice and data on the same trunk group, thereby providing certain economies.

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 five additional routing parameters which are specified on each trunk group preference of the routing pattern. These parameters are:

- Bearer Capability Class (BCC) — Identifies the type of call, such as voice calls and different type data calls.
- Information Transfer Capability (ITC) — Identifies the type of data transmission (restricted, unrestricted, or both).
- 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 five Bearer Capability Classes (BCCs). Customers may specify routing for each BCC according to their particular transmission needs.

Bearer Capability Classes (BCCs)

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.

The system will determine the originating endpoint's BCC from one of the following:

- For an ISDN-BRI set, G3i determines the BCC by using information from the Bearer Capability Information Element (IE) and Low Layer Compatibility IE of the ISDN SETUP message.
- For a non-BRI terminal, G3i creates a BCC by using information about the station administration for the terminal and information obtained by performing a terminal query (as shown in Table 2-31).
- From the administered value of the incoming trunk. For a non-ISDN trunk group, G3i creates a BCC by using the administered BCC value.
- From the ISDN-PRI bearer capability and low-layer compatibility parameters, if the call is an ISDN-PRI trunk-originated call.

The BCC associated with the routing preference in the routing pattern is administered by the system administrator. More than one BCC can be associated with each preference and the same facility can appear multiple times in a routing pattern and in multiple routing patterns.

The BCC of the originating endpoint (trunk or terminal) is matched with the BCCs of the routing preferences. An exact match is not always required. The system determines when conversion/insertion resources must be used to successfully complete a call via a compatible, but not identical, BCC.

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 2-31. 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 Processor Data Module-M1	1	1	For ACCUNET Switched 56 kbps Service
Modular Trunk Data Module	2	2	
Digital Terminal Data Module	2	2	
510D Personal Terminal	2	2	
Digital Communications Protocol Interface	0,2,3	2,3,4	See Note
7400A Data Module	2	2	
7400B Data Module	2	2	
3270T Data Module	3	3	
3270C Data Module	3	3	
3270A Data Module	2,3	2,3	See Note

NOTE:
For all endpoints, 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 for G1.1, 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

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 in the Bearer Capability Information Element (BC IE) and Low-Layer Information Element (LLC IE) have the following four values:

- Voice (speech)
- Voice-grade data (3.1 kHz transmission)
- Unrestricted digital transmission
- Restricted digital transmission.

With data calls, the switch distinguishes the information transfer capability (restricted or unrestricted) of the originating data endpoint (trunk or terminal), and then uses the information transfer capability of the data endpoint to route the call onto the appropriate facility. For BRI and PRI originating data endpoints, the information transfer capability is contained in the ISDN SETUP message. For non-ISDN data endpoints, G3i uses the information transfer capability specified by the system administrator. In G3i, the default of the information transfer capability of an endpoint is "restricted". The system administrator may change the information transfer capability to restricted or unrestricted for each non-ISDN originating endpoint.

More than one Information Transfer Capability can be supported by one BCC. (See Table 2-32.)

Table 2-32. 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 (async 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 a 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 optional and sent only in case of data calls.

NOTE:
DCP Mode 0 does not send an LLC IE.

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 a 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. This includes the ITC (restricted or unrestricted) if the call is a data call.

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, AVD, or RBAVD (robbed-bit AVD) trunk, then the BCC and ITC are 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 compatible 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.

After matching the BCCs, G3i will then match the ITCs. The originator’s ITC is matched to the route preference ITC. Unrestricted (unre) matches on "unr" or "both". Restricted (rest) matches on "rest" or "both".

NOTE:
ITC matching only applies to data calls (BCC 1-4).

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. The ITC does not help select a preference.

BCC 0 (voice) has no ITC. An ITC is chosen from the routing pattern. The following table shows how the ITC codepoint in the Bearer Capability IE is determined:

Originating Endpoint's ITC	**Routing Preference's Routing Preference's ITC**				**ITC Codepoint in BC IE**
	restricted	**unrestricted**	**both endpoint**	**both unrestricted**	
voice	x				restricted
voice		x			unrestricted
voice			x		unrestricted
voice				x	unrestricted

- 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. In G3i, the ITCs must also match.

When an ISDN-PRI trunk group preference is accessed, the BCC information is encoded and sent in the outgoing ISDN SETUP message to the distant-end 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 and ITC has been found, then that Bearer Capability is encoded and sent in the ISDN SETUP message to the far-end. If the call is a data call, the system uses the ITC of the routing pattern to encode the SETUP message as shown in the following table:

Originating Endpoint's ITC	Routing Preference's ITC				ITC Codepoint in BC IE
	restricted	**unrestricted**	**both endpoint**	**both unrestricted**	
restricted	x				restricted
restricted			x		restricted
restricted				x	unrestricted
unrestricted		x			unrestricted
unrestricted			x		unrestricted
unrestricted				x	unrestricted
voice*	x				restricted
voice		x			unrestricted
voice			x		unrestricted
voice				x	unrestricted

* A voice originated call without data preindication that is routed to a routing pattern with data preferences only.

- 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 and ITC determination on calls from endpoints to ISDN-PRI trunks:

Calls from Endpoints to ISDN-PRI Trunks

Originating

BCC	BCC 0				
BCC 0	P	PT	PT	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. Use the ITC as shown in the following table

PT Allow the call and send the BCC and ITC chosen from the routing pattern in the SETUP message

PM Insert a pooled modem for the call and send the BCC and ITC chosen from the routing pattern in the SETUP message

If BCC 1, 2, 3, or 4 is chosen from the preceding table, the following table is used to determine the appropriate ITC:

Calls from Endpoints to ISDN-PRI Trunks

Originating

ITC	unr			
unr	P	B	P	PU
rest	B	P	P	PU

B Block the call with intercept treatment

P Allow the call and send the originating endpoint's ITC in the SETUP message

PU Allow the call and send "unrestricted" in the SETUP message

2. BCC and ITC determination on calls from trunks to ISDN-PRI trunks:

Calls from Trunks to ISDN-PRI Trunks

Originating

BCC	BCC 0				
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. Use the ITC as shown in the following table

PT Allow the call and send the BCC and ITC chosen from the routing pattern in the SETUP message

If BCC 1, 2, 3, or 4 is chosen from the preceding table, the following table is used to determine the appropriate ITC:

Calls from Trunks to ISDN-PRI Trunks

	Originating			
ITC	unr			
unr	P	B	P	PU
rest	B	P	P	PU

B Block the call with intercept treatment

P Allow the call and send the incoming trunk's ITC in the SETUP message

PU Allow the call and send "unrestricted" in the SETUP message

The system does not insert pooled modem for any interworking trunk-to-ISDN-PRI trunk calls. The BCC and ITC of an incoming trunk is determined as follows:

ISDN-PRI Trunk BCC and ITC are in the received SETUP message

AVD Trunk BCC and ITC are the BCC and ITC values administered on the trunk group form

RBAVD Trunk BCC and ITC are the BCC and ITC values administered on the trunk group form

Data Trunk BCC and ITC are the BCC and ITC values administered on the trunk group form

Voice Trunk BCC is 0

3. BCC and ITC 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	BCC 0				
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. The ITC defaults to restricted in this case.

If BCC 1, 2, 3, or 4 is chosen from the preceding table, the following table is used to determine the appropriate ITC:

Calls from ISDN-PRI Trunks to BRI Endpoints (GRS is not involved)

	Originating	
ITC	unr	
unr	P	P
rest	P	P

P Allow the call and send the incoming trunk's ITC in the SETUP message. The data handshake procedures will establish or drop the call based on compatibility of the endpoints.

Considerations

ACCUNET and SDDN Services

The system will be able to use ARS tables to route calls to these networks. For example, BCC 1 is a 56 kbps service. If an ACCUNET 56 kbps Service trunk group is in a routing pattern that uses GRS, BCC 1 should be set to ‘‘yes’’.

SDI Service

The system will be able to support the AT&T Switched Digital International (SDI) Service, which requires 64 Kbps unrestricted service for Mode 0 calls or 64 Kbps restricted or unrestricted rate adapted to 56 Kbps for Mode 1 calls. SDI will reject calls encoded as 64 Kbps restricted without rate adaption.

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 if voice-grade data calls are assigned BCC 0. Voice-grade data calls could be assigned BCC 2.

Modem Pooling

A system originating a data call over a public or private network is responsible for inserting the pooled modem (when needed). Since the originating switch knows the speed of the call, it can insert the appropriate pooled modem. Since tandem switches do not know the speed of the call, they cannot make the decision on the type of pooled modem needed.

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 and ITC to distinguish voice from data calls. For all trunks, ISDN-PRI as well as non-ISDN-PRI, the BCC and ITC are 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

  Generalized Route Selection 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 and ITC are determined by (default) administration. For ISDN-PRI trunking facilities, the BCC and ITC are 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.

- Data Pre-Indication

  When the Data button is pressed on DCP voice terminals, the switch uses the BCC and ITC of the associated data module.

- Electronic Tandem Network (ETN) Services

  The ISDN-PRI trunks can be used to interconnect DEFINITY 2.1 and DEFINITY Generic 1 or Generic 3i 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. (Traveling Class Marks represent the Facilities Restriction Level 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.

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.

  The ITC (Information Transfer Capability) field administers the type of traffic (restricted, unrestricted, or both) that this routing preference may carry. The value of the ITC field may be set to "rest" (restricted), "unre" (unrestricted), or "both". The default is "rest".

  The BCIE (Bearer Capability Information Element) field specifies how to create the Information Transfer Capability codepoint in the Bearer Capability IE of the SETUP message. This field only applies to ISDN-PRI trunks and is displayed and administerable if the ITC field is administered as "both". The value "both" provides extra flexibility for managing facilities in a mixed unrestricted/restricted environment. The value of the BCIE field may be set to "ept" (endpoint), "unr" (unrestricted), or "both". The default is "ept".

  If the BCIE field is set to "ept", the ITC of the originating endpoint is used to encode the SETUP message. If the BCIE field is set to "unre", then "unrestricted" is encoded in the SETUP message.

- Trunk Group BCCs — Each non-ISDN-PRI, data, AVD, or RBAVD trunk group is administered a BCC to identify the type of traffic on the trunks in that group.

  The ITC (Information Transfer Capability) field is provided for non-ISDN trunk groups only. This field is displayed and administerable only if the Comm Type field is administered as "data", "avd", or "rbavd" and the BCC field is not set to "0". The value of the ITC field may be set to "rest" (restricted) or "unre" (unrestricted). The default is "rest".

- Access Endpoint BCCs — For access endpoints where the Communication Type field is set to "56K-data" or "64K-data", the ITC (Information Transfer Capability) field is displayed and administerable. The value of this field may be set to "restricted" or "unrestricted". The default is "restricted".
- Station BCCs — For all non-ISDN endpoints, the ITC (Information Transfer Capability) field is displayed and administerable on the Station Data Module Page. The value of this field may be set to "restricted" or "unrestricted". The default is "restricted". This field is not displayed for voice-only stations and BRI stations.

  For BRI stations, the Default ITC (Information Transfer Capability) is displayed and administerable on the Station Data Module Page. This field administers the ITC for BRI data modules that originate an administered connection (AC). The value of this field may be set to "restricted" or "unrestricted". The default is "restricted".
- Data Module BCCs — For all non-ISDN endpoints specified as "pdm", "data-line", "netcon", or "tdm", the ITC (Information Transfer Capability) field is displayed and administerable. The value of this field may be set to "restricted" or "unrestricted". The default is "restricted".

  For BRI data modules, the Default ITC (Information Transfer Capability) is displayed and administerable. This field administers the Information Transfer Capability for BRI data modules that originate an administered connection (AC). The value of this field may be set to "restricted" or "unrestricted". The default is "restricted".

For endpoints (DCP data modules, voice terminals, and so on) 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 only after the call is ringing.

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 the call appearance being used for the call.

Single-line Voice Terminal Hold

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, and so on.

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. For a call transfer, the system ignores any subsequent presses of the Recall button or flashing of the switchhook. If the user is the controller of a conference call, and presses the Recall button or flashes the switchhook, the last party added to the conference is dropped. The user can transfer the call by hanging up after conferencing the call.

It is possible to receive priority ringing and have a call on soft hold.

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 by a single-line voice terminal user. A call involving an attendant can be held by a multi-appearance voice terminal user, unless the user attempts a conference or transfer of the call.

A call dialed by a single-line voice terminal can be dropped within the first ten seconds (after dialing is completed) by flashing the switchhook.

One party on hold can hear music if the Music-on-Hold 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.

- LWC

  A held multi-appearance voice terminal user can activate LWC toward the holding user.

  A single-line voice terminal user cannot activate LWC toward another user while a call is on soft hold.

- Personal Central Office Line

  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.

Hold - Automatic

This feature is new for the **G3i-Global**. System-Wide Administrable Automatic Hold (hereinafter referred to as Automatic Hold) is a station/attendant feature. The system comes with this feature turned off by default.

The feature is described for stations as well as the attendant because of the similarity of the operation. Automatic Hold allows attendants and Multi-Function station set users to alternate easily between two (or more) calls. For example, depression of a second call appearance automatically puts the active call (if any) on hold, and makes the second call appearance active. If the Automatic Hold feature is disabled (the default), the active call appearance is dropped.

The ability to administer the Automatic Hold feature allows the customer to enable and disable the Automatic Hold Feature on a system-wide basis. A user of a multi-button station may place on Automatic-Hold as many calls as the number of call-appearances minus one.

The controlling station can have only one ’soft’ auto-held call at a time. A ’soft’ hold is the state of a line after the conference or transfer button has been pressed but before either process is completed. The controlling station is guaranteed the ability to reenter any auto-held call later unless the auto-held party or parties disconnect or the auto-held tone times out.

Considerations

The Held Call Timed Reminder does not apply to conference calls and will, therefore, not be initiated when a conference is placed on hold. The net result is that the attendant Automatic Hold call will be treated the same as an attendant call placed in Hold by the depression of the HOLD button.

The Automatic Hold feature will operate in conjunction with the START key or Automatic Start feature of an attendant console. The START key/Automatic Start operation has precedence over the Automatic Hold feature. Any feature that uses the Start key/Automatic Start operation places the call on the active loop on Soft-Hold.

Whenever only the Automatic Hold feature is involved, for example, when the attendant on an active loop presses a second loop, the active call is placed in Hard-Hold. Soft-Hold allows the attendant to extend calls and conference calls together. When Automatic Hold is disabled, if the Multi-function Digital Telephone (MFDT) selects an inactive call appearance while there is still a call on an active call appearance, the active call is dropped. If Automatic Hold is enabled, when the MFDT selects an inactive call appearance while there is a call on an active call appearance, the active call is put on Hard-Hold. This is the same as if the MFDT had depressed the HOLD button and then selected a call appearance.

Interactions

As mentioned earlier, the Automatic Hold feature is identical to the action generated when an MFDT places a call on hold using the HOLD button and then selecting the (another) inactive call appearance. The features, therefore, work identically, except for the saving of a button push. The Automatic Hold feature is implemented as an equivalent to pressing the HOLD button. A call placed in Automatic Hold will be in Hard-Hold, as with the depression of the HOLD button.

Therefore, Automatic Hold, when invoked by the methods already mentioned, will operate identically to the HOLD feature and its interactions. See the feature description for the HOLD feature for additional information.

Automatic Hold interacts with DCS and Centralized Attendant Service (CAS). It operates transparently with DCS and CAS. Note that the Auto-Hold feature is administered separately for each node in a DCS network.

Administration

Automatic Hold is administrable on a system-wide basis only by persons whose administration ID provides access to system parameters. Administration of the auto hold feature takes place on the System-Parameters Miscellaneous Form.

Hardware/Software Requirements

No special hardware 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, AAR, 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 COR. Call reception is not affected by Hot Line Service. Likewise, the Hot Line Service destination is not affected by Hot Line Service.

A DDC, a UCD, a 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.

Interactions

A Hot Line Service user cannot activate any feature unless the access code is, or is part of, the destination number:

- Bridged Call Appearance — Single-Line Voice Terminal

  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 COR. 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 ACD, Call Coverage, DDC, and UCD features. The order of hunting is defined under each individual 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.

Agents should not be used for hunt group calls and ACD split calls simultaneously. Otherwise, all of the calls from one split (either ACD or hunt group) will be answered first. For example, if the ACD calls are answered first, none of the hunt group calls will be answered until all of the ACD calls are answered.

The oldest call waiting termination is only supported for agents who are servicing ACD calls only.

Interactions

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 ACD (G1.1, or G3i), Call Coverage, DDC, and UCD features. Administration of each of these features is discussed under that feature elsewhere in this chapter.

Hardware and Software Requirements

No additional hardware or software is required for Call Coverage, DDC, and UCD. ACD requires ACD software. Call Vectoring is also required for vector-controlled splits (G3i).

Inbound Call Management (G3i)

Enhances the ACD feature by providing improved automation of the handling of inbound calls for applications such as telemarketing s. The result is an increase in agent efficiency and tracking. With Inbound Call Management (ICM), data screen delivery can be automated. The system uses a CallVisor ASAI to interface the switch with a CONVERSANT® voice system adjunct. This two-way link lets the CONVERSANT voice system determine who should get incoming calls and control the routing of these calls.

The switch can interface with a CONVERSANT voice system. ICM uses the CallVisor ASAI capabilities to share information, such as ISDN, CPN, ISDN-BN, DNIS information, and digits collected by Call Prompting, with a CONVERSANT voice system. The CONVERSANT voice system can use the information to re-route the call or to display the proper screen on the agent's data terminal.

The following are some of the ICM applications for an inbound telemarketing. These applications are described later in more detail:

- The switch can pass information (such as ISDN CPN/BN) and call outcomes in forms of event reports to a CONVERSANT voice system adjunct for data screen selection, delivery, transfer, and duplication.
- The switch can request that the CONVERSANT voice system make a call routing decision based on information about the call,
- The switch can collect touch-tone digits (using the Call Prompting feature) which are sent to the host for data screen delivery or adjunct routing,
- The switch can send the DNIS and CPN/BN information to a speech processor for selection of the proper processing.
- A CONVERSANT voice system can transfer a call to a particular ACD agent and have the call treated as an ACD call. This is called Direct Agent Calling.

For full ICM functionality, the switch supports the following CallVisor ASAI capability groups:

- Adjunct Call Control Group
- Adjunct Routing Group
- Event Notification Group
- Request Feature Group
- Set Value Group

Data Screen Delivery

Passing incoming call information (CPN/BN, DNIS, Lookahead Interflow information, digits collected from Call Prompting, agent selected) to a CONVERSANT voice system can be used to deliver the appropriate data screen when the voice call is delivered to an agent. Data screens can also be transferred or duplicated by the CONVERSANT voice system for transferred or conferenced calls. A simplified configuration of this type of application is shown in the following figure.

In this application, the CONVERSANT voice system or host requests notification for events (call offered, call ended, call connected, call dropped, call transfer, alerting, and so on). The switch notifies the CONVERSANT voice system using event reports when the call arrives, when the agent answers, when the call drops, and so on. Knowing a call drops prior to being answered, the CONVERSANT voice system can track abandoned calls or use CPN/BN information for call backs.

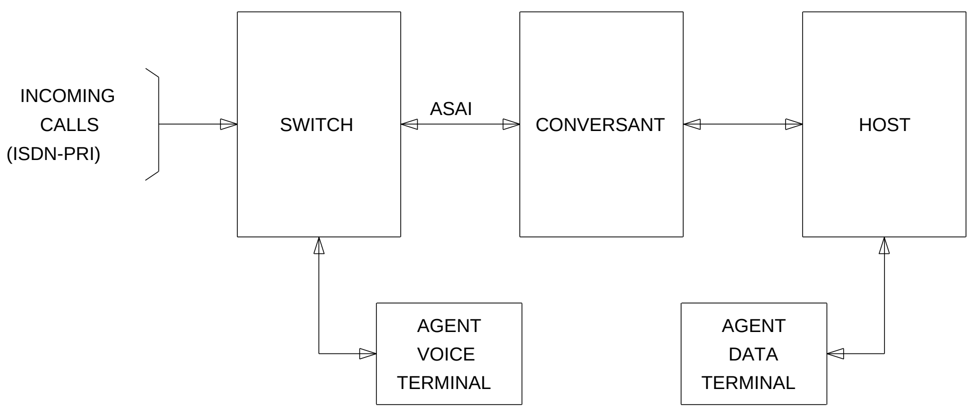


Figure 2-12. Simplified ICM Configuration for Data Screen Delivery

Integration With Speech Processing Adjuncts

ICM can be used to provide integration with Voice Response Units (VRUs). The advantages of using ICM with the CallVisor ASAI in addition to tip/ring interfaces are as follows:

- Data screen integration is provided on transferred calls.
- Notification of answer is provided on internal calls (CallVisor ASAI capabilities let you know what happens with the call).
- Delivery of ISDN network information such as CPN/BN/DNIS is provided (instead of having to prompt for this information).

A simplified configuration of this application is shown in the following figure. In this application, the CallVisor ASAI link is used by the switch to pass incoming call information to the CONVERSANT voice system. The call is distributed by the switch ACD to an available voice line. After collecting digits via a touch-tone keypad, the CONVERSANT voice system transfers the call back to an ACD split or specific agent on the switch via CallVisor ASAI messages. If the call is transferred to a split agent, the CallVisor ASAI link is used by the switch to pass an event report containing which agent in the split receives the call. The CONVERSANT voice system forwards the agent identification to the host for delivery of the associated data screen to the agent selected to handle the call.

Digits collected by the CONVERSANT voice system are not passed to the switch to display on the agent's voice terminals but can be displayed on the agent's data terminals. If the digits collected by the CONVERSANT voice system are the extension to which the call is to be routed, these routing digits are passed to the switch as the destination in the CallVisor ASAI third party make call request. The third party make call request is used by the CONVERSANT voice system to set up various types of calls.

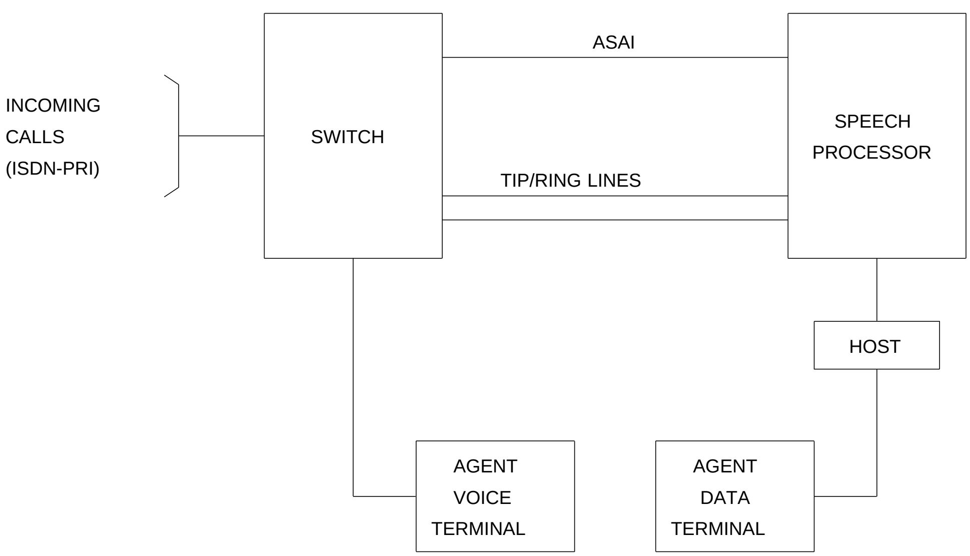


Figure 2-13. Simplified ICM Configuration for Speech Processor Integration

Host/Adjunct Call Routing

Incoming call information can be used by the host or CONVERSANT voice system adjunct to route the call to a split, vector, or particular agent (basically any valid extension number). The call could even be routed off of the switch if desired. The CONVERSANT voice system can also use the incoming call information to tell the switch that the call should be treated as a priority call. Routing can be based on the area code dialed from, the country code, digits collected from the Call Prompting feature, dialed number or service, agent availability, or information in a customer database.

To implement adjunct (CONVERSANT voice system) call routing, calls must come into a vector which contains an **adjunct routing** vector command. When the **adjunct routing** vector command is encountered, the switch initiates the route CallVisor ASAI capability. Vector processing proceeds with the next step (which could provide ringing, announcements, music, and so on) while the caller waits. A default split or answering position can also be specified in the vector, in case the CONVERSANT voice system does not respond in the administered amount of time (determined by the announcement/wait steps). Announcement and wait steps are needed to give the host time to respond.

Direct Agent Calling

DAC is a new function that allows an adjunct to initiate or transfer a call to a particular ACD agent and have the call treated as an ACD call.

Calls that originally enter the switch as ACD calls and are rerouted to a particular agent via adjunct routing, or transferred from a tip/ring agent to a live agent via a "third party make call" request, are treated as ACD calls for the duration of the call. This is important for a number of reasons. Agents need to receive zip tone when these calls are delivered. Agents may have After Call Work associated with these calls. The CMS and BCMS correctly measure these calls as ACD calls.

Direct Agent Calls have the highest priority of any calls.

Adjunct Activation of Direct Agent Calling

CallVisor ASAI third party make calls and route select calls with the direct agent call option are treated as direct agent calls. The receiving agent’s extension appears as the destination and the split extension in the direct agent call option.

Delivery of DAC

If the agent receiving the direct agent call is available to answer an ACD call in the associated split, the direct agent call is delivered to the agent. Zip tone (480 Hz for a 1/2 second, not repeated) is applied if the agent is automatic answer.

If the receiving agent is not available to answer an ACD call (for example, the agent is busy on a call, in the After Call Work mode, or in the Auxiliary Work Mode), the receiving agent is notified with a ring-ping if the agent has a multi-function voice terminal or is on-hook. If the receiving agent has a single-line voice terminal and is not available, the receiving agent will hear call waiting tone (even when the Call Waiting feature is not assigned) if the agent is off-hook. The ring-ping or call waiting tone is given only once per call when the call is queued. The active work mode button lamps for the associated split on the receiving agent’s voice terminal will do a fast flutter, indicating a direct agent call is waiting. This starts when the first call queues and stops when all direct agent calls leave the queue (answered, abandoned, or sent to coverage).

The originating agent hears normal call progress tones and ringback. If the originating agent drops from the call, the caller hears call progress tones and ringback. A forced first announcements will not be heard by the originating agent or caller.

Direct agent calls are queued and served in a first-in first-out order, before any non-DAC. Therefore, when an agent becomes available, the switch first checks for any direct agent calls before serving normal ACD calls in queue.

The voice terminal display for the receiving agent, before a transfer is complete, shows the originating agent's name and number. The voice terminal display for the receiving agent, after the transfer is complete, shows the caller identification (CPN/BN or trunk group name for external calls, and name/number for internal calls) and the original split or VDN name.

Direct agent calls follow the receiving agent's coverage and call forwarding, if activated. Once the call goes to coverage or is forwarded, the call is no longer treated as a DAC. CMS is informed that the call has been forwarded.

Answering a Direct Agent Call

The receiving agent answers a direct agent call by becoming available in the split with which the direct agent call is associated. While on a direct agent call, the agent becomes unavailable to subsequent ACD calls.

If the receiving agent logs off by unplugging the headset, the agent may still answer a direct agent call in queue by logging back in and becoming available.

Considerations

A maximum of eight CallVisor ASAI links may be assigned.

Information from a CONVERSANT voice system (except for the dialed number) cannot be carried with the call and displayed on a voice terminal. For example, digits collected in a CONVERSANT voice system adjunct cannot be passed to the switch for display.

CallVisor ASAI and BX.25 CPN/BN-ANI are not supported simultaneously.

Interactions

The following features and functions interact with the ICM feature:

- Automatic Answer

  DACs to agents assigned automatic answer will receive single zip tone answer.

- Queue Status Indications

  DAC are not included in the number of calls waiting and oldest call waiting on the queue status indication for the split.

- Call Coverage

  If the split associated with a direct-agent call has call forwarding or night service activated for the split, then the direct-agent call will be forwarded. If the priority calling option is requested, the direct-agent call will forward with priority ringing at the night service destination.

  After a DAC call successfully terminates to a split, if the destination agent has Call Forwarding or Send All Calls activated, then the DAC calls will be forwarded. If the priority calling option is requested, then the DAC call will forward with priority ringing at the call forwarded destination, but will not forward to the covering point in the case of Send All Calls when the priority calling option is requested.

Direct-agent calls follow the destination agent's coverage path. If the priority calling option was requested, the DAC follows the standard priority call rules for coverage, meaning the call will not go to coverage. Calls (either regular ACD or direct-agent) in queue will remain in queue until the caller abandons or agent answers.

- Call Forwarding

  New DAC will forward if the agent activates Call Forwarding. Direct Agent Calls already in queue prior to Call Forwarding activation will remain in queue.

- Call Vectoring

  Call Vectoring is used in conjunction with CallVisor ASAI capabilities. The **adjunct routing** vector command is required. For adjunct routing, if the call queues to a split or the call leaves vector processing, a route end request is sent to the CONVERSANT voice system.

  Direct agent calls cannot go through vectors.

- Call Prompting

  Digits collected by the Call Prompting feature become part of the current call information that is passed to a CONVERSANT voice system adjunct.

- Call Waiting

  Call waiting tone is used to notify single-line users that a DAC is waiting, whether the call waiting feature is optioned or not.

- DCS

  DAC cannot be made over a DCS link. If the receiving agent is not an internal extension, DAC is denied.

- Night Service

  DACS will go to Night Service if Night Service is activated for the receiving agent's split.

- Priority Calling

  CallVisor ASAI capabilities permit both Priority Calling and DAC for the same call. Priority DAC will not go to coverage.

- Send All Calls

  If an agent activates send all calls, *new* DACs will immediately go to the agent's coverage. DACs already in queue prior to Send All Calls activation will remain in queue until the coverage ringing timeout occurs.

Administration

ICM is administered by the System Manager.

The ACD feature must be administered as described in the ACD feature description elsewhere in this manual.

If the Call Vectoring and/or Call Prompting features are to be used, these features must be administered as described in the Call Vectoring and Call Prompting feature descriptions elsewhere in this manual.

In order to make or receive direct agent calls, an agent must be assigned a COR that allows DAC.

Direct agent calls wait in split queues. Split queues must be properly sized.

Hardware and Software Requirements

The only additional hardware requirements for ICM are those for the CallVisor ASAI. These requirements are in the following paragraphs.

The system supports both the TN748C (TN420C, TN744 support A-law) and the TN744 circuit packs for use as tone detectors. With respect to CallVisor ASAI features, the TN744 is required for those customers who desire switch call classification.

Each port on the eight port TN744 acts as a touch-tone receiver or call classifier. Each call classifier port is capable of detecting tones as well as Special Information Tones.

This tone detection will work only if the public network provides similar tones to those used in the U.S.

Each CallVisor ASAI BRI Interface Link requires a port on a TN556 ISDN-BRI circuit pack.

CallVisor ASAI Interface software is required.

Individual Attendant Access

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 individual attendant has a queue that allows two incoming calls to wait. This individual attendant queue has priority over all other attendant seeking calls. The Individual Attendant Access is placed in the Attendant queue according to the priority assigned in Attendant Priority Queueing.

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 DDC group or a 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 individual attendant extension can have its own Class of Restriction and Class of Service.

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 could receive calls as a hunt group member for more efficient handling of calls.

Interactions

The Individual Attendant Access is placed in the Attendant Queue according to the priority assigned in Attendant Priority Queuing.

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.

- CAS

  Individual attendants can be accessed when CAS is in effect.

- COR and 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.

- DDC and 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.

- LWC

  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
- COR
- COS

Hardware and Software Requirements

No additional hardware or software is required.

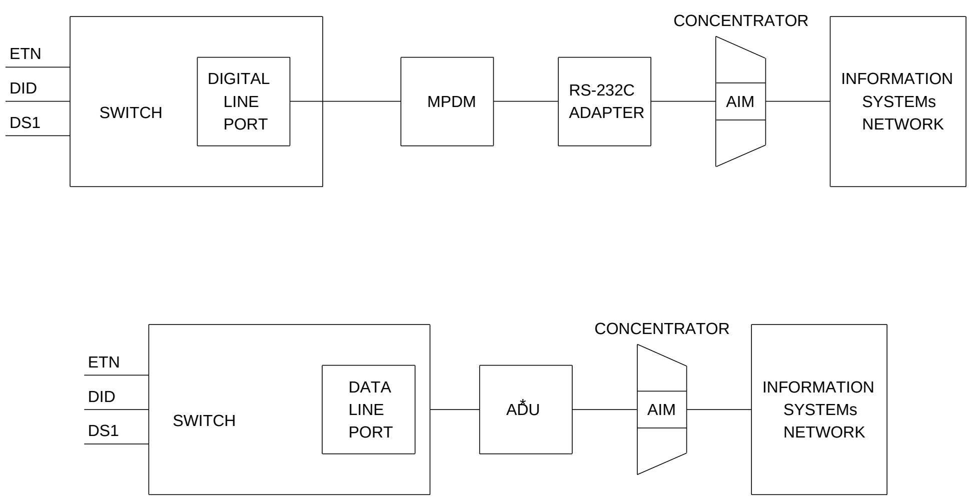
Information System Network (ISN) Interface

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 the system is via an ADU. A 7400A data module (G3i), 7400B data module (G3i), or a MPDM may be used, but the ADU is more economical.

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 MPDM may be used instead of the ADU, but the ADU is more economical. The MPDM or ADU connects to an AIM on the Packet Controller or Terminal Concentrator (see the following figure This interface allows the system and the ISN to share data capabilities.



* FUTURE VERSIONS OF ISN WILL HAVE INTEGRATED ADUs.

Figure 2-14. 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 1,920 data ports.

Interactions

The following features interact with the 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 two-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 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.
- 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:

- **7** (PRS) is also used for a Q.
- **9** (WXY) is also used for a Z.
- ***** is used for a space or comma.
- # 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 **2** to key in the letter C, the display might show `Abbott,Lynn A` and an extension number. (**2** matches A before it matches C.) If the user presses **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 **7** to key in an R, the display might show `Carr,Danny` and an extension number.

At this point, the user can press **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.` You can then enter another name. 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 1,600 entries. The maximum length of the name is 15 characters (including spaces and commas). The extension number cannot exceed five digits.

A maximum of 10 users can activate Integrated Directory at the same time. If more than 10 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 **2**. Pressing **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 **2**.

When the voice terminal is in the Integrated Directory mode, it cannot be used to make calls or access features by dial code. It can, however, still be used to activate other features or 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.

The set of characters allowed in the Integrated Directory database are the alphanumeric characters "A" to "Z", "a" to "z" and "0" to "9" as well as "blank" and "comma" which are used as delimiters. In addition, the following special characters are allowed: "hyphen", "apostrophe", "period", and "ampersand". These special characters, though considered legal characters, will not be entered into the Integrated Directory. Instead, a "period" will be replaced by a "space" while "apostrophe", "hyphen" and "ampersand" are all ignored.

If a character outside the allowed set is entered as the name of a station or data module, the directory search for that name will fail.

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. The Return Call button can be used to dial prefixed extensions because the 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 — Basic Rate Interface (G3i)

Description

Allows connection of the system to equipment or endpoints that support an Integrated Services Digital Network (ISDN) by using a standard ISDN frame format called the Basic Rate Interface (BRI).

An ISDN provides end-to-end digital connectivity and uses a high-speed interface which provides service-independent access to switched services. Through internationally accepted standard interfaces, an ISDN provides circuit or packet-switched connectivity within a network and can link to other ISDN supported interfaces to provide national and international digital connectivity. Two types of ISDN interfaces are currently defined: the PRI and the BRI. This description focuses on ISDN-BRI.

The ISDN-BRI is a 192-kbps interface that carries two 64-kbps B-channels, which transport voice and/or data, and one 16-kbps D-channel, which transports data, signaling, and other bits for framing. Although BRI supports data transmission on the D-channel, DEFINITY Generic 3i does not support this capability. DEFINITY Generic 3i provides only signaling on the D-channel. This feature is available in environments that support mu law companding. See the System Description for a fuller description of companding.

NOTE:
The word ‘‘endpoint’’ is used whenever statements apply to BRI voice terminals, BRI data modules, and integrated BRI voice/data terminals.

ISDN-BRI Endpoint Configurations

With ISDN-BRI, there are two possible configurations:

- point-to-point — only one endpoint connected to a BRI port
- multipoint — multiple endpoints connected to a BRI port. This configuration is also referred to as ‘‘passive bus configuration.’’

Because DEFINITY G3i BRI provides non-blocking voice and data services, the number of endpoints supported on one BRI port in a multipoint configuration is restricted to a maximum of two.

Both B-channels provided on each BRI interface are resources which are dynamically chosen by the switch or BRI endpoints for voice or data service requests. Because there are two B-channels, only two simultaneous service requests can be granted at any time on a BRI port to provide non-blocking service (either through point-to-point or multipoint configurations).

When one endpoint is capable of providing two service requests (such as an integrated voice/data endpoint), the endpoint must be configured point-to-point because both B-channels can potentially be used simultaneously by the two services. You can have the following endpoint types in the point-to-point configuration:

- one voice-only endpoint
- one stand-alone data endpoint
- one integrated voice/data endpoint

NOTE:
Even though the integrated voice/data endpoint supports two services requests (that is, both voice and data), this integrated endpoint is not considered to be in multipoint configuration because it is only one endpoint.

Two endpoints that are each capable of providing only one service request can be administered on the same BRI port. By connecting two endpoints on the same BRI port, you have a multipoint configuration. In this situation, both B-channels can potentially be used simultaneously by the two service requests. Since each BRI port provides two B-channels, no more endpoints can be administered on this BRI port. You can have the following endpoint types in the multipoint configuration:

- two voice-only endpoints
- two stand-alone data endpoints
- one voice-only endpoint and one stand-alone data endpoint

Terminal Equipment Identifier (TEI)

The terminal equipment identifier (TEI) is used to set up communication between the switch and an endpoint. DEFINITY G3i supports two types of TEIs: fixed and automatic. A fixed TEI endpoint supports one fixed TEI value (zero to 63), which is encoded into the terminal equipment, and the fixed TEI initialization procedure. When administering a fixed TEI endpoint, you must assign the endpoint's fixed TEI value to the station or data module forms for that endpoint. If the endpoint's fixed TEI value differs from the TEI assigned to the station or data module forms for that endpoint, no communication will be established between the switch and the endpoint. As a result, the endpoint will be incapable of providing services. Normally, the manufacturer specifies the fixed TEI value encoded into the terminal or provides procedures for modifying the fixed TEI value.

An automatic TEI endpoint supports automatic TEI initialization procedures and receives a TEI from the system during initialization. With automatic TEI endpoints, you are not entering any TEI values to the station or data module forms. In DEFINITY G3i, only the automatic TEI endpoints are permitted to be used in multipoint configurations. Currently, all supported BRI endpoints are automatic TEI endpoints.

Service Profile IDentifier (SPID)

When more than one endpoint is connected to a BRI port (for example, a multipoint configuration), the switch uses the Service Profile Identifier (SPID) to associate endpoints with the administered station or data module extensions. The SPID enables the switch to differentiate between the endpoints connected to the same BRI port.

You must administer the SPID on the station or data module forms, and then program the SPID in the BRI endpoint using the procedure in the endpoint's users' manual. During initialization, the endpoint sends the SPID to the switch. The SPID administered on the station or data module administration forms must match the SPID which is programmed into the endpoint. If the SPID on the station or data module administration forms does not match the SPID programmed into the endpoint, the system will restrict service to that endpoint.

SPID administration and programming are required for a multipoint configuration. However, SPID administration is optional in a point-to-point configuration because there there is only one endpoint connected to the BRI port. If the SPID is administered in a point-to-point configuration, it must match the SPID programmed into the endpoint. If the SPID is not administered, the switch will use the port to associate the endpoint to the administered station or data module extension.

ISDN-BRI Voice/Data Services

Voice transmission on ISDN-BRI is provided by the 7505, 7506, 7507, and 8503T voice terminals. All tests and services available to DCP users are also available to BRI users.

Data transmission on ISDN-BRI is provided by the 7500B Data Module and the ADM. The 7500B Data Module is a stand-alone unit that supports asynchronous or synchronous DCE and asynchronous DTE. In asynchronous mode, the 7500B supports packet- or circuit-switched data communications, and can be controlled via the front panel or the keyboard of a connected terminal. In synchronous mode, the 7500B supports circuit-switched or nailed-up data communications, requires either the Multi-purpose Enhancement Board or the High-Speed Synchronous Enhancement Board, and can only be controlled via the front panel.

The ADM may be used with asynchronous DTE as a data stand for 7500-series BRI voice terminals. Consisting of a board located inside the BRI voice terminal, the ADM allows the transmission of integrated voice and data through one voice terminal. The ADM supports the Hayes command set for compatibility with PC communications packages.

Endpoint Initialization

BRI endpoints have to successfully complete endpoint initialization procedures in order to be fully operative. It is usually carried out at the time of installation, or as part of reconfiguration.

Multipoint Configurations on BRI ports

In a passive bus multipoint configuration, the system supports two BRI endpoints per port, thus doubling the capacity of the BRI circuit pack. When changing the configuration of a BRI from point-to-point to multipoint, the original endpoint need not be reinitialized. However, only endpoints that support SPID initialization can be administered in a multipoint configuration.

Exchange of User Information

The BRI protocol provides the users the capability of exchanging up to 128 octets of user information end-to-end. The information is passed in the User Information IEs to the receiving endpoint without being interpreted by the switch. However, there are some limitations to the exchange of User Information IEs.

ISDN-BRI Data Services

Basic Digit Dialing

Regular digit dialing is provided through the Asynchronous Data Module (ADM). Digits from 0 to 9, "*", and "#" can be entered. This feature can be used by the user either from the set keypad or from the EIA terminal interface.

Default Dialing

Default Dialing is also an enhancement to the user dialing capabilities of the Data Call Setup feature. By either typing a **d** followed by ***Return*** or pressing the data button twice, if a default address is administered, the switch will terminate the call to the default address. If no default dialing has been administered, the call will be disconnected in less than one second. This feature is mutually exclusive with the Data Hotline feature.

Data Hotline

Data Hotline is a security feature. It allows a user to enter just a Dial command with no address specified followed by a ***Return***. The switch will terminate the call to a preadministered hotline destination. If a user enters an address either intentionally or unintentionally, the call processing will discard the address string received for the hotline endpoint. The call processing will automatically route the call just as if the hotline destination address had been entered by the user. This service does not impose any restriction on incoming calls received at the endpoint. This feature is mutually exclusive with the Default Dialing feature.

Administered Connections

An Administered Connection is an end-to-end connection between two access endpoints or data endpoints that is automatically established by the system whenever the system is restarted or the Administered Connection is administered and the Administered Connection is due to be active. The attributes of these connections are user-defined. To administer Administered Connections, use the Administered Connection form via the SAT.

Once the ADM has been administered as one endpoint of an administered connection, the system waits for the scheduled time to set up the connection. At the scheduled time, the system establishes the connection and maintains it for the length of time specified. Once the call is accepted, the set will enter into the continuous mode for the length of time specified. If the switch is rebooted during the continuous connection, the connection will reinitiate the call setup. At any time that the connection drops (for example, disconnected cabling), the switch will re-initiate the call setup.

Call Request

DEFINITY Generic 1 and Generic 3i call processing will handle all various BRI Bearer data call requests that are presently defined. Some capabilities that are not supported by AT&T terminals may be provided by a non-AT&T terminal. The switch will complete most call requests. For those capabilities that the switch doesn't support, a proper cause value will be sent back to the terminal.

Cause Value

BRI stations will receive a cause or reason code that identifies why the call is being cleared. The BRI data modules will convert certain cause values to text messages and display them for the user.

Considerations

A system that is fully configured for BRI can support a maximum of 1,000 voice and data BRI endpoints. The actual system maximum depends on the mix of the various types of BRI endpoints. For example, the capacity of the system will be considerably reduced if all of the endpoints are BRI stations with 30 call appearances/bridged appearances and display capabilities.

The system can support a maximum of 800 data modules. This value includes both BRI and DCP data modules.

The following features are not provided to BRI users:

- The following data functions are not provided to the BRI voice users, since the ISDN-BRI protocol requires that the Bearer Capability should be specified at the time of sending the SETUP message, and cannot be changed during the call:
  - — One button voice call setup transfer to data
  - — One button data call setup transfer to voice
  - — Preindication of a data call
  - — Voice call transfer to data and data call transfer to voice

  These functions require a change in the Bearer Capability after the establishment of the call, which is currently not allowed by the BRI protocol.
- BRI attendant is not supported on the G3i switch.

- Features that use the switchhook and Recall button (for example, Call Waiting and Analog Conference/Transfer/Hold/Drop) are applicable to analog voice terminals only.

If you are using a 7506D or 7507D to make calls that require additional digits, a comma may appear in the dial sequence after you receive second dial tone or after the call has been set up. The comma is used to separate the called number from subsequent information.

Interactions

The following features interact with the ISDN-BRI services:

- Data Button

  Besides the call appearance and feature function buttons, BRI voice/data terminals have a fixed, dedicated data button (button 7 on the 7505D and 7506D voice terminals, and button 31 on the 7507D voice terminal) that is used for data call setup. In general, feature function buttons such as Call Forwarding or Send All Calls buttons are always associated with voice features, and cannot be used in conjunction with the data button. For example, the user cannot activate call forwarding for the associated data endpoint by using the data button followed by the call forwarding button and the designated extension.

- Interworking

  The same off-premises call types are permitted as for DCP, with the exception of voice to data and data to voice transfer.

- Modem Pooling

  The Modem Pooling feature provides the necessary protocol conversion between Mode 2 digital data endpoints and analog data facilities. A modem pool resource needs to be inserted by call processing during call setup for both call origination and call answering. This resource translates data between DMI Mode 2 protocol used by BRI data endpoints and the modulated signal used by the modem.

- Voice Terminal Display

  BRI terminals take control of the display. For example, when the user is in dialing state (BRI terminal is in the Overlap Sending state) any display information sent to the terminal from the switch will be buffered until the state changes and will be displayed when the state changes.

  The 7506 BRI voice terminal, which has a two-line twenty-fourth character display, will split a message when it recognizes a blank closest to the 24th character. This is left to the discretion of the terminal. As a result, the switch has no control over it.

Administration

BRI Voice/Data

Administration of BRI voice terminals requires all the fields associated with the standard DCP station administration. Additional fields are used to enter the following information:

- TEI information: If the BRI terminal supports a fixed TEI value, it has to be entered at the time of station administration. The allowed values are 0 - 63. There are two fields: `Fixed TEI` and `TEI value` If the answer to the first field is `yes`, the second field shows up where the TEI value is administered. The TEI value on the administration form must match the value supported by the terminal.
- MIM support: This is an administerable field on the BRI station form. If the answer to this field is `yes`, the following two fields need to be filled in:
    1. `Endpoint Initialization:` If the BRI terminal supports endpoint initialization, the administrator has to enter the SPID value. The default value is the extension; however the value can be changed at the time of administration. The SPID can be up to 10 digits, and uniquely identifies the terminal on the BRI. The SPID on the Administration form and the SPID programmed into the endpoint must be the same. (Refer to the terminal's user manual in order to change the terminal's SPID.) All SPIDs must be different for each endpoint on the same port. All SPIDs must be different from the service SPID, which is administered on the System Maintenance form.
    2. `MIM Maintenance / Management support:` This is another BRI specific field that indicates if the terminal supports other maintenance and management messages.

For multipoint (passive bus) environment, the system administration will check the number of B-channels used for a port. Administration will deny any attempt which might cause call blocking by restricting the number of endpoints on a port.

BRI Data

The 7500B Data Module is administered through the data module administration form. The ADM is administered through the station administration form, using the data module administration page. In addition to the fields used by the DCP endpoints, the following new fields are used for BRI data module administration:

- Default Duplex: Full/Half (default Full)
- Default Mode: Synchronous/Asynchronous (default Asynchronous)
- Default Speed: 1,200, 2,400, 4,800, 9,600, 19,200, 56,000, and 64,000 (default 1,200)

  In synchronous mode, the speed of the 7500B data module may be set to 56,000 or 64,000 bps.

- Default Data Application: mode 0, mode 1, mode 2 sync, mode 2 async, and mode 3/2 adaptable

Default Duplex, Default Mode, and Default Speed values are used for initializing data module default attributes. The defaults are required for modem pooling conversion resource insertion when the endpoint does not support MIM query capability. If you are using supported endpoints (7500B and ADM), you should not change the default values. Changing the default values with the MIM Maintenance/Management support option as y will have no effect on modem pooling. These endpoints support the MIM query capability, which enables the switch to query the endpoint when a call arrives.

Default Data Application specifies the default data protocol to be used for originating data calls if mode is not specified with the calling parameters. This mode will also be used for Administered Connections and for terminating trunk calls that do not have bearer capability specified.

Hardware Environment

BRI services require the following hardware:

- The TN778 Packet Control circuit pack which provides the interface to the LAN (packet) bus on S75 R1V5, for establishing the signaling connectivity
- The TN556 BRI port circuit pack which is the Basic Rate Line circuit pack. Each BRI port board can support 12 line interfaces, each operating at 192 kbps
- ISDN-BRI Type B and Type D Terminal Management S/T interface terminals
- The AT&T ISDN 7505, 7506, 7507, and 8503T voice terminals
- The 7500B Data Module and the ADM. The ADM is supported by the AT&T ISDN 7505, 7506, and 7507 voice terminals with firmware version FP2.0 or later.

Integrated Services Digital Network — Primary Rate Interface

Description

Allows connection of the system to an Integrated Services Digital Network (ISDN) by using a standard ISDN frame format called the PRI. The ISDN gives the system users access to a variety of public and private network services and facilities.

The T1 is a digital transmission standard that in North America carries traffic at the digital signal level-1 (DS1) rate of 1.544 Mpbs. T1 facilities are also used in Japan and some Middle-Eastern countries. It 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 CCITT Recommendation Q.931 and Q.921 for ISDN signaling.

The E1 is a digital transmission standard that carries traffic at a rate of 2.048 Mbps. The E1 facility is divided into 32 channels (DS0s) of 64 Kpbs information numbered from 0-31. Channel 0 is reserved for framing and synchronization information. When a D-channel is present, it occupies channel 16.

ISDN-PRI signaling in the system is supported by the TN767 (for 24 channels), the TN464B (for 32 channels), or the TN464C or later version (for 24 or 32 channels) DS1 Trunk circuit packs 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:

- 4ESS Switch
- DEFINITY Communications System Generic 2.1
- DEFINITY Communications System Generic 1
- DEFINITY Communications System Generic 3i
- Some of the 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 the following figures. As seen in these figures, the ISDN-PRI can be used to interface a PBX to a Public Switched Network, a PBX to a Host Computer, or a PBX to another PBX.

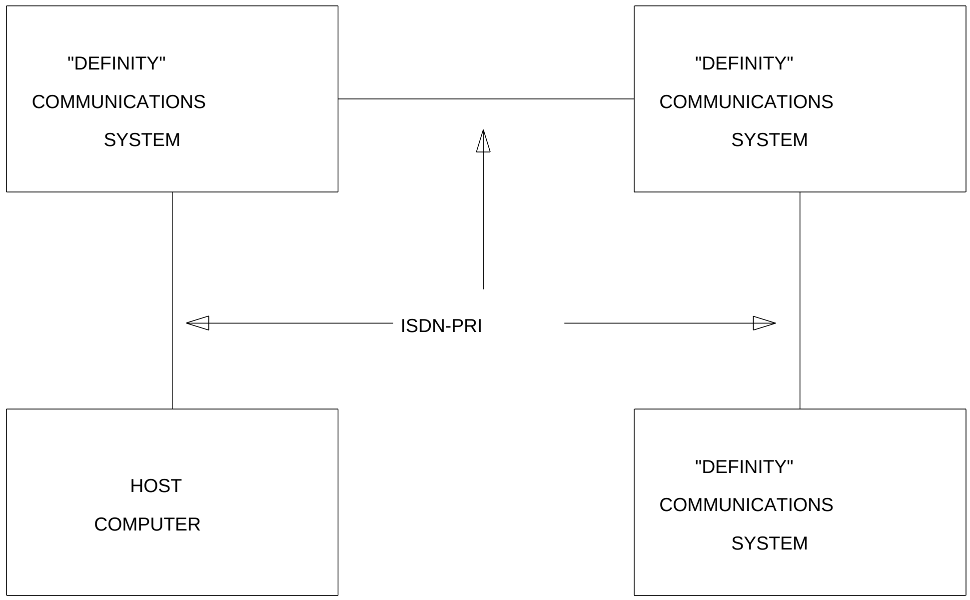


Figure 2-15. ISDN-PRI Private Network Configuration

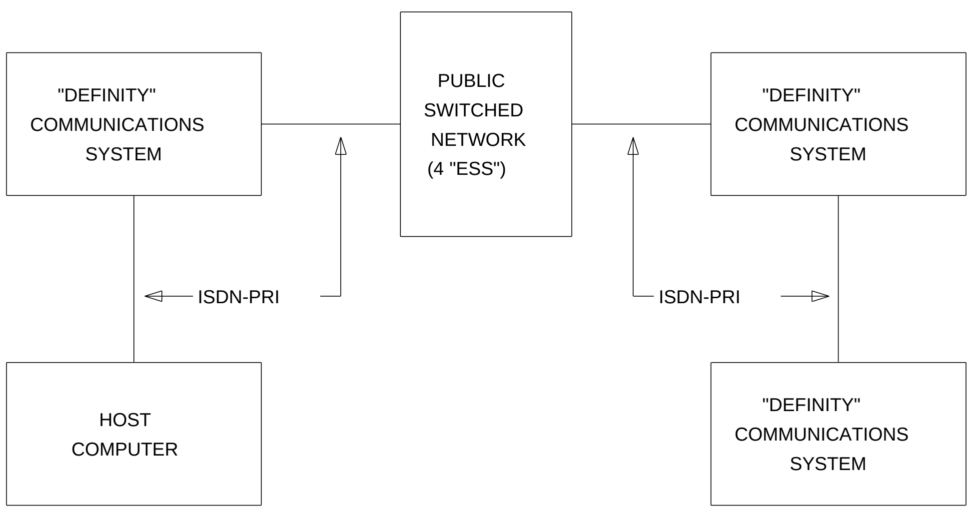


Figure 2-16. ISDN-PRI Public Network Configuration

ISDN-PRI Services

The ISDN-PRI provides system users with following services:

- Call by Call Service Selection

- Access to the Software Defined Data Network (SDDN) (G3i)
- Access to Switched Digital International (SDI) (G3i)
- Call Identification Display
  - Station Identification (SID) (G1.1), or CPN (G3i)
  - Automatic Number Identification (ANI) (G1.1), or Billing Number (BN) (G3i)
  - Calling and Connected Number Display
  - Calling and Connected Party Name Display
- SID/ANI (G1.1) or CPN/BN (G3i) to Host Call Identification
- 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 SDN, MEGACOM telecommunications service, MEGACOM 800 service, and so on) 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 manual.

Software Defined Data Network (SDDN) (G3i)

With ISDN-PRI, the Software Defined Data Network (SDDN) service may be accessed. SDDN provides virtual private line connectivity via the AT&T switched network (4ESS switch). The services provided by SDDN include voice, data, and video applications. SDDN services complement the Software Defined Network (SDN) voice services.

SDDN offers an attractive alternative to the traditional design of data communications networks, through high speed data networking, extensive functionality, advanced call handling capabilities, and network flexibility.

SDDN also offers the Automatic Restoration capability, which restores disrupted connections between access endpoints (non-signaling trunks) and data endpoints (devices that connect the switch to data terminal/communications equipment). This restoration is achieved within seconds of a service disruption so that critical data applications can remain operational. The Auto Restoration function is described in more detail in the Administered Connections feature, described elsewhere in this chapter.

End-to-end connections used for the SDDN service may be established via the Administered Connections feature, described elsewhere in this chapter.

SDDN is well suited for data communications applications which have one or more of the following characteristics:

- Have a time window for completion (performed during night, morning, or other specified time periods).

- Have requirements for high speed data or high volume data.
- Have restoration requirements. These are critical applications which must remain operational in the event of a network failure.
- Have multiple end-point destinations (require serial or non-simultaneous communication between an originating point and several endpoints).
- Would benefit from network flexibility (traffic patterns show daily or seasonal variations, and may change as the network grows).

Switched Digital International (SDI)

SDI provides 64 kbps unrestricted connectivity to international locations via the AT&T switched network. It is also the backbone for the AT&T International ISDN network. SDI complements the ACCUNET digital service already available to United States locations. This service can be accessed using the Call-by-Call Service Selection feature. SDI can provide economical high speed data transfer to international locations.

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:

- SID (G1.1) or CPN (G3i): A 0-15 digit DDD number associated with a specific station. When a system user makes a call that uses the ISDN, that user's SID (G1.1) or CPN (G3i) is provided by the system for the ISDN.
- ANI (G1.1) or BN (G3i): 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 (G1.1) or BN (G3i) is the customer's billing number stored in that network. If the SID (G1.1) or CPN (G3i) is not provided on an incoming ISDN-PRI call, the system uses the ANI or BN 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 0-15 digit DDD number. This number is provided only if the outgoing ISDN-PRI trunk group is administered to send the SID (G1.1) or CPN (G3i) to the network. On calls incoming to a system, the network may provide either the SID (G1.1) or CPN (G3i) or ANI (G1.1) or BN (G3i) 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 0-15 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, discussed 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 15 characters
- Miscellaneous Call Identification — Maximum of eight characters
- Reason for Call Redirection — Maximum of two 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 (G1.1) or CPN/BN (G3i) to Host Call Identification

The SID/ANI (G1.1) or CPN/BN (G3i) to Host feature enables SID/ANI or CPN/BN information to be passed from the switch to the ISDN Gateway, so that the ISDN Gateway can forward the information to a host for data screen delivery to agents in an ACD split.

By delivering call identification information such as SID/ANI or CPN/BN 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.

The following figure shows a simplified diagram of a SID/ANI or CPN/BN 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.

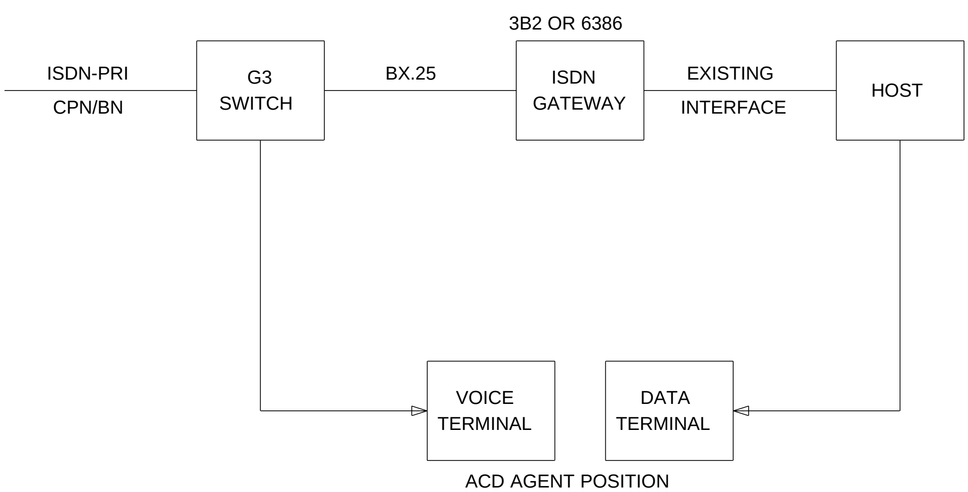


Figure 2-17. SID/ANI (G1.1) or CPN/BN (G3i) to Host Configuration

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 or Generic 3i 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 originates a call over the ISDN-PRI to the tandem switch.

AAR and 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 teh call that is originated 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, 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 in order to provide end-to-end signaling when different types of trunk facilities are used on a call. See Figure 2-14 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 CDR billing information on calls that are not Interworked. Accuracy of CDR 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.

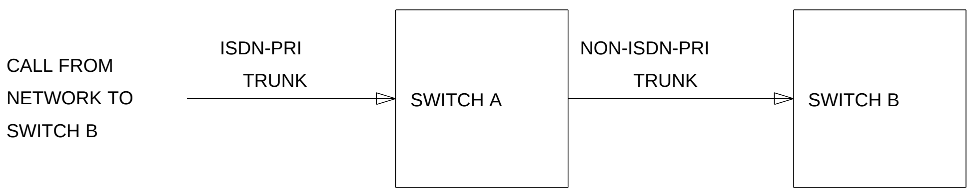


Figure 2-18. Interworking Example

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 is 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.

- DCS

  If both DCS and ISDN-PRI features are provided over the same facility with a DEFINITY Generic 1 or Generic 3i system, the ISDN-PRI display information is displayed in DCS format.

- FRL and TCM

  The TCM used to pass on the originating facility's 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.

- TEG

  On ISDN-PRI calls to a 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)
- Communication - Interface Link
- Communication - Interface Processor
- ISDN-PRI Trunk Group
- GRS Routing Patterns
- Signaling Group (G3i) (See the Facility and Non-Facility Associated Signaling feature description elsewhere in this manual

The following administration items are required for ISDN-PRI Call Identification Display:

- A Direct Distance Dialing SID (G1.1) or CPN (G3i) Prefix Table which 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 (G1.1) or CPN (G3i) Prefix — Used to create a 0-15 digit DDD number for an ISDN-PRI SID (G1.1) or CPN (G3i). The SID or CPN prefix can be from 0 through 10 digits in length. 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 TEG).

The following administration is required for SID/ANI (G1.1) or CPN/BN (G3i) to Host Call Identification:

- SID/ANI (G1.1) or CPN/BN (G3i) and 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 or CPN/BN to Host on a call-by-call basis. The choices for each trunk group are: SID (G1.1) or CPN (G3i) only, ANI

(G1.1) or BN (G3i) only, prefer SID or CPN but accept ANI or BN, prefer ANI or BN but accept SID or CPN, and blank.

Hardware and Software Requirements

A TN767 DS1 circuit pack (TN464B/C/D support A-law) is required for assignment of a T1 signaling link and up to 24 ISDN-PRI Trunk Group members. The DS1 provides 24 ports. A TN768 or TN780 (G3i) 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.

A TN464B DS1 circuit pack is required for assignment of an E1 signaling link and up to 31 ISDN-PRI Trunk Group members. The DS1 provides 31 ports. A TN768 or TN780 (G3i) 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.

A TN464C or later DS1 circuit pack is required for assignment of a T1 or E1 signaling link and up to 24 or 31 ISDN-PRI Trunk Group members. The DS1 provides 24 or 31 ports. A TN768 or TN780 (G3i) 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 (G1.1) or CPN/BN (G3i) to Host Call Identification 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.

  If a voice terminal extension is assigned a 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 DID and incoming Private Network Access calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations. The System Manager selects and records the message.

  Toll charges do not apply to DID and Private Network Access calls routed to Recorded Announcement.

- Intercept Treatment — Attendant

  Allows attendants to provide information and assistance to callers on all DID or incoming Private Network Access calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations. Normal toll charges apply to these calls.

- Intercept Treatment — Station

  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 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 DID 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

Sixty-four (G1.1) or 128 (G3i) 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. DID calls cannot be assigned Intercept Treatment — Tone.

Administration

The Intercept Tone is standard and requires no administration. However, administration is required to determine whether DID and Private Network Access calls are routed to the attendant or to an announcement. With DEFINITY Generic 1 and Generic 3i, 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. Intervals are administrable with the system parameters form.

Hardware and Software Requirements

Requires announcement equipment and one port on an Analog Line circuit pack for each announcement. With DEFINITY Generic 1 and Generic 3i, a TN750 Announcement circuit pack can be used to provide 64 (G1.1) or 128 (G3i) 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 one- or two-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 one- or two-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 1,024 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.

- 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.

- 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

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 were 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 DEFINITY Generic 1 or Generic 3i 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 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 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 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)

  CAS and Inter-PBX attendant calling cannot be enable 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

Description

Allows 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 and may be activated using Call Coverage or Call Forwarding All Calls. Interflow uses the Call Forwarding All Calls feature to redirect ACD calls to an external location.

Intraflow allows splits to be assigned coverage paths or forwarded. 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. Additionally, Call Forwarding can be used to unconditionally intraflow a split's calls.

An ACD call is intraflowed to another split whenever it is forwarded or the assigned Call Coverage redirection criteria is met. For a detailed description of Call Coverage redirection criteria, see the Call Coverage feature description in this chapter. For a detailed description of Call Forwarding, see the Call Forwarding feature description 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 one of the following conditions exists:

- The destination split's ''inflow threshold'' is met
- The queue is full
- There are no agents logged in
- All the logged-in agents are in the AUX work mode

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 (zero to 999 seconds), then ACD calls cannot be intraflowed into that split. If an ACD call is forwarded or 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 via Call Coverage from that split to another 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 (via Call Coverage) to a covering split is never connected to the first delay announcement assigned to the covering split. Calls redirected via Call Forwarding will receive a delay first announcement at the forwarded-to split. (A forced first announcement is never delivered at the destination split.)

As an illustration of how Intraflow works, assume the following:

- A call is intraflowed from split 1 to split 2 via Call Coverage.
- 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. Calls can be forwarded to destinations off the PBX (that is, domestic and international phone numbers on the public-switched telephone network). Calls cannot conditionally interflow. If a coverage point station or split is forwarded/interflowed, it is taken out of the coverage path. For details on how calls are forwarded to an off-premises extension, see the Call Forwarding All Calls feature description in this chapter.

A Coverage 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.

With G3i, more advanced Interflow capabilities are supported by the Call Vectoring and Look Ahead Interflow features discussed in this chapter.

Considerations

Intraflow and Interflow provide the means to redirect ACD calls to alternate splits. Intraflow via Call Coverage provides for conditional redirection of ACD calls when certain conditions are met (such as Busy or Don't Answer); Call Forwarding provides unconditional redirection. 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 the same location or another location.

The inflow threshold associated with Intraflow can be from zero 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.

- ACD

  When Intraflow via Coverage is provided, the Coverage Don't Answer Interval (one to 99 maximum ringing cycles) associated with Call Coverage begins 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.

- 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 (do not use "all" or "send all calls" as the redirection criterion).

- Don't Answer Interval and Don't Answer Interval for Subsequent

Redirection

The Don't Answer Interval specifies the number of ringing cycles that occur while the call is in queue or, if the call has been sent to an agent, the number of ringing cycles heard at the agent's terminal before the call is redirected to the first coverage point. This interval can be administered from one 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 one to 99 rings. This interval is administered as a system parameter.

NOTE:
A ringing cycle is typically 5.2 seconds but can be changed with administration.

- Whether or not each split has priority on Intraflow
- Inflow threshold
- Coverage ICI buttons as required
- All other 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 24 digits 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

  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 COR, 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 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 DDC group, a UCD group, an ACD split, a TEG, and a 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.

Messages are retrivable with the Voice Message Retrieval 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

LWC 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 2,000 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 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 LWC feature:

- AUDIX Interface

  LWC Cancel cannot be used to cancel an AUDIX message.

- Bridged Call Appearance

  A LWC 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 LWC, 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.

- Centralized Attendant Service (CAS)

  LWC Message Retrieval does not work with CAS.

- Conference

  A member of a conference call cannot activate LWC because that user cannot be uniquely identified.

  After LWC 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

LWC is administered by the System Manager. The following items require administration:

- 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 80 allowed per extension, including an extension number for a DDC group, UCD group, TEG, and PCOL group; 80 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. These intervals are administrable.

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.

Look Ahead Interflow (G3i)

Description

Provides flexible and intelligent ACD load balancing based on the Call Vectoring feature. Look Ahead Interflow enhances Call Vectoring interflow by ensuring that calls do not interflow to a backup switch that cannot accept the calls.

A Look Ahead Interflow call is attempted when a **route to vector** or **route to number** command successfully seizes an ISDN-PRI trunk. A vector on the receiving switch then either accepts or denies the Look Ahead Interflow call attempt. The sending switch does not relinquish control of the call until it is accepted by the receiving switch. Until the call is accepted, the caller continues to hear any tones applied by the sending switch vector and the call remains in any switch queues.

If the call is accepted, the call is removed from any queues at the sending switch and control of the call is passed to the receiving switch.

If the call is denied, vector processing simply continues at the sending switch. Audible feedback and the call's position in any queues at the sending switch remains unaltered so the caller is unaware that a Look Ahead Interflow call attempt has been made. The call vector may then apply alternate treatment, which may include placing another Look Ahead Interflow call to an alternate backup switch.

Since Look Ahead Interflow uses the Call Vectoring feature, it is recommended that the Call Vectoring feature description, discussed elsewhere in this chapter, be read and understood before continuing with this description. This should make the Look Ahead Interflow description easier to comprehend.

Look Ahead Interflow is provided by using private network ISDN-PRI connections. The sending switch generates an ISDN-PRI Look Ahead Interflow Information Element that is included with the ISDN SETUP message when the call routes on an ISDN facility. This information element contains interflow-related data, including DNIS information. The DNIS name may then be displayed if the call goes to an agent on the receiving switch.

Look Ahead Interflow Basics

When one switch has an overload of incoming calls, it may become necessary to route some of the incoming calls to another switch so that they can be handled more efficiently and will not be lost. Look Ahead Interflow is simply the means used to determine whether the other switch is able to handle the calls. When preset thresholds at one switch are reached (for example, the number of calls in queue), and another call comes in, that switch checks to see if another switch can handle the call. The other switch then checks to make sure it can handle the call. If it can, the call is sent to that switch. If it cannot, the sending switch must try to process the call in another way, such as intraflowing to a backup split or placing a second Look Ahead Interflow call attempt to an alternate backup switch.

Look Ahead Interflow is accomplished through the use of call vectors and their associated commands. These call vectors are administered for both the sending and receiving switches:

- Sending Switch

  Vectors at the sending switch use conditional **go to** vector commands to test outflow threshold conditions, and **route to** commands to send the call to another switch. The sending switch may provide alternate treatment if the call is denied at the receiving switch.

- Receiving Switch

  Vectors at the receiving switch use conditional **go to** vector commands to do inflow checking and decide whether the call should be accepted or denied. Call acceptance is accomplished when commands such as **queue-to-main**, **check-backup**, **announcement**, **collect**, and **wait** are reached in the call vector at the receiving switch. (See Table 2-33 for more information on acceptance conditions.) Call denial is accomplished when commands such as **busy** and **disconnect after announcement none** are reached in the call vector at the receiving switch.

Two-Switch Look Ahead Interflow Configuration

An example of a Two-Switch Look Ahead Interflow configuration is shown in the following figure. The operation of the sending switch and the receiving switch are described in the following paragraphs.

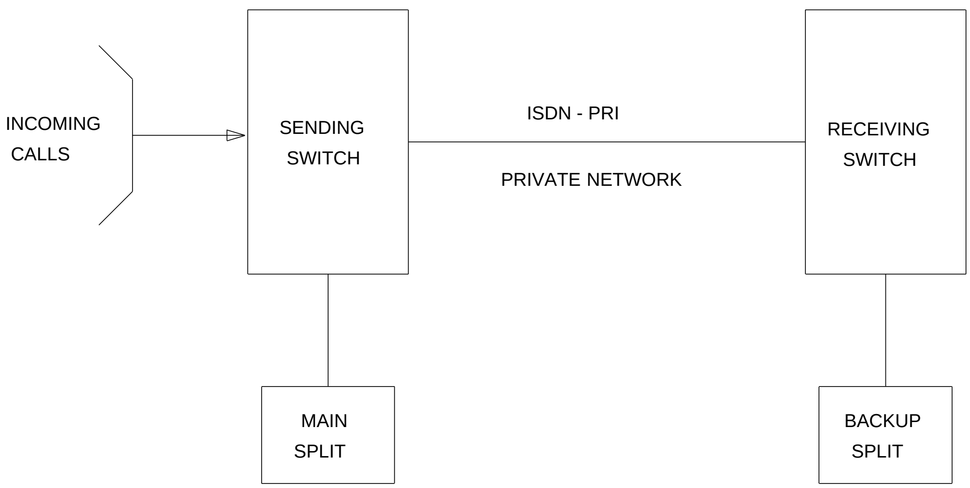


Figure 2-19. Two-Switch Look Ahead Interflow Connections

Sending Switch Operation

As with standard vectoring interflow, outflow checking and outflow is accomplished by means of conditional **go to** and **route to** vector commands. There are no new vector commands for this feature. If the Look Ahead Interflow option is enabled, and the call is being routed over an ISDN PRI facility, interflow will automatically be carried out on a look ahead basis. (There is one exception to this rule. A **route-to digits with coverage y** vector command will never result in a Look Ahead Interflow call.)

For Look Ahead Interflow calls, vector processing does not immediately terminate when ISDN-PRI facilities are successfully seized for a route-to operation. Instead, the call remains in any hunt group or split queues at the sending switch until it is accepted at the receiving switch. Any audible feedback initiated by the vector continues. If an agent becomes available at the sending switch during a Look Ahead Interflow call attempt, the caller is immediately connected to the available agent, and the Look Ahead Interflow call attempt is dropped.

The sending switch attempts to interflow the call to the receiving switch. We will assume a successful ISDN-PRI connection. One of the following then occurs:

- Call Acceptance

  If a call acceptance message is returned by the receiving switch before any call denial message, the sending switch will terminate vector processing, disconnect any tones applied by the sending switch vector and remove the call from all queues at the sending switch. Control of the call is now passed to the receiving switch.

- Call Denial

  If a call denial message is returned by the receiving switch before any call acceptance message, the sending switch will drop the Look Ahead Interflow call attempt and continue vector processing at the next vector step.

- Timeout

  If a call acceptance or call denial message is not returned from the receiving switch within 10 seconds after the receiving switch receives the Look Ahead Interflow call request, the Look Ahead Interflow attempt is dropped and the sending switch continues vector processing.

An example of an outflow vector used by a sending switch is as follows:

1. Queue to main split 3 priority m
2. Announcement extension 1001
3. Go to Step 5 if oldest-call-waiting in split 3 is > 30
4. Wait 20 secs hearing music
5. Route to number x if unconditionally
6. Announcement extension 1002
7. Wait 20 secs hearing music

If split 3 has no available agents, Step 1 will place the caller in split 3's queue at medium priority. In Step 2, an announcement is played apologizing for the delay. Step 3 does the outflow checking. If calls have been queued up for longer than 30 seconds, the vector goes to Step 5 and does Look Ahead Interflow. Otherwise, the vector proceeds to Step 4 and music is played for 20 seconds. If the call is still not answered after 20 seconds, then the vector goes to Step 5 and attempts Look Ahead Interflow. In Step 5, a Look Ahead Interflow call is placed to a remote switch. If the call is accepted, the incoming call is removed from split 3's queue and control of the call is passed to the receiving switch. If the call is denied, the caller remains in queue and hears announcement 1002 followed by music.

Receiving Switch Operation

When the receiving switch receives the Look Ahead Interflow request, the call routes to a VDN, the VDN maps the call to the receiving switch's inflow vector and vector processing begins, starting with inflow checking. Inflow checking is accomplished with conditional **go to** commands in the inflow vector. The decision to accept or reject a call can be based on checks of the following:

- Number of staffed agents
- Number of available agents
- Time of day/week
- Number of calls in split's queue
- Number of seconds that the oldest call has been waiting in the split's queue

Once inflow checking is complete, acceptance of the look ahead call can be accomplished by executing any of the vector commands shown in the following table.

Table 2-33. Call Acceptance Vector Commands and Qualifications

Call Acceptance Vector Commands	Qualification
Announcement	Announcement available OR Queued for announcement OR Retrying announcement
Check Backup	Call terminates to agent OR Call queued for split
Collect	Always
Disconnect	With announcement and announcement available OR With announcement and queued for announcement OR With announcement and retrying announcement
Messaging	Successful OR Queued
Queue to Main	Call terminates to agent OR Call queued for split
Route	Terminates to valid local destination OR Successfully seizes a non-PRI trunk OR Results in Look Ahead Interflow call attempt and the call is accepted by the far end switch
Wait	Always

If, during inflow checking, the receiving switch decides that it is unable to accept the look ahead call, call denial can be accomplished by executing any of the vector commands listed in the following table.

Table 2-34. Call Denial Vector Commands and Qualifications

Call Denial Vector Commands	Qualification
Busy	Always
Disconnect	With no announcement OR With announcement but announcement unavailable

The vector commands shown in the following table do not generate either call acceptance or denial messages and are considered neutral. Use of the **busy** command is recommended over the **disconnect** command.

Table 2-35. Neutral Vector Commands and Qualifications

Neutral Vector Commands	Qualification
Adjunct Routing	Always
Announcement	Announcement unavailable
Check Backup	The call neither terminates nor queues
Goto Step	Always
Goto Vector	Always
Messaging	Failure
Queue to Main	The call neither terminates nor queues
Route	Unsuccessful termination OR Trunk not seized OR Look Ahead Interflow call denied by far end switch
Stop	Always

An example of an inflow vector used by a receiving switch is as follows:

1. Go to Step 7 if staffed-agents in split 12 is < 3
2. Go to Step 7 if queued-calls in split 12 is > 4
3. Queue to main split 12 priority h
4. Go to Step 6 if queued-calls in split 12 < 1
5. Wait 30 secs hearing music
6. Stop
7. Disconnect after announcement extension none

Steps 1 and 2 do inflow checking. For split 12, if the number of staffed agents is less than three or the number of calls in queue is greater than 4, Step 4 checks for the condition of all agents in Aux-Work. If all agents are in Aux-Work, Step 3 fails. No calls can be in queue, so Step 4 succeeds. Step 7 will deny the look ahead call. Otherwise, Step 3 will queue the call for split 12 and return a call acceptance message to the sending switch.

Tandem Switch Configuration

Tandem Look Ahead Interflow can be accomplished with Call Vectoring and Look Ahead Interflow active at the receiving switch by using **route to** commands that contain external destinations which use ISDN-PRI facilities.

An example of a tandem Look Ahead Interflow configuration is shown in the following figure. The operation of the sending switch and the receiving switch are described in the following paragraphs.

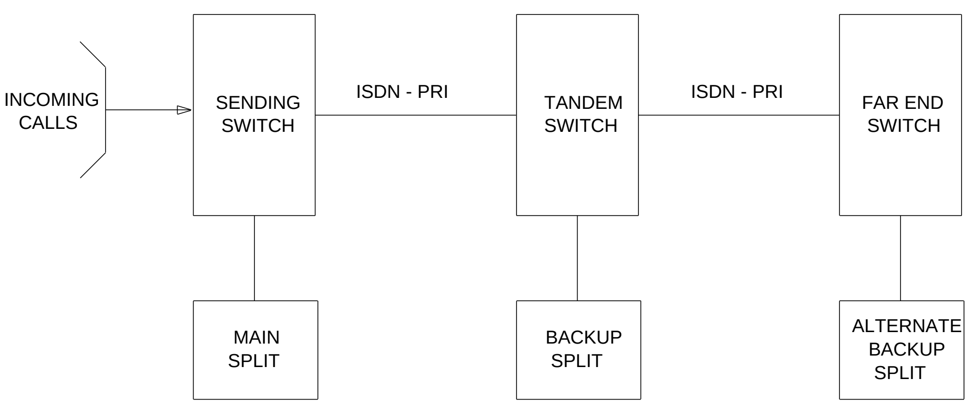


Figure 2-20. Look Ahead Interflow Using a Tandem Switch

Sending Switch Operation

The sending switch is unaware of the fact that its Look Ahead Interflow call is being tandemed to an alternate switch. The operation of the sending switch in the tandeming configuration is exactly as for the two switch configuration.

Tandem Switch Operation

If the receiving switch executes a **route to** command that routes the call over an ISDN facility before call acceptance, the **route to** command is performed on a ‘‘look ahead’’ basis in the same manner as a sending switch. If the call is accepted at the far end switch, acceptance is passed to the sending switch, and control of the call is passed to the far end switch with tandeming of the original calling party information and the original DNIS name. If the call is denied, the next step of the tandem switch vector is executed.

Great care should be taken by the vector programmer to ensure that the sending switch is not used as a backup location for the tandem switch or any of the far end switches. If administered in this manner, all trunk facilities could be tied up by a single call.

Example of Tandem Switch Vector

An example of a tandem switch vector follows:

1. go to Step 3 if staffed-agents in split 30 is < 1
2. go to Step 5 if oldest call-waiting in split 30 is < 6
3. route to number x if unconditionally
4. busy
5. queue to main split 30 priority m
6. go to Step 3 if calls queued in split 30 < 1
7. wait 10 secs hearing silence
8. announcement extension 2300
9. go to Step 7 if unconditionally

Steps 1 and 2 check the inflow thresholds. If the inflow criteria are acceptable, Step 5 (Queue-to Main) will provide acceptance to the sending switch. Step 6 checks for the condition of all agents in Aux-work. No ringback is provided in this example since the sending switch has already returned an announcement before look ahead interflow to the receiving switch. If the call cannot be handled at split 30, the **route to** command checks another Look Ahead Interflow equipped switch on a ‘‘look ahead’’ basis. If the far end switch rejects the call, the Forced Disconnect causes a denial to be sent to the sending switch. If the far-end switch accepts the call, then the acceptance is relayed back to the sending switch.

Far End Switch Operation

The far end switch is also unaware that tandeming has taken place. The far end switch will operate the same as in the case of the receiving switch in the two-switch configuration.

Display Information

Answering Agent’s Display

The DNIS information in the Look Ahead Interflow information element provided by ISDN-PRI is presented on the answering agent’s display on the receiving switch if the Look Ahead Interflow option is enabled, the call routes to a VDN, and the DNIS name is not blank.

The DNIS name included in the Look Ahead Interflow information element is as follows:

- If this is a tandemed look ahead call, then the Look Ahead Interflow DNIS information from the original look ahead call is used.

- If no redirection has taken place at the sending switch, the VDN name according to display override rules are used.
- If redirection has occurred, display override rules do not take effect. The Look Ahead Interflow DNIS will contain the original VDN name, or, if multiple VDNs are accessed, the name of the VDN last accessed by means of a **route to** vector command.

VDNs that map to vectors which place Look Ahead Interflow calls must have their ISDN CPN prefixes administered. Failure to administer the ISDN CPN prefix results in a Look Ahead Interflow DNIS of all spaces being sent and displayed on the answering agent's terminal.

Originator's Display

On internal calls, the originator's display is the same as for normal Call Vectoring. However, the following precaution should be taken to prevent undesirable display updates from being received during Look Ahead Interflow call attempts.

Since most customers would not like the originator's display to be updated on each Look Ahead Interflow call attempt, Look Ahead Interflow calls should normally go out over trunk groups with the Outgoing Display field set to ''no''.

Audible Feedback

Audible ringback is provided to the caller when a **wait with ringback** vector command is executed or when the call is successfully routed to a local destination.

Care must be taken by the vector programmer not to confuse callers by providing incompatible audible feedback at both the outflow and inflow vectors. For example, providing ringback at the receiving switch might be confusing to the caller if ringback and an announcement had previously been supplied by the sending switch.

Normally a vector for human callers should initially return audible feedback such as ringback, music, or an announcement so that the caller knows that the call got through. However, vectors which do not return any audible feedback may be used in situations where VDNs are accessed by machines and audible feedback may be inappropriate and/or undesirable.

The receiving switch vector will not give look ahead calls any locally denied termination treatment such as busy tone, reorder tone, or intercept tone.

Considerations

The system supports a maximum of 256 vectors and 500 VDNs.

The sending/receiving vectors should not be administered to have calls route back to the outflowing switch. If administered as such, all trunk facilities may eventually be tied up with the same call.

All calls routed over ISDN-PRI facilities by means of **route to number** and **route to digits with coverage n** vector commands on a switch where the Look Ahead Interflow option is enabled are treated as Look Ahead Interflow call attempts.

Initial audible feedback may be provided to the caller before interflow is attempted. Therefore, another audible feedback from the receiving switch may not be appropriate. For example, a caller hearing ringback on the sending switch may be confused if music is suddenly applied when the call interflows to the receiving switch.

Delay in interflowing should be minimized. An acceptance or denial response should be provided to the sending switch as quickly as possible by the vectors on the receiving switch.

If, during Look Ahead Interflow, the call terminates to an agent on the sending switch or the call is abandoned by the originator, the Look Ahead Interflow call is dropped, vector processing terminates, and the original call is removed from all split queues.

It is possible during a Look Ahead Interflow call attempt for a call to be accepted at a receiving switch, by means of a **queue to main** or **check backup** command, an instant before the call is answered at the sending switch. If the acceptance message is delayed due to signal propagation delay, there could be a short interval when the caller and the receiving switch agent are connected. An agent at the sending switch may then answer the call before the acceptance message arrives at the sending switch. The caller would then be disconnected from the receiving switch agent and connected to the sending switch agent. These phantom calls can be eliminated by appropriate programming of the inflow vector. If calls are accepted by **wait** or **announcement** vector commands before the call is queued to a split, there is no possibility of a phantom call occurring.

It is perfectly acceptable for a vector to route a call over an ISDN-PRI facility to a destination which is not a VDN. As far as the sending switch is concerned, this call will be treated as a Look Ahead Interflow call even though this is not in fact the case. Generic ISDN processing at the receiving switch will cause the call to be accepted. The DNIS name and any other information in the Look Ahead Interflow information element will be ignored.

If a Look Ahead Interflow call terminates to a VDN on a receiving switch where the Look Ahead Interflow option is not enabled, intelligent interflow will result. However, the DNIS information in the Look Ahead Interflow information element will be ignored and no intelligent interflow to far-end switches is possible.

Interactions

The following features interact with the Look Ahead Interflow feature:

- Attendant Control of Trunk Group Access

  Calls will not route over a trunk with Attendant Control of Trunk Group Access set.

- Authorization Codes

  Authorization Codes must not be required for interflow routing. The FRL assigned to the VDN should be high enough so that the route desired for routing interflow calls can be used without requiring an Authorization Code entry. If a route choice is encountered that requires a higher FRL, the interflow is considered an invalid destination (rejected for Look Ahead Interflow or not available for standard interflow) without the application of recall dial tone. Vector processing will continue.

- AAR

  ISDN-PRI facilities used to provide Look Ahead Interflow to a VDN on another switch in the customer's network can use the AAR feature if private facilities are to be used for call routing.

- ARS

  ISDN-PRI facilities used to provide Look Ahead Interflow to a VDN on another switch in the customer's network can use the ARS feature for call routing.

- ACD

  Look Ahead Interflow calls can be interflowed directly to ACD splits on remote switches which do not have Call Vectoring or Look Ahead Interflow enabled. In this case, the calls will be accepted or denied by generic call processing, even though full look ahead functionality will be lost.

- Call Vectoring

  Call vectoring is required at the sending switch and the receiving switch.

  Call vectoring operates differently if Look Ahead Interflow is enabled at the sending switch. If ISDN-PRI facilities are successfully seized for a route-to operation, vector processing does not terminate until the call is accepted at the receiving switch.

  A forced disconnect at the receiving switch will not cause the originator's call to be dropped, but will only result in the failure of the Look Ahead Interflow attempt.

- CAS

  A centralized attendant can be a Look Ahead Interflow destination.

- DID

  The Look Ahead Interflow routing over ISDN PRI facilities can enter the receiving switch on a ''DID'' basis. That is, a destination extension address (a VDN) is included in the ISDN SETUP message in the Called Number information element.

- FRL and TCMs

  The FRL for interflow over ARS/AAR route choices is assigned to the original VDN used for the incoming call.

- Incoming Call Management

  The adjunct routing capabilities of vectoring can be used at the sending switch to determine if a call should be interflowed. Adjunct routing at the receiving switch can be used to tandem the call to a far-end switch.

- ISDN-PRI

  ISDN-PRI connectivity end-to-end over a private network is required for Look Ahead Interflow.

- Intercept Treatment

  No intercept treatment should be applied toward the caller as part of Look Ahead Interflow operation unless it results from a tandem connection on a non-Look Ahead Interflow basis. If the Look Ahead Interflow results in an intercept condition, the interflow should be rejected as part of the interflow vector programming, with alternate treatment provided at the sending switch.

- CDR (Sending Switch)

  No Ineffective Call Attempt or Outgoing Call CDR records will be generated for Call Vectoring **route to** commands which are unsuccessful including denied Look Ahead Interflow attempts.

  If a local (on-switch) call to a VDN generates a Look Ahead Interflow call attempt that is accepted, and answer supervision is returned from the receiving switch, then one Outgoing Call CDR record is generated with the originating extension as the calling number.

  If an incoming (off-switch) call to a VDN generates a Look Ahead Interflow call attempt that is accepted, and no answer supervision is returned from the receiving switch, then one incoming CDR record will be generated. The VDN is the called number, and the duration is from the time answer supervision was provided to the incoming trunk.

  If an incoming (off-switch) call to a VDN generates a Look Ahead Interflow call attempt that is accepted, and answer supervision is returned from the receiving switch, then two incoming CDR records will be generated:

  1. An incoming record with the VDN as the called number and the duration as the time since answer supervision was provided to the incoming trunk.
  2. An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.

  The sending of a Look Ahead Interflow information element is counted toward Message Associated User-to-User Information (MA-UUI) counts.

- CDR (Receiving Switch)

  On the receiving switch, an incoming Look Ahead Interflow call is treated like any other incoming vector call as far as CDR is concerned. An incoming CDR record is recorded on all calls which return answer supervision. The duration of the call is the time since answer supervision was returned.

  If answer supervision is returned by the vector (via an announcement, collect, disconnect, or wait with music command), and the call is never terminated to another destination, then the VDN extension is recorded as the called number in the CDR record.

  If the call terminates to a hunt group, then the VDN, hunt group, or agent extension is recorded as the called number. If the ‘‘Record VDN in Record’’ option of the Feature Related System Parameters is administered as ‘‘y,’’ then the VDN extension will override the ‘‘Call to Hunt Group - Record’’ administration option for Vector Calls.

  If the call terminates to a trunk (tandem), then two CDR records will be generated:

  1. An incoming record with the VDN as the called number and the duration as the time since answer supervision was provided to the incoming trunk.
  2. An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.

- Trunk-to-Trunk Transfer

  Interflowed calls may be transferred by a receiving switch agent to another trunk connection.

Administration

For full functionality, this feature requires the ISDN PRI, Call Vectoring and Look Ahead Interflow options to be enabled at both the sending and receiving switches.

Call Vectoring and ISDN-PRI must be administered as described in their separate feature descriptions discussed elsewhere in this chapter.

Hardware and Software Requirements

Existing ISDN-PRI hardware can be used for ISDN connectivity to the receiving switch. No new hardware is required. Interconnecting facilities must be ISDN-PRI with no interworking for the full capabilities of the feature to be operational. Look Ahead Interflow calls which interwork may interflow successfully but the ability to do so on an intelligent basis will be lost as will the Look Ahead Interflow DNIS information.

Sending and receiving switches must support the ISDN-PRI, Call Vectoring, and Look Ahead Interflow features.

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. The trunk access codes are used to activate Loudspeaker Paging Access. A user can activate Loudspeaker Paging Access by dialing the trunk access code of the desired paging zone. In addition, the trunk access codes 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 trunk access code 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:

- 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 trunk access code 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 one- or two-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.

An LDN or DID 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 two-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 screen must be administered as `no`.
- Up to 10 (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).
- CDR activation.

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).

If a PagePac paging system is accessed through a CO or analog trunk, administration is not done through the Loudspeaker Paging Access form. Instead, the line is accessed as a standard trunk (trunk access code of a CO) or a standard extension (dialed extension that connects to PagePac).

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) for each individual 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 (TN746B supports A-law), TN769, or TN746B, Analog Line circuit pack, or TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) is required (depending on which PagePac system is used).

No additional software is required.

Loudspeaker Paging Access — Deluxe

Description

Provides attendants and voice terminal users with integrated dial access to voice paging equipment and Call Park capabilities.

When Loudspeaker 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 Time-out 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 (discussed elsewhere in this document) or Loudspeaker Paging Access — Deluxe (Deluxe Paging). This description discusses 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. The trunk access codes are used to activate Deluxe Paging. A user can activate Deluxe Paging by dialing the trunk access code of the desired paging zone. In addition, the trunk access codes 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 trunk access code 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:

- 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 trunk access code 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 one- or two-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 Time-out 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 time-out 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 Time-out 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 unanswered, the call follows the coverage path of the paging user.

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 Time-out 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 time-out 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 Time-out 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 unanswered, the call follows the coverage path of the paging user.

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 Time-out interval starts. If the Split button is pressed, the timer for the Call Park Time-out 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 time-out 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.

An LDN or DID 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, and so on) 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 TEG, a shared PCOL, 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 10 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 Time-out 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. If unanswered, the call follows the coverage path of the paging user.

- 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 two-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 conferee 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.

- 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.

- LWC

  If a user parks a call for his or her extension with the Conference button, any parking or parked parties cannot activate LWC 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 LDNs

  An LDN 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.

- 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 COR from doing so.

- Ringback Queuing

  Ringback Queuing is not provided for Deluxe Paging.

- 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 one to 90 minutes in intervals of one minute).
- CDR activation.
- Console permissions to allow voice terminal extensions to park calls on common shared extensions (assigned via COS).
- 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 (TN763D supports A-law) for each individual 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 or TN746B (A-law) Analog Line circuit pack, or TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) 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 COR. 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 two-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 two 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

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 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 and combined conversion resources 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 cabled to a 212 Modem. This integrated type is not available for countries that use A law. The combined type is a Trunk Data Module cabled to any Trunk Data Module-compatible modem to provide a conversion resource. Combined type applies to all system independent of system companding.

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.

DEFINITY Generic 1 and Generic 3i provide 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).

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 switched network modems supporting asynchronous and synchronous communications, and autobaud at 300, 1200, or 2400 bps.
- DEFINITY Generic 1 and Generic 3i can operate up to 19.2 kbps.
- Different pools 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, DEFINITY Generic 1 and Generic 3i accumulate data on modem pooling calls separate from voice calls. Measurements on the pools are also accumulated.

DEFINITY Generic 1 and Generic 3i can support up to five pools; all five combined, integrated, or any mix. Each pool has a capacity of up to 32 conversion resources. DEFINITY Generic 1 and Generic 3i have a limit of 160 conversion resources for all five 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 Trunk Data Module 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 Trunk Data Module bus blocking characteristics.

Tandem Switches will not insert a pooled modem. It is the responsibility of the originating switch to do so.

Interactions

The following features interact with the Modem Pooling feature:

- Data Call Setup

  Data calls to or from a Trunk Data Module 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 Trunk Data Module.

- Data Privacy and Data Restriction

  With G1.1 or G3i, the insertion of a modem pool does not turn off Data Privacy and/or Data Restriction.

- 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

  Conversion resources used for Modem Pooling can only be connected to AVD DS1 tie trunks via Data Terminal Dialing or by dialing the feature access code for data origination.

- CDR

  With G3i, Data Call CDR records the use of modem pools on trunk calls.

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 Trunk Data Module and associated modems ports, speed (up to three speeds), and duplex and synchronization characteristics.
- HOLD Time (per pool basis) — Specify the maximum time (one to 99 minutes) any conversion resource may be held and not used (while a data call waits in a queue). Default value is five 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. 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

Description

Allows a CMS user to move agents (that are not currently logged into a split) 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 DEFINITY Manager I terminal (G1) or the G3 Management 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 ACW button is a split-associated button. If an agent is assigned an ACW button for one split, and is moved to another split via the CMS screen, the ACW 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)
- 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 member of the old split.
- The agent is logged into the old split.
- The agent is already a member of the new split.
- A Move Agents, Move Vector Directory Numbers (G3i), or Change Vector (G3i) request is being made from another CMS terminal.

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 individual 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 Manager I terminal (G1) or the G3 Management Terminal, 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 members of splits 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 Manager I terminal (G1) or the G3 Management Terminal.

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.

Move agents, Move Vector Directory Numbers (G3i), or Change Vector (G3i) requests from other CMS terminals will fail if one CMS terminal's Move Agents request is in progress.

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. 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 five 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 five 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 three.

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.

Multi-Language Displays

Description

The Multi-Language Display feature allows 40-character display station users or an attendant user to select a display message language. This language selection is made via administration; the choice of languages is English (default), French, Italian, or Spanish. Note that the messages themselves do not change but the language used to present them does.

Considerations

For G3i-Global, the Multi-Language Display feature will only apply to sets that have at least a 40-character display. Displays will only support upper and lower-case English letters, punctuation, and digits. Diacritical marks will not be supported in this release.

One user at a location can have a set administered for the Italian language and another can see English displays, if desired.

Junior attendant consoles are not supported by G3i-Global software and therefore, their displays will not be translated.

Users of a 32-character display set will not have the option of choosing a display language. These sets (in particular the 7315H and the7317H) will default to English.

There are a set of messages that appear on a MFDT/ISDN-BRI Station/Attendant users set and provide call related information. These messages (known as the "display message set") have been translated into three additional languages: Italian, Spanish, and French. The message set and the format in which it appears on the display will not be changed (only the wording that is used to present the message will be changed).

Each MFDT/ISDN-BRI user and each Attendant can select the language of their choice. Language selection is made via administration (English is the default language). Once the language is selected and administered on the station/attendant form, all display messages (except those that are administered) will be in the language selected.

Interactions

Several features display messages; see the tables below for a complete listing.

Administration

The attendant form will be modified to ask for the type of display language. The station form will be modified to ask for the type of display language only if that station is equipped with a 40-character display module. Both forms will allow four options: English, French, Italian, and Spanish, with English being the default.

Hardware and Software Requirements

None required.

Feature Displays

This section shows the displays related to each feature. Since none of the feature functionalities are modified, no explanation of the feature will be made. When the time is displayed, the Italian, French, and Spanish languages use the 24 hour clock. The English language uses the "am"/"pm" notation.

Automatic Wakeup

The following displays are associated with the Automatic Wakeup feature. Messages can be a maximum of 40 characters:

- *"AUTO WAKEUP - Ext: xxxxx Time: --:-- xM"* (English)
  - French - *"REVEIL AUTO. - POSTE: xxxxx HEURE: --:--"*
  - Italian - *"SERVIZIO SVEGLIA - Tel: xxxxx Ora: --:--"*
  - Spanish - *"DESPERT AUTOMA - EXT: xxxxx HORA: --:--"*
- *"INVALID EXTENSION - TRY AGAIN"* (English)
  - French - *"NUMERO DE POSTE EST ERRONE - REESSAYER"*
  - Italian - *"NUMERO ERRATO - RIPETERE"*
  - Spanish - *"EXTENSION NO VALIDO - INTENTE DE NUEVO"*
- *"WAKEUP ENTRY DENIED - INTERVAL FULL"* (English)
  - French - *"DEM. REVEIL REFUSEE - INTERVALLE PLEIN"*
  - Italian - *"SVEGLIA NON ATTIVATA - ORARIO OCCUP."*
  - Spanish - *"ENTRADA DENEGADA - INTERVALO COMPLETO"*
- *"WAKEUP ENTRY DENIED - NO PERMISSION"* (English)
  - French - *"DEM. REVEIL REFUSEE - SANS AUTORISATION"*
  - Italian - *"SVEGLIA NON ATTIVATA - NON PERMESSO."*
  - Spanish - *"ENTRADA DENEGADA - SIN PERMISO"*
- *"WAKEUP ENTRY DENIED - SYSTEM FULL"* (English)

  - French - *"DEM. REVEIL REFUSEE - ENCOMBREMENT"*
  - Italian - *"SVEGLIA NON ATTIVATA - CONGESTIONE"*
  - Spanish - *"ENTRADA DENEGADA - SISTEMA COMPLETO"*
- *"WAKEUP ENTRY DENIED - TOO SOON"* (English)
  - French - *"DEM. REVEIL REFUSEE - TROP TOT"*
  - Italian - *"SVEGLIA NON ATTIVATA - TROPPO PRESTO"*
  - Spanish - *"ENTRADA DENEGADA - MUY PRONTO"*
- *"WAKEUP REQUEST CANCELED"* (English)
  - French - *"DEMANDE DE REVEIL EST ANNULEE"*
  - Italian - *"RICHIESTA SVEGLIA CANCELLATA"*
  - Spanish - *"SOLICITUD DE DESPERTADOR CANCELADA"*
- *"WAKEUP REQUEST CONFIRMED"* (English)
  - French - *"DEMANDE DE REVEIL EST CONFIRMEE"*
  - Italian - *"RICHIESTA SVEGLIA CONFERMATA"*
  - Spanish - *"SOLICITUD DE DESPERTADOR CONFIRMADA"*
- *"Wakeup Call"* (English)
  - French - *"APPEL DE REVEIL"*
  - Italian - *"Serv. Sveglia"*
  - Spanish - *"LLAMADA DE DESPERTADOR"*

Busy Verification of Stations and Trunks

The following displays are associated with the Busy Verification of Stations and Trunks feature. Messages can be a maximum of 15 characters.

"English Display"	"French Display"	"Italian Display"	"Spanish Display"
"ALL MADE BUSY"	*"TOUS OCC."*	*"TUTTI OCCUPATI"*	*"TODAS OCUPADAS"*
"BRIDGED"	*"EN DERIVATION"*	*"OCCUPATO"*	*"PUENTEADA"*
"DENIED"	*"INTERDIT"*	*"NON PERMESSO"*	*"DENEGADO"*
"INVALID"	*"ERRONE"*	*"NON VALIDO"*	*"NO VALIDO"*
"NO MEMBER"	*"AUCUN MEMBRE"*	*"NESSUN ELEMENTO"*	*"NINGUN MIEMBRO"*
"OUT OF SERVICE"	*"HORS SERVICE"*	*"FUORI SERVIZIO"*	*"FUERA SERVICIO"*
"RESTRICTED"	*"RESTREINT"*	*"RISTRETTO"*	*"RESTRINGIDO"*
"TERMINATED"	*"TERMINE"*	*"TERMINATO"*	*"TERMINADO"*
"TRUNK SEIZED"	*"CIRCUIT SAISI"*	*"GIUNZIONE IMP."*	*"ENLACE OCUPADO"*
"VERIFIED"	*"VERIFIE"*	*"VERIFICATO"*	*"VERIFICADO"*

Call Appearance Designation

For each of the four languages (English, French, Italian, & Spanish), the display to indicate call appearance designation will appear as:

- *"a = "* (English)
  Call appearance buttons are designated on the display by a lower case letter (a-z for the first 26 call appearances then A-Z) in position 1, followed by an "=".

Call Progress Feedback

Following is a list of the different call progress displays. Messages can be a length of 8 characters:

"English Display" (stands for)	"French Display" (stands for)	"Italian Display" (stands for)	"Spanish Display" (stands for)
"busy" (Extension Busy, Intrusion Not Allowed, Call Waiting Not Allowed)	*"OCCUPE"* (Occupe)	*"occ"* (Occupato)	*"OCUPADA"* (Ocupada)
"busy(I)" Extension Busy, Intrusion Allowed, Call Waiting Not Allowed)	*"OCC.(E)"* (Entree ligne occupe)	*"occ(I)"* (Occupato-Intrusione)	*"OCUP(I)"* (Ocupada-intrusion)
"ringing" (Extension Ringing)	*"SONNE"* (Libre)	*"libero"* (Libero)	*"LIBRE"* (Libero)
"wait" (Extension Busy, Intrusion Not Allowed, Call Waiting Allowed)	*"ATTENTE"* (Attente)	*"auat"* (Autoattesa)	*"ESPERA"* (Espera)
"(I) wait" (Extension Busy, Intrusion Allowed, Call Waiting Allowed)	*"(E) ATTENTE"* (Entree ligne attente)	*"(I) auat"* (Intrusione-Autoattesa)	*"(I) ESPERA"* (Intrusion, en espera)

Class of Restriction

The following displays are associated with the Class of Restriction feature. Messages can be a length of 4 characters.

Restriction	"English Display"	"French Display"	"Italian Display"	"Spanish Display"
Toll	*"TOLL"*	*"INT."*	*"TASS"*	*"TARF"*
Full	*"FULL"*	*"COM."*	*"DISB"*	*"LLEN"*
No Calling Party	*"NONE"*	*"AUC."*	*"ABIL"*	*"NING"*
Origination	*"ORIG"*	*"DEP."*	*"ORIG"*	*"ORIG"*
Outward	*"OTWD"*	*"SOR."*	*"USCN"*	*"SALI"*

Date/Time Mode and Formats

The following displays are associated with the Date & Time feature.

If the time is not available (this message can be a length of 40 characters):

- *"SORRY, TIME UNAVAILABLE NOW"* (English)
  - French - *"HEURE ET DATE INDISPONIBLES"*
  - Italian - *"ORA E DATA TEMP. NON DISPONIBILI"*
  - Spanish - *"HORA Y FECHA NO DISPONIBLES AHORA"*

Date and Time information begins in position 3 and may extend as far as position 40.

— English -

```
<DATE/TIME>     <TIME><b><b><DATE>
<TIME>          <HR>:<MIN><b><M>
<HR>            1-12 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<M>             "am" or "pm"
<DATE>          <DOW><b><MONTH><b><DOM>,<b><YEAR>
<DOW>           Day of week, upper case, unabbreviated
<MONTH>         Month of year, upper case, unabbreviated
<DOM>           1-31 (day of month, no leading zeroes)
<YEAR>          Year in 4 digits
<b>             Blank
```

— French, Italian, & Spanish -

```
<DATE/TIME>     <TIME><b><b><DATE>
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<DATE>          <DOW><b><DOM><b><MONTH>,<b><YEAR>
<DOW>           Day of week, upper case, unabbreviated
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         Month of year, upper case, unabbreviated
<YEAR>          Year in 4 digits
<b>             Blank
```

Days of the Week

"English Display"	"French Display"	"Italian Display"	"Spanish Display"
"SUNDAY"	*"DIMANCHE"*	*"DOMENICA"*	*"DOMINGO"*
"MONDAY"	*"LUNDI"*	*"LUNEDI"*	*"LUNES"*
"TUESDAY"	*"MARDI"*	*"MARTEDI"*	*"MARTES"*
"WEDNESDAY"	*"MERCREDI"*	*"MERCOLEDI"*	*"MIERCOLES"*
"THURSDAY"	*"JEUDI"*	*"GIOVEDI"*	*"JUEVES"*
"FRIDAY"	*"VENDREDI"*	*"VENERDI"*	*"VIERNES"*
"SATURDAY"	*"SAMEDI"*	*"SABATO"*	*"SABADO"*

Months of the Year

"English Display"	"French Display"	"Italian Display"	"Spanish Display"
"JANUARY"	*"JANVIER"*	*"GENNAIO"*	*"ENERO"*
"FEBRUARY"	*"FEVRIER"*	*"FEBBRAIO"*	*"FEBRERO"*
"MARCH"	*"MARS"*	*"MARZO"*	*"MARZO"*
"APRIL"	*"AVRIL"*	*"APRILE"*	*"ABRIL"*
"MAY"	*"MAI"*	*"MAGGIO"*	*"MAYO"*
"JUNE"	*"JUIN"*	*"GIUGNO"*	*"JUNIO"*
"JULY"	*"JUILLET"*	*"LUGLIO"*	*"JULIO"*
"AUGUST"	*"AOUT"*	*"AGOSTO"*	*"AGOSTO"*
"SEPTEMBER"	*"SEPTEMBRE"*	*"SETTEMBRE"*	*"SEPTIEMBRE"*
"OCTOBER"	*"OCTOBRE"*	*"OTTOBRE"*	*"OCTUBRE"*
"NOVEMBER"	*"NOVEMBRE"*	*"NOVEMBRE"*	*"NOVIEMBRE"*
"DECEMBER"	*"DECEMBRE"*	*"DICEMBRE"*	*"DICIEMBRE"*

Do Not Disturb (Hotel/Motel feature)

The following displays are associated with the Do Not Disturb feature. Messages can be a maximum of 40 characters:

- *"DO NOT DIST - Group: xx   Time: --:-- xM"* (English)
  - — French - *"NE PAS DERANGER GROUPE: xx HEURE: --:--"*
  - — Italian - *"NON DISTURBARE - Grp: xx   Ora: --:--"*
  - — Spanish - *"NO MOLESTAR - GRUPO: xx   HORA: --:--"*
- *"DO NOT DIST - Ext: xxxxx Time: --:-- xM"* (English)
  - — French - *"NE PAS DERANGER POSTE:xxxxx HEURE: --:--"*
  - — Italian - *"NON DISTURBARE - Tel: xxxxx   Ora: --:--"*

  - Spanish - *"NO MOLESTAR - EXT: xxxxx   HORA: --:--"*
- *"DO NOT DIST ENTRY DENIED - INTERVAL FULL"* (English)
  - French - *"DEMANDE EST REFUSEE - INTERVALLE PLEIN"*
  - Italian - *"SERVIZIO NON ATTIVATO - ORARIO OCCUP."*
  - Spanish - *"ENTRADA DENEGADA - INTERVALO COMPLETO"*
- *"DO NOT DIST ENTRY DENIED - NO PERMISSION"* (English)
  - French - *"DEMANDE EST REFUSEE - SANS AUTORISATION"*
  - Italian - *"SERVIZIO NON ATTIVATO - NON PERMESSO."*
  - Spanish - *"ENTRADA DENEGADA - SIN PERMISO"*
- *"DO NOT DIST ENTRY DENIED - SYSTEM FULL"* (English)
  - French - *"DEMANDE EST REFUSEE - ENCOMBREMENT"*
  - Italian - *"SERVIZIO NON ATTIVATO - CONGESTIONE"*
  - Spanish - *"ENTRADA DENEGADA - SISTEMA COMPLETO"*
- *"DO NOT DIST ENTRY DENIED - TOO SOON"* (English)
  - French - *"DEMANDE EST REFUSEE - TROP TOT"*
  - Italian - *"SERVIZIO NON ATTIVATO - TROPPO PRESTO"*
  - Spanish - *"ENTRADA DENEGADA - MUY PRONTO"*
- *"INVALID GROUP - TRY AGAIN"* (English)
  - French - *"GROUPE ERRONE - REESSAYER"*
  - Italian - *"GRUPPO NON VALIDO - RIPETERE"*
  - Spanish - *"GRUPO NO VALIDO - INTENTE DE NUEVO"*
- *"THANK YOU - DO NOT DIST ENTRY CONFIRMED"* (English)
  - French - *"MERCI - DEMANDE EST CONFIRMEE"*
  - Italian - *"NON DISTURBARE - RICHIESTA CONFERMATA"*
  - Spanish - *"NO MOLESTAR - ENTRADA CONFIRMADA"*
- *"THANK YOU - DO NOT DIST REQUEST CANCELED"* (English)
  - French - *MERCI - DEMANDE EST ANNULEE"*
  - Italian - *"NON DISTURBARE - RICHIESTA CANCELLATA"*
  - Spanish - *"MUCHAS GRACIAS - SOLICITUD CANCELADA"*

Field Separator

The following displays show field separation:

- <calling party> *"to"* <called party> (English)

- French - <calling party> *" a"* <called party>
- Italian - <calling party> *" a"* <called party>
- Spanish - <calling party> *" a"* <called party>

Integrated Directory Display Mode

The following displays are associated with the Integrated Directory feature. Messages can be a maximum of 40 characters:

- *"DIRECTORY - PLEASE ENTER NAME"* (English)
  - French - *"ANNUAIRE - ENTRER LE NOM"*
  - Italian - *"ELENCO UTENTI - INTRODURRE NOME"*
  - Spanish - *"GUIA TELEFONICA - INTRODUZCA NOMBRE"*
- *"DIRECTORY UNAVAILABLE - TRY LATER"* (English)
  - French - *"ANNUAIRE INDISPONIBLE - REESSAYER"*
  - Italian - *"ELENCO UTENTI TEMP. NON DISPONIBILE"*
  - Spanish - *"GUIA TEL INDISPONIBLE - INTENTE DESPUES"*
- *"NO MATCH - TRY AGAIN"* (English)
  - French - *"INTROUVABLE - REESSAYER"*
  - Italian - *"NESSUNA CORRISPONDENZA - RIPETERE"*
  - Spanish - *"NO CORRESPONDE - INTENTE DE NUEVO"*

ISDN

The following displays are associated with the ISDN feature:

- *"ANSWERED BY"* (English) 15 characters maximum
  - French - *"REPONDU PAR"*
  - Italian - *"RISPOSTA DA"*
  - Spanish - *"RESPONDIDO POR"*
- *"CALL FROM"* (English) 15 characters maximum
  - French - *"APPEL DE"*
  - Italian - *"CHIAMATA DA"*
  - Spanish - *"LLAMADA DE"*
- *"INTL"* (English) - "International" 4 characters maximum.
  - French - *"INTL"*
  - Italian - *"INTL"*

— Spanish - *"INTL"*

Leave Word Calling

The following displays are associated with the Leave Word Calling feature.

The format of the leave word calling message is as follows:

- English -
  <CALLER_ID><b><DATE><b><TIME><M><b><C><b>CALL<EXT_NO>

```
<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <MONTH>/<DOM>
<MONTH>         1-12 (month of year, no leading zeroes)
<DOM>           1-31 (day of month, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            1-12 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<M>             "a" or "p"
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
<b>             blank
```

  *note: If nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

- French -
  <CALLER_ID><b><DATE><b><TIME><b><C><b>APPL<b><EXT_NO>

```
<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <DOM>/<MONTH>
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         1-12 (month of year, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
```

  *note: If nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

- Italian -

```
<CALLER_ID><b><DATE><b><TIME><b><C><b>TEL<b><EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <DOM>/<MONTH>
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         1-12 (month of year, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
```

*note: If nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

- Spanish -

```
<CALLER_ID><b><DATE><b><TIME><b><C><b>LLAM<b><EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <DOM>/<MONTH>
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         1-12 (month of year, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
```

*note: If nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

Following are the list of Leave Word Calling messages and their respective translations. Messages can be a maximum of 40 characters:

- *"CANNOT BE DELETED - CALL MESSAGE CENTER"* (English)
  - French - *"NE PEUT ETRE SUPP./APPELER RECEP. MESS."*
  - Italian - *"NON CANCELLATO. CHIAMARE CENTRO MESSAGGI"*
  - Spanish - *"NO ELIMINADO-LLAMA CENTRO DE MENSAJES"*
- *"DELETED"* (English)
  - French - *"SUPPRIME"*
  - Italian - *"MESSAGGIO CANCELLATO"*
  - Spanish - *"ELIMINADO"*
- *"END OF MESSAGES (NEXT TO REPEAT)"* (English)
  - French - *"FIN DES MESSAGES (SUIVANT POUR REPETER)"*
  - Italian - *"FINE MESSAGGI. <successivo> PER RIPETERE"*

— Spanish - *"FIN DE MENSAJES (SIGUIENTE PARA REPITIR)"*

- *"GET DIAL TONE, PUSH Cover Msg Retrieval"* (English)
  - — French - *"TONALITE D'ENVOI - <LECT. MESS. COUV.>"*
  - — Italian - *"<rec mess copert> DOPO IL TONO DI CENTR"*
  - — Spanish - *"OBTENGA TONO OPRIMA <RECUP MNSJE COBERT>"*
- *"IN PROGRESS"* (English)
  - — French - *"EN COURS"*
  - — Italian - *"ATTENDERE..."*
  - — Spanish - *"EN CURSO"*
- *"MESSAGE RETRIEVAL DENIED"* (English)
  - — French - *"LECTURE DE MESSAGES INTERDITE"*
  - — Italian - *"LETTURA MESSAGGIO NON PERMESSA"*
  - — Spanish - *"RECUPERACION DE MENSAJES DENEGADA"*
- *"MESSAGE RETRIEVAL LOCKED"* (English)
  - — French - *"LECTURE DE MESSAGES BLOQUEE"*
  - — Italian - *"LETTURA MESSAGGIO BLOCCATA"*
  - — Spanish - *"RECUPERACION DE MENSAJES BLOQUEADA"*
- *"MESSAGES FOR"* (English)
  - — French - *"MESSAGES POUR"*
  - — Italian - *"MESSAGGI PER"*
  - — Spanish - *"MENSAJES PARA"*
- *"MESSAGES UNAVAILABLE - TRY LATER"* (English)
  - — French - *"MESSAGES INDISPONIBLES - REESSAYER"*
  - — Italian - *"MESSAGGI TEMPORANEAMENTE NON DISPONIBILI"*
  - — Spanish - *"MENSAJES NO DISPONIBLES, INTENTE DESPUES"*
- *"Message Center (AUDIX) CALL"* (English)
  - — French - *"APPEL DE LA RECEPTION DE MESS. (AUDIX)"*
  - — Italian - *"Chiamata dal Centro Messaggi (AUDIX)"*
  - — Spanish - *"LLAMADA DEL CENTRO DE MENSAJES (AUDIX)"*
- *"NO MESSAGES"* (English)
  - — French - *"PAS DE MESSAGES"*
  - — Italian - *"NESSUN MESSAGGIO"*
  - — Spanish - *"NINGUN MENSAJE"*

- *"WHOSE MESSAGES? (DIAL EXTENSION NUMBER)"* (English)
  - French - *"MESSAGES DE QUEL NO.? (ENTRER NO. POSTE)"*
  - Italian - *"LETTURA MESSAGGI. INTRODURRE NUMERO TEL."*
  - Spanish - *"MENSAJES DE QUIEN? (MARCAR EXTENSION)"*

Miscellaneous Attendant Features

The following displays are associated with Miscellaneous Attendant features.

Caller Information (6 character maximum)

- *"Info:"* (English)
  - French - *"INFO.:"*
  - Italian - *"Info:"*
  - Spanish - *"INFORM:"*

Emergency Access to Attendant (40 characters maximum)

- *"a=xxxxxxxxxxxxxxx Ext xxxxx xx in EMRG Q"* (English)
  - French - *"a=xxxxxxxxxxxxxxx POSTE xxxxx xx FIL URG"*
  - Italian - *"a=xxxxxxxxxxxxxxx Der xxxxx xx in C EMRG"*
  - *Spanish - "a=xxxxxxxxxxxxxxx EXT xxxxx xx EN C EMRG"*

Queue Status (40 characters maximum)

- *"HUNT GROUP <x> NOT ADMINISTERED"* (English)
  - French - *"GROUPE DE DIST. <x> NON ADMINISTRE"*
  - Italian - *"GRUPPO <x> NON AMMINISTRATO"*
  - Spanish - *"GRUPO BUSQUEDA <x> NO ADMINISTRADO"*

Queue Status Indication [14] (40 characters maximum)

- *"<15 chrs> Q-time xx:xx calls xx"* (English)
  - French - *"<15 chrs> TEMPS-F xx:xx APPELS xx"*
  - Italian - *"<15 chrs> T-coda xx:xx chiam xx"*
  - Spanish - *"<15 chrs> HORA-C xx:xx LLAMADAS xx"*

Miscellaneous Call Identifier

The following displays are associated with Miscellaneous Call Identifiers. Messages can be a maximum of 2 characters:

"English Display" (stands for)	"French Display" (stands for)	"Italian Display" (stands for)	"Spanish Display" (stands for)
"sa" (ACD Supervisor Assistance)	*"AS"* (Assistance surveillant)	*"as"* (Assistenza Supervisoree)	*"AS"* (Ayuda del supervisor)
"ac" (Attd Assistance Call)	*"AA"* (Appel assistance)	*"ao"* (Assistenza Operatore)	*"AO"* (Ayuda de operadora)
"tc" (Attd Control Of A Trunk Group)	*"CF"* (Commande faisceau)	*"fc"* (Fascio Controllato)	*"CE"* (Control enlaces)
"an" (Attd No Answer)	*"TR"* (Telephoniste sans reponse)	*"on"* (Operatore Non Risponde)	*"ON"* (Operadora no responde)
"pc" (Attd Personal Call)	*"AP"* (Appel personnel)	*"cp"* (Chiamata Personale)	*"LP"* (Llamada personal)
"rc" (Attd Recall Call)	*"RA"* (Rappel)	*"rc"* (Richiamata)	*"RL"* (Rellamada)
"rt" (Attd Return Call)	*"RE"* (Retour)	*"rt"* (Ritornata)	*"RT"* (Retorno)
"sc" (Attd Serial Call)	*"AS"* (Appel en serie)	*"ic"* (Inoltro a Catena)	*"LS"* (Llamada en serie)
"co" (Controlled Outward Restriction)	*"RD"* (Restriction de depart)	*"cu"* (Controllata Uscente)	*"RS"* (Restriccion saliente)
"cs" (Controlled Station to Station Restriction)	*"RP"* (Restriction vers postes)	*"cd"* (Controllata Derivati)	*"CS"* (Control estacion)

Miscellaneous Call Identifier <CONTINUED>

"English Display" (stands for)	"French Display" (stands for)	"Italian Display" (stands for)	"Spanish Display" (stands for)
"ct" (Controlled Termination Restriction)	*"AR"* (Restriction d'arrivee)	*"ct"* (Controllata Terminante)	*"RE"* (Restriccion entrante)
"db" (DID Find Busy Station With CO Tones)	*"OP"* (Occupation du poste)	*"po"* (Passante Occupata)	*"EO"* (Estacion occupada)
"da" (DID Recall Go To Attd)	*"RT"* (Rappel telephoniste)	*"pr"* (Richiamata su Passante)	*"RD"* (Rellamada directa)
"qf" (Emerg. Queue Full Redirection)	*"FP"* (File d'urgence pleine deviation)	*"de"* (Deviata Emergenza)	*"DE"* (Desvio de emergencia)
"hc" (Held Call Timed Reminder)	*"AG"* (Indicatif d'appel en garde)	*"at"* (Avviso Chiamata in tenuta)	*"LR"* (Recordatorio de llamada retenida)
"ic" (Intercept)	*"IN"* (Interception)	*"in"* (Intercettata)	*"IN"* (Intercepcion)
"ip" (Interposition Call)	*"AI"* (Appel interposition)	*"ip"* (Interposizione)	*"EP"* (Entre posiciones)
"ld" (LDN Calls on DID Trunks)	*"SD"* (Selection directe)	*"pd"* (Diretta Passante)	*"LD"* (Larga distancia)
"so" (Service Observing)	*"ES"* (ecoute du service)	*"is"* (Inclusione Supervisore)	*"SS"* (Supervision del servicio)
"na" (Unanswered or Incomplete DID Call)	*"SR"* (Sans reponse)	*"pn"* (Passante Non Risposta)	*"SR"* (Sin respuesta)

Miscellaneous Call Identifier

The following displays are also associated with Miscellaneous Call Identifiers. These messages can be a maximum of 8 characters:

"English Display" (stands for)	"French Display" (stands for)	"Italian Display" (stands for)	"Spanish Display" (stands for)
"ACB" (Automatic Callback)	*"R. AUTO."* (Rappel automatique)	*"PRN"* (Prenotazione Automatica)	*"RA"* (Rellamada automatica)
"callback" (Callback Call)	*"RAPPEL"* (Rappel)	*"prenotaz"* (Prenotazione)	*"RELLAM"* (Rellamada)
"park" (Call Park)	*"G. I."* (garde par) indicatif	*"parch."* (Parcheggiata)	*"ESTAC"* (Estacionamiento de llamada)
"control" (Control)	*"CONTROLE"* (Controle)	*"cntr.op."* (Controllo Operatore)	*"CONTROL"* (Control)
"ICOM" (Intercom Call)	*"INTERCOM"* (Intercommunication)	*"ICOM"* (Intercom)	*"INTERF"* (Llamda interfono)
"OTQ" (Outgoing Trunk Queuing)	*"FFD"* (File faisceaux de depart)	*"RFO"* (Richiamata su Fascio Occupato)	*"EES"* (Espera de enlace de salida)
"priority" (Priority Call)	*"PRIORITE"* (Appel prioritaire)	*"priorita"* (Priorita')	*"PRIORIT"* (Llamada prioritaria)
"recall" (Recall Call)	*"APP.RAP."* (Appel rappel)	*"richiam"* (Richiamata)	*"REPET"* (Rellamada)
"return" (Return Call)	*"RETOUR"* (Retour)	*"ritorno"* (Chiamata Ritornata)	*"RETORNO"* (Llamada de retorno)
"ARS" (Autmomatic Route Selection)	*"SAA"* (Selection de l'acheminement automatiqe)	*"SAI"* (Selez. Autom. Instradam.)	*"SAR"* (Seleccion automatica) de rutas)
"forward" (Call Forwarding)	*"RENVOI"* (Renvoi)	*"deviata"* (Deviata)	*"REENVIO"* (Reenvio de llamada)
"cover" (Cover)	*"SUPPL."* (Suppleance)	*"copert."* (Copertura)	*"COBER"* (Cobertura)
"DND" (Do Not Disturb)	*"NPD"* (Ne pas deranger)	*"nd"* (Non Disturbare)	*"NM"* (No molestar)

Miscellaneous Call Identifier

The following displays are also associated with Miscellaneous Call Identifiers. These messages can be a maximum of 1 character:

"English Display" (stands for)	"French Display" (stands for)	"Italian Display" (stands for)	"Spanish Display" (stands for)
"p" (Call Pickup)	*"P"* (Prise)	*"a"* (Assente)	*"C"* (Captura de llamada)
"c" (Cover All Calls)	*"s"* (Suppleance)	*"c"* (Copertura)	*"c"* (Cobertura de toda llamada
"n" (Night Sta. Serv., Incoming No Answer)	*"N"* (Service nuit, entrant pas reponse)	*"n"* (Serv. Notte, Esterna Non Risposta)	*"N"* (Servicion noct. ext. no responde)
"B" (All Calls Busy)	*"O"* (Tous occupes)	*"O"* (Tutte Occupate)	*"O"* (Todas ocupadas)
"f" (Call Forwarding)	*"R"* (Renvoi)	*"d"* (Deviata)	*"R"* (Reenvio de llamada)
"b" (Cover Busy)	*"o"* (Suppleance occupee)	*"o"* (Copertura per Occupato)	*"o"* (Cobertura ocupada)
"d" (Cover Don't Answer)	*"n"* (Suppleance pas de reponse)	*"n"* (Copertura per Non Risposta)	*"n"* (Cobertura sin respuesta)
"s" (Send All Calls)	*"E"* (Envoi tous appels)	*"r"* (Rinvio)	*"E"* (Envio de toda llamada)

Party Identifiers

The following displays are associated with Party Identifiers. Party Identifiers can show up in two different ways (through administration and through DCS). Identifiers that are administrable will not be translated. Party identifiers that appear on a display due to DCS calling, will be translated. Messages can be a maximum of 15 characters:

Identifier	"English Display"	"French Display"	"Italian Display"	"Spanish Display"
*Attendant	*"OPERATOR"*	*"TELEPHONISTE"*	*"OPERATORE"*	*"OPERADORA"*
Conference Call	*"CONFERENCE"*	*"CONFERENCE"*	*"CONFERENZA"*	*"CONFERENCIA"*
Extension	*"EXT"*	*"POSTE"*	*"DER"*	*"EXTENSION"*
+Paging	*"PAGING"*	*"PAGING"*	*"PAGING"*	*"PAGING"*
*Trunk Group	*"OUTSIDE CALL"*	*"APPEL EXT."*	*"ESTERNA"*	*"LLAMADA EXT."*
Unknown	*"UNKNOWN NAME"*	*"INTROUVABLE"*	*"NOME SCONOSC."*	*"DESCONOCIDO"*

* These displays are administrable and will only appear translated if associated with a DCS call. If not associated with a DCS call, the name that will appear will be that name administered on the associated administration form.

\+ This display will never be translated.

Property Management System Interface

The following displays are associated with the Property Management System feature. Messages can be a maximum of 40 characters:

- *"CHECK IN - Ext:"* (English)
  - French - *"ENREGISTREMENT - POSTE:"*
  - Italian - *"CHECK IN - Tel:"*
  - Spanish - *"REGISTRARSE - EXTENSION:"*
- *"CHECK IN: ROOM ALREADY OCCUPIED"* (English)
  - French - *"ENREGISTREMENT: CHAMBRE OCCUPEE"*
  - Italian - *"CHECK IN: CAMERA OCCUPATA"*
  - Spanish - *"REGISTRARSE: HABITACION OCUPADA"*
- *"CHECK IN COMPLETE"* (English)
  - French - *"ENREGISTREMENT EFFECTUE"*
  - Italian - *"CHECK IN COMPLETATO"*
  - Spanish - *"REGISTRO TERMINADO"*
- *"CHECK IN FAILED"* (English)
  - French - *"ECHEC D'ENREGISTREMENT"*
  - Italian - *"CHECK IN ERRATO"*
  - Spanish - *"REGISTRARSE: FALLIDO"*
- *"CHECK OUT - Ext:"* (English)
  - French - *"DEPART - POSTE:"*

  - Italian - *"CHECK OUT - Tel:"*
  - Spanish - *"PAGAR LA CUENTA - EXTENSION:"*
- *"CHECK OUT COMPLETE: MESSAGE LAMP OFF"* (English)
  - French - *"DEPART: PAS DE MESSAGES"*
  - Italian - *"CHECK OUT COMPLETATO: NESSUN MESSAGGIO"*
  - Spanish - *"PAGO TERMINADO: NINGUN MENSAJE"*
- *"CHECK OUT COMPLETE: MESSAGE LAMP ON"* (English)
  - French - *"DEPART: MESSAGES"*
  - Italian - *"CHECK OUT COMPLETATO: MESSAGGI IN ATTESA"*
  - Spanish - *"PAGO DE CUENTA TERMINADO: MENSAJES"*
- *"CHECK OUT FAILED"* (English)
  - French - *"ECHEC PROCEDURE DE DEPART"*
  - Italian - *"CHECK OUT ERRATO"*
  - Spanish - *"PAGAR LA CUENTA: FALLIDO"*
- *"CHECK OUT: ROOM ALREADY VACANT"* (English)
  - French - *"DEPART - CHAMBRE INOCCUPEE"*
  - Italian - *"CHECK OUT: CAMERA NON OCCUPATA"*
  - Spanish - *"PAGAR LA CUENTA: HABITACION VACANTE"*
- *"MESSAGE LAMP OFF"* (English)
  - French - *"PAS DE MESSAGES"*
  - Italian - *"NESSUN MESSAGGIO IN ATTESA"*
  - Spanish - *"LUZ DE MENSAJE APAGADA"*
- *"MESSAGE LAMP ON"* (English)
  - French - *"MESSAGES"*
  - Italian - *"MESSAGGI IN ATTESSA"*
  - Spanish - *"LUZ DE MENSAJE ENCENDIDA"*
- *"MESSAGE NOTIFICATION FAILED"* (English)
  - French - *"ECHEC D'AVIS MESSAGES"*
  - Italian - *"NOTIFICA MESSAGGI ERRATA"*
  - Spanish - *"AVISO DE MENSAJE FALLIDO"*
- *"MESSAGE NOTIFICATION OFF - Ext: xxxxx"* (English)
  - French - *"AVIS DE MESSAGES DESACTIVE - POSTE:xxxxx"*
  - Italian - *"NOTIFICA MESSAGGI DISABIL. - Tel: xxxxx"*

  - Spanish - *"AVISO DE MENSAJE APAGADO - EXT: xxxxx"*
- *"MESSAGE NOTIFICATION ON - Ext: xxxxx"* (English)
  - French - *"AVIS DE MESSAGES ACTIVE - POSTE:xxxxx"*
  - Italian - *"NOTIFICA MESSAGGI ABILITATA - Tel: xxxxx"*
  - Spanish - *"AVISO DE MENSAJE ENCENDIDO - EXT: xxxxx"*

Security Violation Notification

The following displays are associated with the Security Violation Notification feature. Messages can be a maximum of 40 characters:

- *"Barrier Code Violation"* (English)
  - French - *"VIOLATION DU CODE D'ENTREE"*
  - Italian - *"Violazione di codici di taglio"*
  - Spanish - *"VIOLACIAON CONDIGO LIMITE"*
- *"Login Violation"* (English)
  - French - *"VIOLATION DE L'ACCES A L'ADMINISTRATION"*
  - Italian - *"Violazione di inizio di registrazione"*
  - Spanish - *"VIOLACION CLAVE ACCESO"*

Stored Number

The following displays are associated with the Stored Number feature.

If a number is not stored (40 characters):

- *"NO NUMBER STORED"* (English)
  - French - *"AUCUN NUMERO EN MEMOIRE"*
  - Italian - *"NESSUN NUMERO IN MEMORIA"*
  - Spanish - *"NINGUN NUMERO ALMACENADO"*

Stored numbers are displayed just as dialed. Numeric and touch-tone characters will not be changed. Special codes, one character in length, will appear as follows.

"English Display" (stands for)	"French Display" (stands for)	"Italian Display" (stands for)	"Spanish Display" (stands for)
"m" (Mark)	*"M"* (Marquer)	*"m"* (Marcato)	*"M"* (Marca)
"p" (Pause)	*"P"* (Pause)	*"p"* (Pausa)	*"P"* (Pausa)
"s" (Suppress)	*"S"* (Supprimer)	*"s"* (Soppresso)	*"S"* (Suprimir)
"w" (Wait)	*"A"* (Attendre)	*"a"* (Attesa)	*"E"* (Espera)

Time of Day Routing

The following displays are associated with the Time of Day Routing feature. Messages can be a maximum of 40 characters:

- *"ENTER ACTIVATION ROUTE PLAN, DAY & TIME"* (English)
  - French - *"ENTRER PLAN D'ACTIVATION, JOUR ET HEURE"*
  - Italian - *"INTRODURRE PIANO DA ATTIV., GIORNO E ORA"*
  - Spanish - *"INTRODUZCA PLAN ACT DE RUTAS, DIA Y HORA"*
- *"ENTER DEACTIVATION DAY AND TIME"* (English)
  - French - *"ENTRER JOUR ET HEURE DE DESACTIVATION"*
  - Italian - *"INTRODURRE GIORNO E ORA DI DISATTIVAZ."*
  - Spanish - *"INTRODUZCA DIA Y HORA DE DESACTIVACION"*
- *"OLD ROUTE PLAN: x ENTER NEW PLAN:"* (English)
  - French - *"ACHEMINEMENT ANT.: x ENTRER NOUVEAU:"*
  - Italian - *"INSTRADAMENTO PREC: x INTROD IL NUOVO:"*
  - Spanish - *"PLAN RUTAS ANT: x INTRODUZCA EL NUEVO:"*
- *"OLD ROUTE PLAN: x NEW PLAN: y"* (English)
  - French - *"ACHEMINEMENT ANT.: x NOUVEAU PLAN: y"*
  - Italian - *"INSTRADAMENTO PREC: x NUOVO PIANO: y"*
  - Spanish - *"PLAN RUTAS ANT: x NUEVO PLAN: y"*
- *"ROUTE PLAN: x FOR yyy ACT-TIME: zz:zz"* (English)
  - French - *"ACHEM.: x POUR yyy ACT-HEURE: zz:zz"*
  - Italian - *"INSTRADAMENTO: x PER yyy ATTIV ORE:zz:zz"*

  — Spanish - *"PLAN RUTAS: x PARA yyy HORA-ACT: zz:zz"*

- *"ROUTE PLAN: x FOR yyy DEACT-TIME: zz:zz"* (English)
  — French - *"ACHEM.: x POUR yyy DESACT-HEURE: zz:zz"*
  — Italian - *"INSTRADAM.: x PER yyy DISATTIV ORE:zz:zz"*
  — Spanish - *"PLAN RUTAS: x PARA yyy HORA-DESACT:zz:zz"*

For the above displays, x and y denotes the Route Plan Number (RPN 1-8), yyy is a three letter abbreviation for the day of the week and zz:zz is the activation time (military time). The three-letter abbreviation for the day of the week is as follows:

"English Display"	"French Display"	"Italian Display"	"Spanish Display"
"Mon"	*"LUN"*	*"Lun"*	*"LUN"*
"Tue"	*"MAR"*	*"Mar"*	*"MAR"*
"Wed"	*"MER"*	*"Mer"*	*"MIE"*
"Thu"	*"JEU"*	*"Gio"*	*"JUE"*
"Fri"	*"VEN"*	*"Ven"*	*"VIE"*
"Sat"	*"SAM"*	*"Sab"*	*"SAB"*
"Sun"	*"DIM"*	*"Dom"*	*"DOM"*

— English, French, Italian, Spanish - To enter the day of the week, the user dials 1 for Sunday, 2 for Monday, and so on.

Multiple Listed Directory Numbers

Description

Allows a publicly published number for each incoming and two-way (incoming side) FX and local CO trunk group assigned to the system. Also allows up to eight DID numbers to be treated as 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
- ACD split
- DDC group
- UCD group
- Remote Access

All DID LDN calls route directly to the attendant group.

Considerations

Multiple LDNs provide 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 LDNs 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 LDNs 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. 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.

For G1.1 and G3i, the treatment of transferred trunk calls can be controlled by the System Parameter field "Music (or Silence) on Transferred Trunk Calls".

Interactions

When any one of the following features is activated, music is provided when one party is waiting or held:

- Hold
- Conference — Terminal
- Transfer (application of music or silence as opposed to ringback tone can be controlled for trunk calls in G1.1 and G3i)

In addition to these three features, a single party in Call Park can receive music. Also, a call placed in queue for a DDC group, UCD group, or ACD 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 the assignment of whether music,tone, 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 (TN763D supports A-law), or one port on a supported Analog Line circuit pack such as the TN742. A KS-23395L3 coupler is required to connect the music source to the Analog Line port. 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. No additional software is required.

Names Registration

Description

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, 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, and so on). 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, would see John's name and room extension and could 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 (G1) or the G3 Management Terminal. Normally, in a hotel environment where the daily turnover of guests is large, manual administration of the updates using a Manager I terminal (G1) or a G3 Management 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 PMS. 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 checkin 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 discussed in the PMS description elsewhere in this document.

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 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 checkin 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 could 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, and so on) 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 upper-case letters, all lower-case 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 document), the guest's name must be entered using one of the following methods:

- Last Name, comma, First Name (for example, Jones,Fred)
- Last Name, comma, First Name, space, Title/Middle Initial/Name (for example, Jones,Fred Mr)
- Last Name Only (for example, 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 and may use periods. 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 s, 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 (one through 600), and manual administration of the specific path can be performed through the Manager I terminal (G1) or the G3 Management 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 (G1) or the G3 Management 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 checkin 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 (G1) or the G3 Management Terminal, while automatic updates can still be sent from the PMS for guests' extensions.

- 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 in memory is not affected. However, if the translations are read in, existing extensions will be affected until a database swap synchronizes the switch and PMS.

- 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 PMS Interface description, elsewhere in this chapter, is required. In addition to this, the items in the following paragraphs should be taken into consideration.

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 would be 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, prearranged paths can be administered on the switch and the numbers stored within the PMS that would 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 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 an MPDM and port on a Digital Line circuit pack or through an 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 PMS Interface feature.

Network Access — Private

Description

Allows calls to be connected to the following types of networks:

- CCSA
- ETN
- EPSCS
- 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 99 trunk groups can be assigned to the system, including private network trunk groups.

Unless prohibited by the 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.

In Italy, the Traslatore Giunzione Uscente/Entrante/Interno trunks are supported to provide Private Network Access between two systems and also provide some feature transparency for COR (Inward Restriction), DID (when reaching busy stations), and Intrusion features.

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:

- COs
- FX offices (distant COs)
- WATS offices (COs receiving toll-free calls)

Incoming access is provided from the following:

- Local COs
- FX offices
- 800 Service office (COs sending toll-free calls)

Considerations

The ARS 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.

Some central offices do not provide disconnect supervision but disconnect supervision can be provided in this case through the trunk group admin screen.

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 circuit pack (TN464B/C/D support A-law) for each trunk assigned. No additional software is required.

Night Service — Hunt Group

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 mode, 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 the hunt group is 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 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:

- ACD

  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 for the original split, if administered, before redirecting. 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: forced or delayed first announcement, ringback, 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 destination of a hunt group NSE's coverage path is AUDIX, AUDIX answers with the mailof the original hunt group. 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.

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 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 DID 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:

- DID-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. 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 as discussed above. A different extension number can be assigned as the night destination for each incoming central office, foreign exchange, or 800 Service trunk 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, DDC group, UCD group, or TEG.

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.

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.

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.

If Night Service is activated and the DID LDN night extension is busy, an incoming DID LDN call receives busy tone or may be forwarded to another number.

If a trunk without disconnect supervision goes to night service, it will be dropped after a certain period of time to avoid locking up the trunk.

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 premise from the night extension) will not work.

- Call Forwarding All Calls

  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)
- 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.

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 which 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

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, TN746B (A-law), or TN769 Analog Line circuit pack. No additional software is required.

Night Service — Trunk Group

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 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 DID 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:

- LDN

  In the System Night Service mode, all incoming LDN calls, except those using 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 CO trunks are used, the voice terminal must be analog and must be FCC-registered (or, outside the U.S., registered by the appropriate governmental agency).

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 20,000 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 one-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, TN746, or TN769 Analog Line circuit pack or one port on a TN767 DS1 Tie Trunk circuit pack (TN464B/C/D support A-law) for each interface to be provided. No additional software is required.

PC/PBX Connection

Description

Brings the voice terminal and 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.

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 Digital Line Circuit Pack (TN413, TN754B support A-law) is required for each PC to be connected. See the previously discussed descriptions for software/hardware packages 1, 3, and 5 for additional information.

Permanent Switched Calls (PSC) (G1.1)

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 with data modules, 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 two-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 ''DATA CALL SETUP'' for the dialing format).

With G3i, the PSC feature is replaced with the Administered Connections feature.

Interactions

The following features interact with the PSCs 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.
- CORs — Determine CORs 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 PMS and CDR 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 (PCOL)

Description

Provides a dedicated trunk for direct access to or from the public network for multi-appearance voice terminal users.

Each 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.

CO, FX, and WATS trunks can be assigned to this feature.

PCOLs are not assigned a COR.

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 40 PCOLs. These lines (trunks) are not included in the trunk groups supported by the system. They are, however, included in the 400-trunk system limit.

Interactions

The following features interact with the PCOL 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.

- LWC

  LWC 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.

- CDR

  The CDR feature can be activated for PCOL calls, but the CDR 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.

- Send All Calls

  Send All Calls cannot be activated for a PCOL group.

- 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:

- ARS
- 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

- CDR 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 CDR)
- 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)
- Timers based on board capabilities

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

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, 7410D, 7505D, 7506D, 7507D, 8503T, 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, 7410D, 7505D, 7506D, 7507D, 8503T, 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. The user specified ringing pattern for a 7410D is lost if the set loses auxiliary power.

Hardware and Software Requirement

No additional hardware or software is required.

Power Failure Transfer

Description

Provides service to and from the local telephone company CO, including 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 five 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 or DTMF) 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.

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.

Priority Calling

Description

Provides a special form of call alerting between internal voice terminal users. The called voice terminal user receives a distinctive three-burst alerting signal.

An active single-line voice terminal user who receives a Priority Calling call will hear a distinctive three-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 COS.

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 three-burst ringing signal.

- Call Forwarding All Calls

  Priority Calling calls (except callback calls) will forward, and the forwarded call remains a Priority Calling call.

- 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 three-burst priority Call Waiting tone.

- Consult

  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.

- Dial Access to Attendant

  A Priority Calling call cannot be originated to the attendant. However, the attendant can originate Priority Calling calls.

- 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 COS).

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

  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 PCOL, TEG, 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

Description

Provides a communications link between the System and a customer-owned 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
- Guest Information Input/Change
- Support of five-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, and Guest Information Input/Change 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 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 (1,200 bps in V3 and 1200, 2400, 4800, or 9,600 bps in G1.1 and G3i).

Two protocol modes are provided; the normal protocol mode as described above, and the transparent protocol mode. The transparent protocol mode supports ASCII character transmission and is required for PMS features such as Names Registration and Guest Information Input/Change. 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

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 data bases.
- 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 Manager I terminal (G1) or the G3 Management Terminal) 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
- 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
- Call Coverage path

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 AUDIX or LWC, 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 LWC.

Any console or terminal used to activate and deactivate Message Waiting Notification must be assigned a Console Permissions COS. The affected extension must have a Client Room COS.

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 data base 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 screen is administered as `act-nopms`), then Housekeeping Status information is written to a log, can be accessed through the Manager I terminal (G1) or the G3 Management Terminal, 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 or DTMF 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 or DTMF buttons to normal operation.

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 (transparent mode), and the guest’s Call Coverage path (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

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 checkout. The Names Registration feature is described in detail elsewhere in this document.

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 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 checkin 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 LWC 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.

The normal protocol mode allows extensions of up to four digits in length. The transparent protocol mode allows extensions of up to five digits in length.

When a ''save translations'' is done on a system with the transparent protocol mode active, station names with Client Room COS are saved as ''blank'' and coverage paths are saved as the ''Default Coverage Path for Client Rooms''.

Interactions

The following features interact with the PMS 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.

- 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.

- LWC

  Message lamps activated by this feature cannot be deactivated by Manual Message Waiting feature buttons.

  With a system using the transparent protocol mode, any LWC messages that are present when a check-In request is processed are deleted.

  If Room Change is activated, LWC messages for the old room will *not* be moved to the new room. If Room Swap is activated, LWC messages for the two rooms will *not* be swapped. Therefore, use of the LWC 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 zero to six 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 CDR or PSC extension.
- Seconds Before PMS Link Idle Time-out — 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 five to 20.
- Milliseconds Before PMS Link Acknowledgment Time-out — 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 acknowledgment or send an inquiry for an acknowledgment from the PMS for a message before giving up on on the message transmission. Choice is a number from one 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 (acknowledgment or negative acknowledgment) that the system did not receive before giving up on the incoming message. Choice is a number from one 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 COS. The choice is a number from one 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 COS. The affected extension must have Client Room COS.

Hardware and Software Requirements

A PMS, if used, can be connected through an MPDM and port on a Digital Line circuit pack or through an ADU and a port on a Data Line circuit pack. A 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.

Pull Transfer

Description

This feature is new for G3i and the system comes with this feature turned off by default. Pull transfer is an enhancement of the standard transfer operation. Standard transfer allows voice terminal users to transfer trunk or internal calls to other voice terminals within the system without attendant assistance. The Pull Transfer feature allows either the calling or the called party (the party to whom the held party will be transferred) to complete the transfer operation.

Analog telephone called parties who wish to pull transfer the party that the controlling party has on hold should momentarily flash the switchhook (or press the Flash key or the Recall button). This completes the transfer of the held party to the called party.

Digital telephone called parties who wish to pull transfer the party that the controlling party has on hold should press the Transfer key. This completes the transfer of the held party to the called party.

Please see the Transfer feature for a description of the regular (push) transfer feature.

Considerations

The pull 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.

If the attendant is the controlling party, any attempt to complete a Pull Transfer operation by the called party is ignored. Pull Transfer cannot be completed if the Attendant is the called party. A held party can only be transferred by the attendant with Push Transfer. A held party can only be transferred to the attendant with Push Transfer.

Pull Transfer can only be completed over TGU/TGE tie trunks.

Interactions

Analog Station Recall Operation and Feature Activation: If the controlling party (with a party on hold) is talking with the called party, and either analog station recall or feature activation is initiated by the called party, the controlling party will not be put into the Hold for Transfer mode but will be Pull Transferred.

Digital Station Tranfer Operation: if the controlling party (with a party on hold) is talking with the called party and the transfer operation is initiated by the called party, the controlling party will not be put into the Hold for Transfer mode but will be Pull Transferred.

CDR: Checks will be made to ensure that calls are correctly recorded with CDR when a Pull Transfer operation is completed.

Administration

The system miscellaneous form is used to enable Pull Transfer.

Hardware and Software Requirements

No special hardware is required to implement the Pull Transfer feature in a standalone configuration. However, in a network environment, the TGU/TGE tie trunks are the only trunks that support the flash signalling necessary to complete the Pull Transfer operation between switches.

Queue Status Indications

Description

Provides indications of queue status 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 the number of calls or 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 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 one to 200 calls. When this threshold is reached, the lamp associated with the Number of Queued Calls 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 Number of Queued Calls 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 steadily.

- 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 Oldest Queued Time (Oldest Queued Time) button with associated lamp on a voice terminal or console. Each split is assigned an Oldest Queued Time warning threshold of zero to 999 seconds. When the oldest call in queue has been in queue for this length of time, the lamp associated with the Oldest Queued Time 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 Oldest Queued Time 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 steadily.

Each Number of Queued Calls and Oldest Queued Time button is associated with a specific split. Display-equipped voice terminals and consoles can display queue status information for a split by pressing the Oldest Queued Time or

Number of Queued Calls 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), OQT, and Number of Queued Calls are displayed for five seconds unless the displaying terminal or console receives an incoming call or the display is put into another mode. Otherwise, at the end of five 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 (DDC and UCD). 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 (Attendant group's Queued Time) (AQT) and the Attendant group's Queued Calls (AQC) 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 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)

- NQC warning threshold (one to 200 calls) (per split or attendant group)
- OQT warning threshold (zero to 999 seconds) (per split or attendant group)
- Port number assigned to auxiliary queue warning lamp (per split)

Hardware and Software Requirements

Each auxiliary queue warning lamp requires one port on a TN742, TN746B (A-law), or TN769 Analog Line circuit pack. A beehive-type lamp may be used as an auxiliary queue warning lamp. This lamp is available from the Custom Work Group. The lamp operates on ringing voltage and can be mounted at a location convenient to the group. A "lamped" button on a multi-line set is required for the NQC/OQT/AQT or AQC buttons.

No additional software is required.

Hardware and Software Requirements

A TN744 Call Classifier circuit pack is required.

R2-MFC Signaling

Description

R2 Multifrequency Compelled Signaling is used primarily in national and international voice-switched networks as a robust and flexible signaling scheme. The R2-MFC signaling provides a third addressing option to the product, complementing the existing rotary and DTMF options. The R2-MFC signaling can be used on both incoming trunk calls and outgoing trunk calls. But only group II signaling protocol is supported on outgoing trunk calls.

Considerations

The R2-MFC signaling can be used in CO, DID, and TIE trunks. Both non-group II signaling and group II signaling are supported on incoming R2-MFC calls. The group II signaling protocol has an extra signal which provides the caller category information. However the closed numbering plan is assumed on incoming R2-MFC calls. It means that the length of extension number should be fixed-length and this length should be administered on the incoming trunk form. Only group II signaling is supported on outgoing R2-MFC calls. ANI(Automatic number identification) is also supported on outgoing R2-MFC call.

Interactions

Introductory sentence needed here.

- MNCR

  For outgoing R2-MFC calls, the number of digits to be collected from a station and to be outpulsed is based on the MNCR translation.

- DID No Answer Timer

  The DID NO Answer timer is applied to R2-MFC DID calls.

- Night Service No Answer Timer

  If a R2-MFC DID call is routed to the night service destination and the call is not answered, the R2-MFC DID call will be dropped when the Night Service no answer timer is expired.

- Call Redirection

  Calls will be redirected if any of the following are active: Call Forwarding, Call Coverage, Send All Calls, or Night Service and the call is redirected to a station other than the one indicated by the R2-MFC signals received from the CO. For incoming group II R2-MFC calls, the backward signal B.X sent by the PBX will correspond to the status of the terminating endpoint after redirection.

- Call Waiting

  If this feature is activated on an analog station, and only one call is active at the analog station and an incoming group II R2-MFC call terminates at the analog station, then status of the analog station is treated as idle and the corresponding B.X signal for the ringing is sent to the CO.

- Call Forwarding

  If the called station has the Call Forwarding feature activated, the incoming R2-MFC call is forwarded to the designated destination.

- Send All Calls

  If this feature is activated, the incoming R2-MFC call is immediately redirected to coverage.

- Call Coverage

  The incoming R2-MFC call may be redirected to coverage according to the coverage criteria assigned for external calls.

- Display Feature

  When display feature is provided, incoming R2-MFC calls will be identified. If the display feature is activated on the trunk group, the display will be updated after the outgoing R2-MFC call is answered.

- LDN and Multiple LDNs

  R2-MFC DID calls to LDN numbers should be routed to a LDN extension with unique identification of each LDN.

- Night Service

  If night service is activated, R2-MFC DID calls route to a designated night extension.

- Call Pickup

  Incoming R2-MFC calls can be answered by Call Pickup (if Call Pickup is assigned to the station being rung by an incoming R2-MFC call).

Administration

The administrator of MFC signaling can turn the feature on or off from the customer option form. The default should be set to "n" to turn the feature off. The incoming call type is also administrable; available options are group II call type and non-group II call type (default). The outgoing call type is administrable; options are group II call type and blank (default). If blank is administered, screens for the outgoing MF administration will not be displayed. Intercept treatment for MF DID calls is administrable; options are to send the B.X signal and intercept tone for the intercept to the CO (default) or not. If the administrator chooses not to send the B.X signal to the CO, then normal DID/TIE/ISDN intercept treatment will be given.

Hardware and Software Requirements

A TN744 version 7 or later Call Classifier circuit pack is required as are analog/digital CO/DID/TIE trunks.

Recall Signaling

Description

Allows the user of an analog station to place a call on hold and consult with another party. After consulting with that third party, the user can conference the third party with the original party by another recall signal, or return to the original party by two recall signals.

The recall signaling can be accomplished by pressing the flashhook, using a Ground Key on a Rotary or DTMF station, or by using the Recall Button on a DTMF station.

Considerations

Use of the flashhook for recall signaling may at times place calls on hold when the user of the analog station intended the previous call to be dropped before dialing the third party.

Interactions

None.

Administration

The length of the time during which the system will recognize the press of the flashhook as recall signaling is administrable. Administrators may also choose not to administer recall signaling at all. In addition, the recall signaling can be disabled for particular analog stations.

Hardware and Software Requirements

Some earlier hardware versions of the Analog Line Board do not support the administration of the length of the flashhook.

Recent Change History

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 (G1) or the G3 Management Terminal, or a remote terminal by the following users:

- Local Customer Administrator
- Local System Technician
- Remote Customer Administrator
- Remote System Technician

Commands

A command is made up of multiple words, typed on the Manager I (G1) or G3 Managment Terminal 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, and so on).

The second command word entered is the **object.** It specifies the specific object to be operated on (station, trunk group, hunt group, and so on).

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 would change the state of the translation data and would be classified as a data command and entered in the log. However, the command **display station 3600** would not change the state of the translation data and would 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 the following screen. 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
  - CDR
  - EPN
  - NET

  Table 2-36 shows the way the software correlates the port number to the user that is displayed under Port on the report.

Table 2-36. 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	CDR	CDR

Legend: EPN — Expansion Port Network
INADS — Initialization and Administration System
MB — Maintenance Board
MTP — Maintenance Tape Processor
Netcon — Network Controller
SAT — Manager I Terminal (G1) or G3 Management Terminal
CDR — Call Detail Recording

- Login — The system login of the user entering the **data** command; for example, ‘‘system technician.’’
- 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 Jul 15, 1987.’’

```
                                History
             Date of Translation Loaded: 19:53 Wed Jul 15, 1987
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  system technician   rel    station       504
07/16 09:22 SAT  system technician   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 cll 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
```

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 system technician 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 maintenance command **Set Time** to change the system clock could make the Recent Change History report look as if it were not in true last-in, first-out order.

Administration

None.

Hardware and Software Requirements

No additional hardware or software is required.

Recorded Announcement

Description

Provides a recorded announcement to the following types of calls:

- DID calls that cannot be completed as dialed
- Incoming Private Network Access calls that cannot be completed as dialed
- DDC and UCD calls that have been in queue for an assigned interval
- ACD calls that have been in queue for an assigned interval
- Any call whose destination is a Recorded Announcement extension
- Any call to a call vector which contains an "announcement," "disconnect with announcement", or "collect after announcement" step (G3i only)
- Incoming calls to a user

With G1.1, the TN750 circuit pack can provide up to 64 integrated announcements. With G3i, the TN750B circuit pack supports up to 128 integrated announcements.

When a call is directed to an integrated announcement the system checks to see if a port on a TN750-type 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 other callers are seeking the same announcement, then up to four additional callers will be connected to the same announcement.

For additional information on how Recorded Announcement functions, see the following features:

- ACD
- DDC and UCD
- Intercept Treatment
- Call Vectoring (G3i)
- Call Prompting (G3i)

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.

G1.1 systems allow as many as 64 recorded announcements. G3i systems allow as many as 128 recorded announcements.

The TN750 Announcement circuit pack has a simultaneous call capacity of 80 calls (16 ports with five 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. With G1.1, the TN750 Announcement circuit pack provides a total of 4 minutes and 16 seconds of announcement time.

With G3i, integrated announcements on the TN750B circuit pack may be recorded at 16 Kbps (for eight minutes and 32 seconds of announcement time), or 32 Kbps (for four minutes and 16 seconds of announcement time) depending on administration. A different recording speed may be used for each integrated announcement. These capacities will be different if a combination of different recording speeds is used.

Up to five callers can be connected at once to either integrated announcement or an external announcement.

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 is currently being uploaded or downloaded

Interactions

Recorded Announcement is used in conjunction with the ACD, Call Vectoring, Call Prompting, Intercept Treatment, DDC, and UCD features.

Administration

Recorded Announcement is administered by the System Manager.

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 (maximum 150) and analog port must be administered for each announcement. If the type is integrated (the announcement is recorded directly onto the TN750-type circuit pack), the System Manager must specify whether the announcement is protected against being overwritten and deleted. (A queue length of 50 is automatically assigned for the integrated announcement.)

With G3i, each integrated announcement must be assigned a specific recording rate of either 16, 32, or 64 kbps.

Hardware and Software Requirements

Announcements can be either external or integrated. Each external announcement channel requires announcement equipment and one port on a TN742 or TN746B (A-law) Analog Line circuit pack. The single or multiple channel digital recorded announcement units available from AT&T can also be equipped with a remote record capability which requires an additional port on an analog line circuit pack.

Since the integrated announcement is recorded onto the circuit pack, no announcement equipment other than the TN750 (G1.1) or TN750B (G3i) 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 or TN746B (A-law) Analog Line circuit pack or one port on a TN763 Auxiliary Trunk circuit pack (TN763D supports A-law) 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 CO, 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 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. A zero is a U.S. standard for reaching the attendant; this can be set to any digit. 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.

Setting up an Abbreviated Dialing List on Remote Access Trunks

Users can access the system, group, and enhanced Abbreviated Dialing lists via the remote access trunk. To set up an Abbreviated Dialing list on a remote access trunk, perform the following steps:

1. Set up the Abbreviated Dialing list on the console form.
2. Administer the Abbreviated Dialing list entries.
3. Dial into G1 or G3i over the remote access trunk.
4. Dial the feature access code followed by the dial code of the list entry.

NOTE:
If a barrier code and authorization code are administered, dial them first.

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 COR and COS, 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 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 event, 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

  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. Dial Tone between the barrier code and authorization code is optional. The authorization code defines his or her calling privileges within the system.

- COR

  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
- Whether or not Dial Tone is applied between the barrier code and authorization code

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

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 (G1) or G3 Management Terminal.

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 (G1) or G3 Management 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 the following screen. By setting the `Print Interval` field (described below) to scheduled or deferred, the additional fields shown in Figures 2-28 and 2-29 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 (G1) or G3 Management Terminal.

When using the **schedule** command line option to schedule a report, the Report Scheduler 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. The following screens show a Report Scheduler screen with the **immediate** option. Screen 2-28 shows a Report Scheduler screen with the **scheduled** option. Screen 2-29 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
```

```
  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_
```

```
  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_
```

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 the following screen. 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_
```

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 (G1) or G3 Management Terminal, or its printer, using the **list report-scheduler** command. A sample list is shown in the following screen. 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 (G1) or G3 Management 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.

```
                                                                    Page 1
                                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.
```

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 (G1) or G3 Management Terminal, the CDR 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 (G1) or G3 Management Terminal screen.

Interactions

There is only one processor board EIA port available for asynchronous output. The port cannot be administered for both CDR 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 MPDM or a 7400A data module and a port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law).
- The printer can be connected to the switch with an ADU and a port on a TN726 Data Line circuit pack.

There is a single EIA port in DEFINITY Generic 1 and Generic 3i. There may be contention between the CDR 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 CDR 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 CDR.

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 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.

Restriction — Controlled

Description

Allows an attendant or voice terminal user with console permission 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. DID calls are routed to the attendant or a recorded announcement. All other calls receive intercept tone. As an exception, the following call types are allowed: calls to a remote access extension, terminating trunk transmission tests, and Emergency Access to Attendant calls.
- Station-to-Station — The voice terminal cannot receive or place station-to-station calls. Such call attempts receive intercept treatment.
- Termination — 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 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 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. When a call is placed, both the individual and the group restrictions are checked.

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 forwarded-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.

- COR

  Both COR and Controlled Restrictions are checked when a call is authorized.

- UDP

  Calls dialed through the UDP 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 individual (user) and group controlled restriction.
- Type of Intercept Treatment for each type of controlled restriction.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Fully Restricted Service

Fully Restricted Service is a Class of Restriction (COR) that prevents assigned stations from having access to public network calls. Stations have access to internal calls only. In addition, fully restricted station users cannot use authorization codes to deactivate this feature.

Any calls from the public network to a station with Fully Restricted Service are redirected to intercept treatment or to the attendant. If the call is redirected to the attendant, the attendant's display indicates the call is being redirected because of Fully Restricted Service. The reason-code displayed is FULL.

When the call is redirected to the attendant, the following may be appropriate actions:

- The attendant connected with a CO call may call or intrude on the called station user.
- The attendant cannot extend, conference or bridge the redirected call.
- The attendant can place a CO call on hold and call the station with Fully Restricted Service for consultation.

Considerations

There is no limit to the number of COR's (Class of Restrictions) that can have the 'Fully Restricted Service' field marked 'y'. If a user with this COR attempts a call and the call is to the public network, then the call is routed to intercept treatment.

Interactions

Because of interactions with other features, Fully Restricted Service should not be assigned to a station under certain conditions (even though the switch allows the assignment) Conditions that prohibit the proper operation of this feature are called prohibitive interactions. Conditions that permit the proper operation of this feature are called allowable interactions. Starting with prohibitive interactions, both are listed below.

Prohibitive

- Station has Abbreviated Dialing - All the calls automatically dialed from a Privileged List are completed without any restriction checking.
- Station is an Attendant - Each Attendant is assigned a COS and a COR. When the attendant is called using its assigned extension, the COR assigned in the Attendant screen is used for authorization checks. Any calls redirected to the attendant group will use the COR assigned in the Console Parameters screen for authorization checks. The attendant group and individual attendants should not be assigned a COR with Fully Restricted Service.

- Station has mismatched Bridged Call Appearance - The appearance of a voice terminal's primary extension number at another voice terminal is called a bridged call appearance. Calls made to the bridged call appearance will use the COR of the station that has the same primary extension as the bridged call appearance for authorization checks. Stations with Fully Restricted Service should not be assigned bridged call appearances without Fully Restricted Service as the restriction can be defeated by using the bridged appearance.
- Station is a call coverage and/or Send All Calls point - The covering station is only checked for Fully Restricted Service when a call is redirected to coverage; no other COR checks are made. Calls from the public network will not be redirected to any stations that have Fully Restricted Service. When defining Call Coverage Paths, the coverage points (Point1, Point2, and Point3) can be an extension number, hunt group, coverage answer group, or attendant.
- Station is a Call Forwarding destination for outside calls - Calls from the public network will not be forwarded to a station with Fully Restricted Service instead they will terminate at the dialed station.
- Station is a Call Pickup point for outside calls - A station with Fully Restricted Service cannot pickup calls from the public network to another station in the same pickup group.
- Station number is used for Night Service - The following describes the interaction with Night Service. Night Console Service directs all calls for daytime attendant consoles to a night console when Night Service is activated. The COR assigned in the Console Parameters screen is used for authorization checks. Multiple Night Service Extension redirects incoming calls from specific trunk groups to assigned Night Service Extension (for that particular trunk group) whenever Night Service is activated. The COR of the trunk group is used for authorization checks. Stations with Fully Restricted Service should not be assigned as the Night Service Extension for public network trunks, to ensure it does not receive calls from the public network. Single Night Service Extension redirects attendant-seeking calls to a designated extension number whenever Night Service has been activated and the Night Console is either not provided or not available. The COR assigned in the Console Parameters screen is used for authorization checks. A station with Fully Restricted Service should not be assigned as the single night service extension.
- Features that have access to an outgoing trunk with an associated queue. When a station is in an outgoing trunk queue for a public network trunk and Fully Restricted Service is activated for its COR, it will still be called back when a trunk becomes available and will be able to place a call over the public network. However, any further attempts to access a public network trunk are redirected to intercept treatment.

Accessible

- Station number represents a loudspeaker paging zone - Each Loudspeaker Paging Access zone is assigned a COR, which is used to check authorization on all calls to Loudspeaker Paging. Code Calling Identification is part of Loudspeaker Paging which allows different Code Calling identifications (chime signals) to be assigned to different Access zones. Each Code Calling Access zone is assigned a COR, which is used to check authorization on all incoming calls to Loudspeaker Paging.
- Station denied for Authorization Codes - An authorization code allows, in certain cases, a terminal user (including an attendant) to dial a four to seven digit code which changes to a new COR that overrides the COR associated with the class of user.The Authorization Code feature can be used with the Automatic Route Selection (ARS), the Automatic Alternate Routing (AAR), certain incoming trunks and/or the Remote Access features. Authorization codes will not override Fully Restricted Service.
- Centralized Attendant Service (CAS) This involves two switches with calls extended by the attendant on switch A to B. Since COR information is not passed over Release Link Trunks (RLTs), Fully Restricted Service will allow all CAS calls.Therefore, this feature will allow a public network call to be completed to a station with Fully Restricted Service.
- Distributed Communication System (DCS) - Since COR information is not transparent for DCS, Fully Restricted Service will allow all DCS calls. Therefore, this feature can allow a public network call to be completed to a station with Fully Restricted Service.
- Emergency Transfer - All authorization features are bypassed when an Emergency Transfer station is connected to an Emergency Transfer trunk in the Emergency Transfer Mode.
- Hunt Group - The COR assigned to the Hunt Group is checked on calls redirected by either Direct Department Calling (DDC) or Uniform Call Distribution (UCD) of the hunt group feature. Extensions in the hunt group that have Fully Restricted Service can receive calls from the public network (via the hunt group), if the COR of the Hunt Group does not have Fully Restricted Service. Stations with Fully Restricted Service should not be assigned to Hunt Groups without Fully Restricted Service.
- Stations assigned a Personal Central Office Line (PCOL) - If a station with Fully Restricted Service is assigned a PCOL, calls can still be placed to/from the PCOL. Stations with Fully Restricted Service should not be assigned PCOL.
- In a system that associates a barrier code with remote access - If a barrier code is entered on connection to remote access, the barrier code's associated COR is used for authorization checks. If remote access does not require a barrier code (because the barrier code length is blank on the Remote Access screen), then the default barrier code's COR is used ('none' is entered as the only barrier code with an associated COR on the Remote Access screen). Remote Access can require an authorization code instead of or in addition to the barrier code. If an authorization code

is required, the authorization code's associated COR overrides the barrier code's COR. CORs assigned to remote access barrier codes and authorization codes, should have CALLING PERMISSION marked 'n' for CORs with Fully Restricted Service, so that remote access cannot call stations with Fully Restricted Service.

- Stations in a non-restricted Terminating Extension Group - A Terminating Extension Group (TEG) is assigned a COR for authorization checks on calls to the TEG extension. Terminating Extension Groups can only receive calls, they cannot originate calls.
- Stations that may use Through Dialing - On Through Dialing calls, the attendant's COR applies (not the COR assigned to the calling party). Transfer and conference checks would still be done for extensions that have Fully Restricted Service active.

Also see Class of Restrictions for detailed feature interactions of restrictions.

Administration

Administerable items are:

- COR identifiers and their associated definitions.
- Each class of user is assigned a COR.

Hardware/Software Requirements

There is no special hardware required for this feature.

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 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 COR feature by the System Manager. 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 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.

- ARS

  This feature overrides the Miscellaneous Trunk Restriction feature. Permission or denial of ARS calls is determined by the FRL.

Administration

Miscellaneous Trunk Restriction is administered via the COR feature by the System Manager. 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 (G3i)

Description

Restricts users at specified voice terminals from placing calls that have been designated as toll calls by system administration.

With the Toll Restriction feature, a Toll Analysis table can be administered to assign certain dialed digit strings to a ‘‘toll list.’’ A call containing one of the dialed digit strings assigned to the toll list is designated as being a ‘‘toll call.’’

When a user attempts to dial a toll call, as defined in the Toll Analysis table, he or she may or may not be able to place the call, depending on his or her assigned COR:

a. If the user has a COR with a calling party restriction of ‘‘All-Toll,’’ that user is prevented from making ARS or CO/FX trunk access calls to certain toll areas as defined in the Toll Analysis table, unless the number is on an UCL associated with the user’s COR. (Information on Restricted and Unrestricted Call Lists is provided in the COR feature description elsewhere in this manual.)

b. If the user has a COR with a calling party restriction of ‘‘TAC-Toll,’’ that user is prevented from making trunk access calls to certain toll areas as defined in the Toll Analysis table, unless the number is on an UCL associated with the user’s COR. TAC-Toll restrictions are included with the All-Toll calling party restriction. The only difference is that All-Toll also applies to ARS calls.

If a user is restricted from making an attempted toll call, the user will receive intercept treatment.

If the system is connected to a CO that uses a step-by-step switch, all seven digits which are normally dialed for a local call may not be required by the CO to route the call. For example, the central office 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 central office. This prevents users from bypassing code and toll restriction by dialing unneeded digits. For example, assume that the central office absorbs leading sevens before processing a number and that a toll-restricted user wants to dial a toll code of the form 1-201 plus seven more digits. The user can dial 77-1-201 plus seven more digits. The Toll Restriction feature will not recognize the call as a toll call and the central office will route the call. With digit absorption, the 77 is absorbed by the system before Toll Restriction is used. Thus, the call will be denied, as intended. Up to five digit absorption lists can be assigned.

If digit absorption is administered, 0/1 toll restriction is used to determine whether or not the call is allowed. The Toll Analysis table is not checked.

Considerations

Toll Restriction is used whenever it is necessary to restrict users at certain voice terminals from making toll calls. The customer can define what numbers are to be considered toll calls.

One toll list can be assigned per system.

Dialed Digit Strings administered as toll calls in the Toll Analysis table can be a maximum of 18 digits in length.

The Toll Analysis table can have a maximum of 1,000 dialed string entries. Each entry can be assigned to the toll list. However, the 1,000 entries in this table are also used to assign restricted and unrestricted call lists.

Interactions

None.

Administration

Toll Restriction is administered by the System Manager, and is normally enabled for each foreign exchange and central office trunk. This means that a toll-restricted user will not be able to make a toll call using the trunk group's TAC. If the System Manager wishes to allow toll-restricted users to place toll calls over a particular CO/FX trunk group, the `Toll Restricted` field should be set to "n." The `Toll Restricted` field is ignored if the originator of the call is not toll-restricted.

Toll Restriction is also administered by the System Manager to the following by the COR:

- 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

The Toll Analysis table must be administered by the System Manager to define which calls are toll calls.

The System Manager may also administer digit absorption lists containing absorption treatment of each digit zero through nine.

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 a code-restricted 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 CO that uses a step-by-step switch, all seven of the digits which are normally dialed for a local call may not be required by the central office to route the call. For example, the central office 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 central office. This prevents users from bypassing code and toll restriction by dialing unneeded digits. For example, assume that the central office absorbs leading sevens 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 central office 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. If digit absorption is administered, 0/1 toll restriction is used to determine whether or not the call is allowed. The Toll Analysis table is not checked.

International Toll/Code Restriction (G1.1SE)

The following information on implementing calling restrictions is only needed for G1.1SE systems. With G1.2SE, these methods for implementing calling restrictions are not needed because of the G3i's implementation of Multi-National Call Routing and Multi-National Toll Analysis, which provide for calling restrictions.

The USA Toll and Code Restriction features will work as documented in standard manuals only if both the public network dialing plan and the G1.1SE follow the North American Numbering Plan (that is, 7- and 10-digit dialing and other USA-specific restrictions). If the switch has a North American Numbering Plan but the CO doesn't, then the system administrator should proceed as described in the two methods outlined below.

The use of the USA Toll Restriction feature requires that long-distance and operator-assisted calls begin with a ''1'' or a ''0.'' The USA Toll Restriction feature is implemented by administering CO and FX trunk groups as toll restricted and then administering certain COR codes as toll restricted. Finally, stations assigned the COR whose calls are routed over the trunk groups are prevented from placing ''1'' or ''0'' calls unless the dialed digits are on the system's ''Allowed Calls List.'' As a result of using another dial plan, USA Toll Restriction cannot be implemented when long-distance and operator-assisted calls begin with digits other than ''1'' or ''0,'' respectively. (However, if the switch is being used in a country that doesn't use ''1'' or ''0'' as the first digits of toll calls, it is still possible for the switch to perform a kind of toll restriction using the two methods described below.)

The USA Code Restriction feature denies the calling party completion of outgoing calls to destinations whose numbers have selected initial three digits (within the local calling area), special service, and area codes. Code Restriction applies when a code restricted user (controlled by COR) accesses a CO or FX trunk group whose Restriction field is set to ''code.'' The system then checks appropriate Code Restriction tables to determine if the call will be allowed or sent to intercept. These Code Restriction tables can be for local calling areas (called HNPAs or Home Numbering Plan Areas in standard documentation) or for long-distance calling areas (called FNPAs or Foreign Numbering Plan Areas in standard documentation). Code Restricted calls are routed over CO or FX trunks which require 7- or 10-digit dialing strings. As a result, the USA Code Restriction feature cannot be implemented in systems with 3-, 5-, 6-, or 8-digit dial plans, for example, because the trunk group requires 7- or 10-digit dialing.

Why you can use the built-in USA Code Restriction feature only if you are in a country that uses 7- or 10-digit dial plans similar to the USA has been discussed. But in other countries, you can create practically the same kind of calling restrictions by one of the following two methods. Method 1 below describes a way to implement call restrictions and Method 2 describes a way to implement the handling of special numbers (such as USA service codes 411 and 811).

Method 1 Calling restrictions can be implemented using Automatic Route Selection and FRLs as described in the standard *Feature Description* and *Implementation* manuals. Translations should be implemented primarily using entries in the ARS FNPA table (for long-distance areas). While the meanings of the various dial plan entries may change from location to location, the basic format should remain intact. Restriction is accomplished by using the FRLs for the various ARS Routing Patterns as described in the *DEFINITY® Communications System Generic 3i — Implementation 555-230-650* manual. Call processing can be controlled as follows:

- **Local Calls** — Route calls based on the first three dialed numbers using the ARS HNPA table.
- **National Toll Calls** — Generally of the format 0NX...X. These calls route via the ARS FNPA table, entry 001 (labeled ‘‘Operator Assist’’ in the standard manual). At locations where 01X...X translations are required, ARS FNPA entry 012 should be set to the same pattern as entry 001.
- **International Toll Calls** — Generally of the format 00NX...X. These route by the ARS FNPA table entry 003 (labeled ‘‘IXC Operator Assist’’ in the standard manual). At locations where 000/001 call routing is required, where 00 is used by itself for international access and sometimes followed by a ‘‘1’’ (country code for USA and Canada), then the ARS FNPA entry 004 should be set to the same pattern as entry 003.

  ARS FNPA entry 000 and 002 should also be set to the same pattern as entry 003 to prevent unauthorized access to toll facilities.

Method 2 When using special numbers such as service codes, the numbers should be translated using Privileged Abbreviated Dialing lists as described in the standard documentation or by defining empty Hunt Groups and forwarding calls to those groups to the appropriate outside numbers.

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 central offices, area codes, and/or service codes.

The Allowed Calls List can include up to ten central office codes (that is, the first three digits of a seven-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 central office 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 ARS feature overrides Toll and Code Restriction. Permission or denial of ARS calls is determined by the FRL.

Administration

Toll or Code Restriction (but not both) is administered by the System Manager to each foreign exchange and central office trunk on a trunk group basis. A trunk group may also be administered with the `Restriction` field blank. This disables Code Restriction, but not Toll Restriction.

Toll or Code Restriction (but not both) is administered by the System Manager to the following by the COR:

- 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 0-9.

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 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.

Administration

Inward Restriction is administered by the System Manager to voice terminals by the COR 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 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 COR feature.

Hardware and Software Requirements

No additional hardware or software is required.

Restriction — Voice Terminal — Public

Description

Restricts callers at specified voice terminals from receiving public network calls. A denied call is routed to intercept tone, a recorded announcement, or the attendant.

Calls can redirect to an public-restricted voice terminal. The COR of the originally called extension number is the only one checked.

Considerations

Public Restriction is used whenever it is necessary that users at certain voice terminals receive only internal calls from other voice terminals or calls extended from the attendant.

Interactions

The following features interact with the Restriction — Voice Terminal — Public 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 public-restricted voice terminal, override the Inward Restriction.

- Tie Trunk Access

  Incoming dial repeating tie trunk calls can be completed directly to an public-restricted extension number.

- Transfer

  Incoming trunk calls can be transferred from an unrestricted extension number to an public-restricted extension number.

Administration

Public Restriction is administered by the System Manager to voice terminals by the COR feature.

Hardware and Software Requirements

Restriction — Voice Terminal — Public

Description

Considerations

Interactions

Administration

Hardware and Software Requirements

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 three-burst alerting signal (Priority Calling) when called back.

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-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 120 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 (two to nine 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:

- ARS

  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 two to nine 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 (three-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:
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 one-burst ring, two-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 three-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 a 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 notification. 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

Description

Allows rotary dialing voice terminals to be used with a system.

When a number is dialed at a rotary dialing voice terminal, the voice terminal out-pulses at a rate of 10 pulses per second. Each digit dialed sends out the corresponding number of pulses. For example, dialing a seven results in seven pulses being sent from the voice terminal. DEFINITY Generic 1 and Generic 3i 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.

Security Violation Notification (SVN)

Description

Notifies a designated referral point (attendant console or display-equipped voice terminal) of an attempt to breach System Management access via an invalid login or Remote Access via an invalid barrier code. Information specific to access violations is available through the Monitor Security Violations status report.

Referral Call Activation/Deactivation

Referral call activation is accomplished through the use of an SVN feature button on the designated station or attendant console. Activation, via the SVN feature button, will allow an SVN priority call to terminate to the referral destination whenever the administered threshold is reached. Deactivation, via the SVN feature button, will disable placement of the referral call. Activation of a SVN feature button results in the associated status lamp being lighted. Deactivation will extinguish the associated status lamp.

Repeated security violations may result in numerous referral calls being made in a short period of time. Deactivation of the appropriate SVN feature button will permit the user to halt the generation of these calls until the situation can be corrected.

The referral call for SVN login or Remote Access barrier code violations will appear at the administered attendant console or station as a priority call. When successfully terminated, the display associated with the call will indicate that the call is a Security Violation call. The display remains in effect until the call is dropped. For login violations, the message `Login Violation` will be displayed. For Remote Access violations, the message `Barrier Code Violation` will be displayed.

If a referral call is not answered within three minutes, it will time out. Referral calls that are not answered will be attempted again at the top of the hour. If the call terminates to an attendant, and all attendants are busy, the call will be queued like a regular attendant-seeking call.

Security Violations Status Report

The **monitor security violations** command (entered by the System Manager) provides a security violations report which contains current status information for invalid Manager I (G1) or DEFINITY Communications System Generic 3 Management Applications (G3-MA) login or Remote Access (barrier code) attempts. The data displayed by the **monitor security violations** command will update every 30 seconds. A total of 16 entries will be maintained for each type of invalid access (Manager I (G1) or G3 Management Applications and Remote Access) in the security violations report. The oldest information is overwritten by new entries.

The security violations report is divided into two distinct sections: System Management violations (Manager I (G1) and G3 Management Applications login violations) and Remote Access violations (barrier code violations). An example of this report is shown in the following screen.

```
-------------------------------------------------------------------------------
monitor security-violations                                               SPE A
-------------------------------------------------------------------------------
                         SECURITY VIOLATIONS STATUS     10:03 FRI APR 13 1990

SYSTEM MANAGEMENT VIOLATIONS                    REMOTE ACCESS VIOLATIONS
Time          Login    Port        Ext      Time          TG No.  Mbr  Ext

05/04 15:22  testid    NET-1                 05/03 21:14  3       50   81111
05/04 15:23  testit    NET-1                 05/04 22:10  21      43   83333
05/04 15:25  testx     NET-1                 05/04 22:13  21      43   83333
05/05 04:30  aaaaaaaa  NET-2       85454    05/04 22:15  21      43   83333
05/05 04:31  aaaaaaab  NET-2       85454
05/05 04:33  aaaaaaac  NET-2       85454
05/05 10:21  root      EPN
05/05 10:22  root      EPN
05/05 10:23  root      EPN
05/05 11:45  init      MGR1
05/05 11:46  init      MGR1
05/05 11:48  init      MGR1

-------------------------------------------------------------------------------

-------------------------------------------------------------------------------
enter command:
```

The following fields are shown on the report for System Management Violations:

- **Time:** The day and time that the violation occurred.
- **Login:** The invalid login ID that was entered as part of the login violation attempt. An invalid password may cause a security violation. If a valid login causes security violation by entering an incorrect password, the Security Violation Status report lists the login ID.
- **Port:** The port on which the failed login session was attempted. The following abbreviations are used:
  - **MGR1:** The dedicated Manager I (G1) or G3 Management Terminal connection (the EIA connection to the Maintenance board)
  - **NET-n:** The network controller dialup ports
  - **EPN:** The EPN maintenance EIA port
  - **INADS:** The INADS port
  - **EIA:** Other EIA ports

- **Ext:** The extension assigned to the data module that the failed login session was attempted on. This field is present only in the case where the System Manager's SAT is administered through a PDM/DTDM on the digital board.

The following fields are displayed on the report for Remote Access Violations:

- **Time:** The day and time that the violation occurred
- **TG No:** The trunk group number associated with the trunk where the remote access attempt terminated
- **Mbr:** The trunk group member number associated with the trunk where the remote access attempt terminated
- **Ext:** The extension used to interface with the Remote Access feature

Considerations

A maximum of one SVN feature button (per system) may be assigned for each component of the SVN feature (one for login security violations and one for Remote Access security violations).

An activated SVN feature button cannot be deleted by administration. The system will deny an attempt to delete an active SVN feature button.

Activation of an administered SVN feature button from a station set or attendant console will be prohibited if the appropriate SVN feature component (either login or Remote Access) is not enabled in the Feature-Related System Parameters screen.

A maximum of one Referral Destination may be administered per switch.

The Call Coverage, Call Forwarding, and Call Pickup (for example, within pickup group) features are not supported for SVN.

CAS attendants will not receive referral calls from branch locations.

Since SVN does not support a DCS network interface, Referral Destination must be local.

Unlike the ACA referral call, the SVN referral call is a priority call.

SVN does not currently support Speech Synthesis Circuit pack functionality.

Interactions

Need introductory text here.

- ACA

  The originating extensions for login and remote access security violations cannot be the same as the originating extensions for ACA long and short holding time originating extensions. The destination extensions for both features can be the same.

- Call Coverage

  SVN referral calls cannot be redirected to an alternate answering position in a call coverage path.

- Call Forwarding

  SVN referral calls cannot be forwarded since they are priority calls.

- Call Pickup

  SVN referral calls cannot be redirected to another voice terminal within a call pickup group.

- CAS

  When CAS is activated in a non-DCS environment, referral call destinations must be resident to the local switch. A ‘‘0’’ as a referral call destination is interpreted as the local attendant. If a local attendant does not exist, and ‘‘0’’ is supplied as a referral call destination, referral calls will not be placed.

  In all cases, in a non-DCS environment, the local attendant or designated voice terminal has control over activation and deactivation of a referral call. The CAS attendant will not receive referral calls from branch locations. Consequently, the CAS attendant will not have control of activation and deactivation of referral calls at branch locations.

- DCS

  This feature does not support a DCS network interface; referral call destinations must reside on the local switch.

- Night Service

  Login security violation referral calls will terminate to the night service extension when night service is in effect only if the night service extension is assigned to a display-equipped voice terminal or attendant console.

Administration

The SVN feature is activated when either or both of the `SVN Login Violation Notification Enabled` or `SVN Remote Access Violation Notification Enabled` fields on the Feature-Related System Parameters form are enabled.

When the SVN Login Violation Notification Enabled field is enabled, the following additional fields appear on the Feature-Related System Parameters screen:

- Originating Extension

  This field requires the entry of an unassigned extension that is local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for Manager I (G1) or G3 Management Applications login violations.

The originating extension initiates the referral call in the event of a login security violation. It also triggers the display of the appropriate alerting message on the referral destination's display module.

- Referral Destination

  This field requires the entry of a previously administered display-equipped voice terminal or attendant console. The destination may enable or disable the referral call through a feature button. A referral call will not be made to the administered destination if its corresponding feature button is deactivated.

  The Referral Destination is the extension assigned to the display-equipped voice terminal or attendant console that will receive the referral call in the event of a security violation.

- Login Threshold

  This field requires the entry of the minimum number of login violations that will be permitted before a referral call is made. The value of this field, in conjunction with the `Time Interval` field, will determine whether a security violation has occurred.

- Time Interval

  This field requires the entry of the time period that the login security violations must occur within. The range for the time interval is one minute to eight hours (00:01 to 07:59), and is entered in the form xx:xx. For example, if you want the time interval to be one minute, you would enter 00:01. If you want the time interval to be seven and one-half hours, you would enter 07:30.

When the `SVN Remote Access Violation Notification Enabled` field is enabled, the following additional fields appear on the Feature-Related System Parameters screen:

- Originating Extension

  This field requires the entry of an unassigned extension that is local to the switch and conforms to the dial plan for the purpose of originating and identifying SVN referral calls for Manager I (G1) or G3 Management Applications login violations.

  The originating extension initiates the referral call in the event of a login security violation. It also triggers the display of the appropriate alerting message on the referral destination's display module.

- Referral Destination

  This field requires the entry of a previously administered display-equipped voice terminal or attendant console. The destination may enable or disable the referral call through a feature button. A referral call will not be made to the administered destination if its corresponding feature button is deactivated.

The Referral Destination is the extension assigned to the display-equipped voice terminal or attendant console that will receive the referral call in the event of a security violation.

- Login Threshold

  This field requires the entry of the minimum number of login violations that will be permitted before a referral call is made. The value of this field, in conjunction with the `Time Interval` field, will determine whether a security violation has occurred.

- Time Interval

  This field requires the entry of the time period that the login security violations must occur within. The range for the time interval is one minute to eight hours (00:01 to 07:59), and is entered in the form xx:xx. For example, if you want the time interval to be 1 minute, you would enter 00:01. If you want the time interval to be seven and one-half hours, you would enter 07:30.

One SVN feature button (per system) may be assigned for each component of the SVN feature (one for login security violations and one for Remote Access security violations).

Hardware and Software Requirements

No additional hardware or 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 section.

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.

Send All Calls does not work with AutoCallback calls.

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 (or DTMF) 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

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 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 two-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 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.

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 COR 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 six-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.

- LWC

  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)
- COR (assigned on a per-agent basis to determine whether or not the agent can be service observed).
- Service Observing and/or ACD must be administered as a feature.

Hardware and Software Requirements

No additional hardware or software is required.

Single-Digit Dialing and Mixed Station Numbering

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 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 two-digit trunk access code 12 because a four-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 three- to four-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 10 through 79)
- 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 trunk access code and an extension number can only share a first digit if the extension number is shorter than the trunk access code.

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 timeout problems.

Extension numbers and feature access codes 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 UDP extensions.

Interactions

The following features interact with the Single-Digit 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. The Return Call button can be used to dial prefixed extensions, because the 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 PMS.

  Five-digit extensions cannot be exchanged with a PMS until modifications are made to the PMS interface.

- UDP

  The following limitations apply to a 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, trunk access codes, and feature access codes) 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, in the U.S., will always be defined as access to the hotel/motel operator (attendant).

Hardware and Software Requirements

No additional hardware or 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

Description

Provides modification of the dialed number so an AAR or 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 CO trunks to the serving central office

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 stand-alone 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 WATS serving office.
- Directly to a local CO or a 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 central office 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 EPSCS, Common CCSA office, or 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 DID and may be useful when a location has Remote Access but does not have DID or 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 — Can be administered in one of two ways. In the first way, delays outpulsing of subsequent digits for a preprogrammed interval (from 5 to 25 seconds) or, if TN748B Tone Detectors (TN420C, TN744 support A-law) are provided, until dial tone is received from the distant switch or the interval expires, whichever occurs first. In the second way, dial tone must be received before any outpulsing is done.
- Convert-to-tone — Causes all remaining digits to be outpulsed using tone signaling.

Use of these special characters is discussed 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 "wait for dial tone" character (a ‘‘+’’ symbol) is used to specify a longer interval with the option of sending or dropping after waiting a period of time for dial tone. The "time to wait" is the "off premises dial tone detect" time on the system parameters-features form. Receipt of dial tone automatically cancels the remainder of an interval when TN748B Tone Detectors (TN420C, TN744 support A-law) 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. If "outpulse without tone" on the system parameters country options form is set to "no", in which case the trunk is dropped and intercept tone is returned to the calling party.

Dial tones will also be heard if Network Feedback During Tone Detect is set to "no" on the system parameters country options form. Silence will be heard otherwise.

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, in the U.S., 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. Likewise in some countries, access to international trunks returns dial tone after dialing the international access code (for example, 00 in Belgium). Note that the use will not typically hear this dial tone, especially if Network Feedback is set to "no" on the system parameters miscellaneouse form.

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 five to 25 seconds (in increments of one 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 items such as trunk group usage, hunt group usage and efficiency, attendant group activity and efficiency, and security violations.

Individual reports are available for each of the following:

- Attendant Groups
- Automatic Circuit Assurance
- Automatic Route Selection
- Call Rate
- Call By Call Trunk Group
- DS1 Link Performance Measurements
- Hunt Groups
- Modem Pool Groups
- Performance Summary
- Processor Occupancy
- Security Violations
- System Status
- TDM Usage
- Tone Receiver
- Trunk Groups

All reports are on-demand reports. None are given automatically. Reports are available on the Manager I terminal (G1) or G3 Management Terminal, or a remote administration terminal. The reports can be printed if a printer is associated with the terminal. The reports can also be scheduled to print at the system printer via the Report Scheduler and System Printer feature.

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 Generic 3i — System Reports, 555-204-510*.

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 CDR feature. Processed CDR 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

Description

Allows the user to view data associated with attendants, major and minor alarms, and traffic measurements. The information is displayed on the Manager I terminal (G1) or G3 Management Terminal, and presents a basic picture of the System 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 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 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 CO call enters a full ACD split queue, CDR and the CMS may show different measurements. CDR 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 TEG or PCOLG to bridge onto an existing group call. If a call has been answered using the Call Pickup feature, the originally called party 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 TEG or PCOLG 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 answer 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 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. 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.

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.

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).

Hardware and Software Requirements

No additional hardware or software is required.

Ten-Digit to Seven-Digit Conversion (G1.1)

Description

The Ten-Digit to Seven-Digit Conversion feature allows users to dial certain 10-digit and seven-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 seven-digit and 10-digit DDD numbers as the numbers of seven-digit ETN location (RNX) numbers. The seven-digit or 10-digit numbers will be replaced by seven-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 AAR and ARS features.

AAR

AAR provides alternate routing choices for private on-network calls and also allows on-network calls to route through the public network when on-network routes are unavailable. 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.

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.

DDD Numbers

A DDD number is a public network number with three parts:

- A three-digit area code
- A three-digit CO (Exchange) code
- A four-digit extension (endpoint) number.

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 CO (Exchange) in the Home Area Code preceded by the digit 1, if required.

- Ten-Digit DDD Calls

  The digit 1 (if required), an Area code, an Exchange code, and a station number.

- Calls to IXC

  A Carrier Access Code, followed by one of the two call types just described. Carrier Access Codes are of the form 10XXX, where XXX are digits to identify a particular long distance carrier.

Ten-digit DDD 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 2-37). 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 seven-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 seven-digit number is dialed after the AAR access code, the call is treated as if it was a seven-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 2-37. Ten-Digit to Seven-Digit Conversion Table Example

Ten-Digit to Seven-Digit Conversion Table	
Match	Replace
...	...
201-957-5	222-5
...	...
212-976-1313	0
212-976-2525	222-0111
...	...
303-526	362
303-538-2	374-3
303-538-3	374-0
...	...
...	...
303-538-56	374-59
...	...
...	...
303-538-975	374-970
...	...
...	...
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 2-23.
- 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 LDN 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 LDNs 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 Ten-Digit to Seven-Digit 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:

- 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.

- 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.

- CDR

  Even if a call is altered by Ten-Digit to Seven-Digit Conversion, the actual digits dialed are recorded by CDR.

  CDR 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 CDR record is not generated.

Administration

The only administrable item unique to the Ten-Digit to Seven-Digit Conversion feature is the Conversion Table (see Table 2-23). 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 UDP is activated.

Hardware and Software Requirements

No additional hardware is required.

ARS software and either Private Network or UDP 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 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 answer 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 four 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 TEG 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 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.

- DDC and UCD

  A TEG cannot be a member of a DDC or UCD 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 call. 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.

- LWC

  LWC 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

TEGs 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.

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 ARS 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.

Time of Day Routing

Description

Provides the most economical routing of ARS and 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 Time of Day Plan Number assigned by the COR feature. A Time of Day Routing Plan (see the following screen) can be administered for each of the eight Time of Day Plan Numbers. 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 Time of Day Plan Number.

```
                                                                   Page 1 of 1
                                     TIME OF DAY ROUTING PLAN 1
      Act    PGN    Act    PGN    Act    PGN    Act    PGN    Act    PGN    Act    PGN
      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      __:__  _      __:__  _      __:__  _      __:__  _      __:__  _
```

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 of eight) is assigned an activation time and a RPN (shown as ‘‘PGN #’’ on the Manager I terminal (G1) or G3 Management Terminal screen form). (PGN # is the Partition Group Number.) With G1.1, each RPN has a complete and distinct group of HNPA, FNPA, and location code (RNX) tables to be used to route AAR/ARS calls. With G3i, the RPN selects the ARS or AAR Digit Analysis Table associated with the RPN to be used to route the 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 Time of Day Plan Number assigned to the user's 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 (‘‘PGN #’’ on the Manager I (G1) or G3 Management Applications screen). The system then uses the HNPA, FNPA, or RNX table (G1.1) or Digit Analysis Table (G3i) 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, ISDN-PRI trunks, and trunks used for call forwarding to external numbers.

For additional information on AAR and ARS, 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 Time of Day Plan Number of 3.
- The Time of Day Routing Plan table for Time of Day Plan Number 3 is administered as shown in the following screen.

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 Time of Day Plan Number assigned to Jim’s COR 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 Time of Day Plan Number 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 (G1.1) or Digit Analysis Table (G3i) associated with RPN 2. Therefore, these 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 RPN 1.

```
                                                                  Page 1 of 1
                              TIME OF DAY ROUTING PLAN 3
     Act    PGN    Act    PGN    Act    PGN    Act    PGN    Act    PGN    Act    PGN
     Time    #     Time    #     Time    #     Time    #     Time    #     Time    #
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      __:__  _      __:__  _      __:__  _      __:__  _      __:__  _
```

Overriding the Time of Day Routing Plan

An attendant or a voice terminal user with console permission and a display can temporarily override the activating user's current routing plan. This can be accomplished by either of two methods:

- Immediate Manual Override
- Clocked Manual Override

Both types of override are discussed 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 (G1) or G3 Management 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 immediately upon activation, and remains in effect for the activating user's Time of Day Plan Number (and all who share this Time of Day Plan Number) 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 (one to eight) 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 Time of Day Plan Number.

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 Time of Day Plan Number 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 three-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 three-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 Time of Day Plan Number.

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 Time of Day Plan Number 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 and ARS 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 Time of Day Plan Number. 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 Time of Day Plan Number 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 Time of Day Plan Number of the attendant's COR.

- Authorization Codes

  If a user's 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.

- AAR

  When Time of Day Routing is assigned, all AAR calls use the Time of Day Routing Plans for routing calls.

- 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 Time of Day Plan Number 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 Time of

Day Plan Number of the calling party is used to route the call.

- DCS

  Care should be taken when making Time of Day Routing assignments in a DCS environment. Depending on a user's Time of Day Plan Number, 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 Time of Day Plan Number 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 Time of Day Plan Number 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 Time of Day Plan Number of the Barrier Code and/or Authorization Code that was entered is used for routing the call.

- CDR

  Normal CDR records are generated for AAR/ARS calls on trunks administered for CDR. However, information about the Time of Day Plan Number used to route the call is not provided.

- 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 Time of Day Plan Number is used to route the call.

Administration

Time of Day Routing must be activated on the System Parameter-Customer Options screen. 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.
- With G1.1, different FNPA, HNPA, and RNX tables must be administered for each Time of Day Plan Number.
- With G3i, a different Digit Analysis Table must be administered for each Time of Day Plan Number.

- A Time of Day Routing Plan must be administered for each Time of Day Plan Number.
- A Time of Day Plan Number must be assigned to each COR table. Up to eight Time of Day Plan Numbers 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.

Hardware and Software Requirements

No additional hardware or software is required.

Timed Reminder and Attendant Timers

Description

The Timed Reminder feature automatically alerts the attendant after an administered time interval 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.

Attendant timers are important for attendant-intensive settings, like those without Direct Inward Dialing. The attendant timers are:

- Unanswered DID Call Timer - routes the call to the administered "DID/TIE/ISDN Intercept Treatment" if the DID call goes unanswered for the length of the timer.
- Attendant Return Call Timer - Unanswered calls that have been extended and then released from an attendant console return to the same attendant position that released it.
- Attendant Timed Reminder of Held Call Timer - allows the administration of an interval to be used when determining whether a held call has held too long.
- Attendant No Answer Timer - allows an administrable amount of time to be specified and used in determining whether an incoming call to the attendant has not been answered.

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.

If a call has been routed to each attendant and it remains unanswered, the system will be placed into night service. If this is happening more frequently than is optimal, the answering intervals may be lengthened by the system administrator.

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.

- 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 and Attendant Timers are administered on a per-system basis by the System Manager. The screen required is console parameters; the fields required to change the default timers are No Answer Timeout and Alerting under the heading Incoming Call Reminders.

Hardware and Software Requirements

No additional hardware or software is required.

World Class Tone Detection

Description

World Class Tone Detection allows the system to identify and handle different types of call progress tones, depending on the system adminstration.

Considerations

When the administered Level of Tone Detection is "medium" or "broadband", multiple-line Data Terminal Dialing is disabled.

Interactions

The following feature interacts with the World Class Tone Detection feature:

- Data Call Setup

  Multiple-Line Data Terminal Dialing is supported ONLY if the adminstered Level of Tone Detection is "precise".

Administration

Administration of World Class Tone Detection consists of two distinct parts: tone detection level and tone detection algorithm. The Level of Tone Detection is administered by the System Manager on page 2 of the system-parameters miscellaneous form. The tone detection algorithm is adminstered by the craft, the local installer or a remote installer on page 1 of the system-parameters country-options form. The following items can be administered: Tone Detection Mode, Dial Tone Validation Timer, and Interdigit Pause.

Hardware and Software Requirements

Tone Detection Modes 1, 2, & 3 are meaningful only if the system tone detectors are TN420Bs or greater. Modes 4 & 5, the Dial Tone Validation Timer, and the Interdigit Pause are meaningful only if the system tone detectors are TN420Cs or greater.

World Class Tone Generation

Allows administrators to specify the base call progress tone set to be generated by the system and to then customize the set by selecting different values for frequency and cadence for up to 6 individually administerable tones.

Considerations

If a particular tone (for example, conference tone) is not defined for the administered base tone generation set (and not administered via the individual tone administration), silence will be used.

Interactions

None.

Administration

The base tone generation set and individual tones, along with companding mode, analog ringing cadence, and digital loss plan fields are administrable on pages 1-7 of the system-parameters country-options form by the craft, the local installer and a remote installer.

Hardware and Software Requirements

Entries in the Base Tone Generation Set and individual tone fields are meaningful only if the system tone generators are TN780s, vintage 4 or greater. An entry of "8" (or of a code which maps internally to 8) in the Base Tone Generation Set field is meaningful only if the system tone generators are TN780s, vintage 5 or greater.

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.

Transfer is also known as Push Transfer. Please see the Pull Transfer feature for a description a type of transfer that can be used by voice terminal users to "pull" a held call to their own extension.

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.

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 the treatment received on transferred trunk calls.

Hardware and Software Requirements

No additional hardware or software is required.

Trunk Flash

Description

Trunk Flash enables multi-function voice terminals to access ‘‘central office customized services’’ that are provided by the far-end/CO located directly behind the Generic 3i system. CO customized services are electronic features (such as conference and transfer) that are accessed by a sequence of flash signals and dialing from the Generic 3i station on an active trunk call. The Trunk Flash feature can help to reduce the number of trunk lines connected to the Generic 3i system by:

- performing trunk-to-trunk call transfers at the far-end/CO, which eliminates the use of a second trunk line for the duration of the call and frees the original trunk line for the duration of the call.
- performing a conference call with a second outside call party, which eliminates the need for a second trunk line for the duration of the call.

Considerations

Generic 3i supports the Trunk Flash signal for incoming, outgoing, or two-way call directions on selected two-wire analog (ground-start or loop-start) or digital (DS1) trunks.

Access to this feature is restricted to trunk ‘‘Group Types’’ of ‘‘co,’’ ‘‘fx,’’ and ‘‘wats’’ with the `Trunk Flash` field enabled.

All types of multifunction voice terminals that can be assigned a Flash button may have access to the Trunk Flash feature.

The Trunk Flash feature is unavailable on analog voice terminals, attendant consoles, or PCOL.

The Flash button is used by the Trunk Flash and CAS features.

Generic 3i features (such as internal conference call, transfer, and call park) may be combined with custom services. However, mixing Generic 3i features with custom services makes the call complicated for the user to track because the Generic 3i system cannot give the local station user status information on the custom services.

The Trunk Flash feature may only be accessed if the call has only one trunk line, and the trunk group of that line has ‘‘Trunk Flash’’ enabled. The Trunk Flash feature is disabled when the call involves more than one trunk line, even if all the trunk lines have ‘‘Trunk Flash’’ enabled.

Any PBX station with a Flash button may access the ‘‘Trunk Flash’’ feature. There may be up to five PBX stations involved in a PBX conference call with the trunk line party. However, to access the Trunk Flash feature, at least one of the PBX stations must have a Flash button.

In a call involving more than one PBX station, one station may press the Flash button, and another station may dial the phone number. The station that dials the phone number is not required to have a Flash button.

There must be no other trunk connection between the PBX that the station is connected to, and the far-end/CO supplying the customized services. If the call connection passes over a ‘‘tie’’ trunk, the station will not have a direct connection to the far-end/CO and, as a result, will not have access to the far-end/CO custom services.

If the far-end/CO does not support customized services, the call may be dropped by the far-end/CO on sending the flash signal or the signal may be ignored and a ‘‘click click’’ sound will be heard.

Interactions

The Trunk Flash feature may be combined with other Generic 3i features.

Administration

The System Manager must perform the following tasks:

- Administer a station form for each voice terminal that will be authorized for Trunk Flash. Each authorized voice terminal must be assigned a Flash button.
- Administer the `Trunk Flash`? field in the appropriate Trunk Group form(s) for CO, FX, and/or WATS group types.

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 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. Twelve 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 six 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 never have Warning lamps.

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.

Administration

This feature is administered by the System Manager. 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

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 conferenced 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 (two-digit) for that trunk group and the trunk group member number (two-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.

The Trunk Identification By Attendant feature is denied if there are exactly two trunks on the call, and the station pressing the Trunk ID button is not the controlling party.

In the case of a conference resulting from an incoming call followed by an outgoing call, the last trunk added to the conference is the incoming one.

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.

Transferred trunk calls can be administered to receive either music or silence.

Some Central Office trunks do not signal the PBX when the Central Office user disconnects from the call. The system assures that incoming Central Office trunks without Disconnect Supervision are not transferred to outgoing trunks, or other incoming Central Office trunks without Disconnect Supervision.

An attendant-assisted call connecting an outgoing trunk or incoming trunk without Disconnect Supervision 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 or an outgoing call and an incoming call without Disconnect Supervision, the user must remain on the call. Otherwise, the call will be dropped. An incoming trunk with Disconnect Supervision 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 if one of the incoming trunks has Disconnect Supervision.

The Trunk-to-Trunk Transfer option on the Feature Related System Parameters form has no affect on tie trunks. Trunk-to-Trunk Transfer only affects calls where both trunks are CO, FX, WATS, DID, or CPE.

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 (UDP)

Description

Provides a common four- or five-digit dial plan that can be shared among a group of switches. Interswitch dialing and intraswitch dialing both require four- or five-digit dialing. The UDP is used with ETN, Main/Satellite/Tributary, and DCS configurations. Additionally, UDP can be used alone to provide uniform four- or five-digit dialing between two or more private switching systems without ETN, Main/Satellite/Tributary, or DCS configurations.

In a UDP, the first one, two, three, or four digits of the four- or five-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 2-21. In this figure, a five-digit UDP is used in an ETN. Three switches are included in the UDP. Each switch has an assigned RNX and a prefix code (discussed 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.

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, 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 four- or five-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 five-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 2-21, 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 four- or five-digit extension number.

Since extensions beginning with 0 may cause problems in a network environment, administrators are discouraged from using this number as the leading digit when assigning extensions.

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 four- or five-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 two-digit and three-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 UDP feature:

- 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 seven-digit AAR number will provide the exact same routing as the UDP.

- DID

  DID calls to five-digit UDP extension numbers require that the DID trunk group insert enough digits to make a five-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 four- or five-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.

  When administering the dial plan and designating a group of extensions as UDP non-local, the system does not check to see is any local extensions match the UDP extension. This allows some flexibility in the changing of extensions from local to non-local. However, after the dial plan is changed to make an extension UDP, nothing can be administered with these extensions on the local switch.

- 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 four- or five-digit extension numbers
- PBX Codes (expands first one, two, three, or four digits of dialed extension to an RNX)
- RNX Table (G1.1) (used by AAR to route calls to the correct switch)
- AAR Analysis Table (G3i) (used by AAR to route calls to the correct switch)
- Routing Patterns

Hardware and Software Requirements

A Processor Interface circuit pack is required for DCS applications. DCS or UDP software is required.

Visually Impaired Attendant Service (VIAS)

Description

This feature is new for G3i and provides voice feedback to a visually impaired attendant in either Italian or British English. Each voice phrase is a sequence of one or more single voiced messages.

Six new attendant buttons are defined for the VIAS feature:

- Visually Impaired Service Activation/Deactivation Button: this button activates or deactivates the feature for the console from which it was pressed. When VIAS is activated, an indicator lamp will light next to the button. In addition, all ringers which were disabled (for example, recall, incoming calls, and so on) will be enabled.
- Console Status Button: this button allows the visually impaired attendant to determine whether the console is in Position Available or Position Busy state, whether the console is a night console, the status of the attendant queue, and the status of system alarms.
- Display Status Button: this button allows the visually impaired attendant to determine what is shown on the console display. The only exceptions to this are class of restriction information, personal names, and a few call purposes.
- Last Operation Button: this button voices the last operation performed.
- Last Voiced Message Button: this button allows the visually impaired attendant to retrieve the last voiced message.
- Direct Trunk Group Selection Status Button: this button allows the visually impaired attendant to obtain the status of an attendant monitored trunk group.

The visually impaired attendant may use the Inspect mode locate each button and determine the feature assigned to each without actually executing the feature. To do this, the attendant presses the Inspect button and then presses each button in turn and listens to the voiced information about it. Afterwards, the attendant presses the Normal button to end Inspect mode.

Considerations

Some changes on the attendant console will be automatically voiced, for example, alarms reported, night service activated, and call threshold reached.

After system initialization, VIAS will not be automatically activated. After a warm restart operation, VIAS will remain activated if it was already activated. After recovery and cold restart operations, VIAS will not be automatically activated even if it was activated before the recovery or restart attempt. Finally, whenever the attendant console goes to a busyout state and VIAS is activated, VIAS will be automatically deactivated.

Interactions

When VIAS is activated, Auto Start is always enabled. The attendant can activate the Don't Split feature as normal if VIAS is activated, however, the Don't Split feature, if currently activated, is automatically deactivated when the attendant deactivates VIAS.

Administration

VIAS is turned off by default. Before an attendant can use the Activate/Deactivate button, or any of the VIAS buttons, these buttons must be administered with the attendant administration form.

Hardware and Software Requirements

This feature requires at least one Speech Processor circuit pack to be installed into a system port carrier since VIAS capabilities are performed with speech synthesis messages that are voiced to the attendant.

Voice Message Retrieval

Description

Allows attendants, voice terminal users, and remote access users to retrieve LWC and Call Coverage messages in the form of a voice output. Voice choices are UK and US English, Italian; these choices depend on your system hardware.

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.
- 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 **NEXT** 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.
- ''[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:

- 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 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.

- LWC

  Voice Message Retrieval enhances 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.

Terminal users may select a display message language. The language choices are English (default), French, Italian, or Spanish. Please see the description under Multi-Language Displays for more information.

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 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 lowercase 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

  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

  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:

a=3602

then

a= TOM BROWN 3062

or

a= EXT 3602 3062

- Outgoing trunk call:

  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:

  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 500 display modules can be provided per system.

Certain voice terminals and the attendant group can be designated for system-wide message retrieval. Users of these voice terminals or consoles can retrieve LWC and Call Coverage messages for other voice terminal users including DDC groups, UCD groups, PCOL groups, and TEGs. Selected users cannot retrieve messages for other selected users. Up to 10 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.

If you are using a 7506D or 7507D to make calls that require additional digits, a comma may appear in the dial sequence after you receive second dial tone or after the call has been set up. The comma is used to separate the called number from subsequent information.

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

  If prefixed extensions are used in the system's dial plan, the prefix is not displayed when the extension is displayed. The Return Call button can be used to dial prefixed extensions, because the 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

DCP services require a display-equipped voice terminal and one port on a TN754 Digital Line circuit pack (TN413, TN754B support A-law). No additional software is required.

ISDN-BRI services require the following hardware:

- TN778 Packet Control circuit pack, which provides the interface to the LAN (packet) bus on G3i for establishing the signaling connectivity
- TN556 BRI Port circuit pack, which is the Basic Rate Line circuit pack. Each BRI port board can support 12 line interfaces, each operating at 192 kbps.
- AT&T ISDN 7506 and 7507 voice terminals

Voice Terminal Flash Timing

Description

Please see the description for Recall Signalling.

World Class Tone Detection

Description

World Class Tone Detection allows System 75 to identify and handle different types of call progress tones, depending on the system adminstration.

Considerations

When the administered Level of Tone Detection is "medium" or "broadband", multiple-line Data Terminal Dialing is disabled.

Interactions

The following feature interacts with the World Class Tone Detection feature.

- Data Call Setup

  Multiple-Line Data Terminal Dialing is supported ONLY if the adminstered Level of Tone Detection is "precise".

Administration

Administration of World Class Tone Detection consists of two distinct parts: tone detection level and tone detection algorithm.

The Level of Tone Detection is administered by the System Manager on page 2 of the system-parameters miscellaneous form.

The tone detection algorithm is adminstered by the craft, the local installer or a remote installer on page 1 of the system-parameters country-options form. The following items can be administered: Tone Detection Mode, Dial Tone Validation Timer, and Interdigit Pause.

Hardware and Software Requirements

Tone Detection Modes 1, 2, & 3 are meaningful only if the system tone detectors are TN420Bs or greater. Modes 4 & 5, the Dial Tone Validation Timer, and the Interdigit Pause are meaningful only if the system tone detectors are TN420Cs or greater.

World Class Tone Generation

Description"

Allows administrators to specify the base call progress tone set to be generated by the system and to then customize the set by selecting different values for frequency and cadence for up to 6 individually administerable tones.

Considerations

If a particular tone (for example, conference tone) is not defined for the administered base tone generation set (and not administered via the individual tone administration), silence will be used.

Interactions

None.

Administration

The base tone generation set and individual tones, along with companding mode, analog ringing cadence, and digital loss plan fields are administrable on pages 1-7 of the system-parameters country-options form by the craft, the local installer and a remote installer.

Hardware and Software Requirements

Entries in the Base Tone Generation Set and individual tone fields are meaningful only if the system tone generators are TN780s, vintage 4 or greater. An entry of "8" (or of a code which maps internally to 8) in the Base Tone Generation Set field is meaningful only if the system tone generators are TN780s, vintage 5 or greater.

System Parameters

3

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 discussed under each applicable feature.

Feature Administration

Administration Not Required

Administration is not required to activate the following features but many of them can be customized with administration options:

1. Attendant Auto-Manual Splitting
2. Attendant Call Waiting
3. Attendant Recall
4. Attendant Release Loop Operation
5. Automatic Incoming Call Display
6. Conference — Attendant
7. Conference — Terminal
8. Hold
9. Line Lockout
10. Move Agents from CMS

11. Recall Signaling
12. Recent Change History
13. Senderized Operation
14. Straightforward Outward Completion
15. Temporary Bridged Appearance
16. Through Dialing
17. Touch-Tone Dialing (for Terminals)
18. Transfer

Administration Required

Administration is required to activate the following features:

1. AAR/ARS Partitioning
2. Abandoned Call Search
3. Abbreviated Dialing
4. Administered Connections (G3i)
5. Administration Without Hardware (G3i)
6. Alphanumeric Dialing (G3i)
7. Agent Call Handling
8. Attendant Control of Trunk Group Access
9. Attendant Direct Extension Selection With Busy Lamp Field
10. Attendant Direct Trunk Group Selection
11. Attendant Display (Buttons only)
12. Audio Information Exchange (AUDIX) Interface
13. Authorization Codes
14. Automatic Alternate Routing
15. Automatic Callback
16. Automatic Call Distribution
17. Automatic Circuit Assurance
18. Automatic Route Selection
19. Automatic Wakeup
20. Basic Call Management System
21. Bridged Call Appearance — Multi-Appearance Voice Terminal
22. Bridged Call Appearance — Single-Line Voice Terminal

23. Busy Verification of Terminals and Trunks
24. Call by Call Service Selection
25. Call Coverage
26. Call Forwarding All Calls
27. Call Park
28. Call Pickup
29. Call Prompting (G3i)
30. Call Vectoring (G3i)
31. Call Waiting Termination
32. Centralized Attendant Service
33. Code Calling Access
34. Customer-Provided Equipment (CPE) Alarm
35. Data Call Setup
36. Data Hotline
37. Data-Only Off-Premises Extensions
38. Data Privacy
39. Data Restriction
40. Default Dialing (G3i)
41. Dial Access to Attendant
42. Digital Multiplexed Interface
43. Direct Department Calling
44. Direct Inward Dialing
45. Direct Inward Dialing and Direct Outward Dialing — International (G1.2SE)
46. Direct Outward Dialing
47. Distinctive Ringing
48. Do Not Disturb
49. DS1 Tie Trunk Service
50. EIA Interface
51. Emergency Access to the Attendant
52. Facility and Non-Facility Associated Signaling (G3i)
53. Facility Busy Indication
54. Facility Test Calls
55. Forced Entry of Account Codes

56. Generalized Route Selection
57. Hot Line Service
58. Hunting
59. Inbound Call Management (G3i)
60. Individual Attendant Access
61. Information Systems Network (ISN) Interface
62. Integrated Directory
63. Integrated Services Digital Network — Basic Rate Interface
64. Integrated Services Digital Network — Primary Rate Interface (G3i)
65. Intercept Treatment
66. Intercom — Dial
67. Inter-PBX Attendant Calls
68. Intraflow and Interflow
69. Last Number Dialed
70. Leave Word Calling
71. Look Ahead Interflow (G3i)
72. Loudspeaker Paging Access
73. Loudspeaker Paging Access — Deluxe
74. Manual Message Waiting
75. Manual Originating Line Service
76. Manual Signaling
77. Modem Pooling
78. Multi-Appearance Preselection and Preference
79. Multiple Listed Directory Numbers
80. Music-on-Hold Access
81. Names Registration
82. Night Service
83. Off-Premises Station
84. PC/PBX Connection
85. Permanent Switched Calls
86. Personal Central Office Line
87. Personalized Ringing
88. Power Failure Transfer

89. Priority Calling
90. Privacy — Attendant Lockout
91. Privacy — Manual Exclusion
92. Property Management System Interface
93. Queue Status Indications
94. Recorded Announcement
95. Recorded Telephone Dictation Access
96. Remote Access
97. Report Scheduler and System Printer
98. Restrictions
99. Ringback Queuing
100. Ringer Cutoff
101. Rotary Dialing
102. Security Violation Notification (G3i)
103. Service Observing
104. Single-Digit Dialing and Mixed Station Numbering
105. CDR Account Code Dialing
106. Call Detail Recording
107. Ten-Digit to Seven-Digit Conversion (G1.1)
108. Terminating Extension Group
109. Timed Reminder
110. Time of Day Routing
111. Touch-Tone Dialing (for Trunks)
112. Trunk Flash
113. Trunk Group Busy/Warning Indicators to Attendant
114. Trunk-to-Trunk Transfer
115. Uniform Call Distribution (see Direct Department Calling)
116. Uniform Dial Plan
117. Voice Message Retrieval
118. 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
  Agent Log-In
  Agent Log-Out
  Manual-In
  Auto-In
  After Call Work
  Auxiliary Work
  Assist
- AP Demand Print
- Automatic Route Selection
- Automatic Callback (activate and deactivate) (applies to single-line voice terminals only)
- Automatic Wakeup
  Wakeup Call
  Verify Wakeup Announcement
- Call By Call Service Selection
- 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
- Emergency Access to the Attendant

- Facility Test Calls
- Generalized Route Selection
- 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
- Private Network Access
- Priority Calling
- Property Management System Interface
- Public Network Access
- Recorded Telephone Dictation Access
- Send All Calls (associated with Call Coverage)
- CDR Account Code Dialing
- Trunk Answer From Any Station (associated with Night Service)
- Voice Message Retrieval
  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
- Busy Verification of Terminals and Trunks
- 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
  Message Waiting Notification (Activate)
  Message Waiting Notification (Deactivate)
  Checkout
- Queue Status Indications
  NQC (number of queued calls)
  OQT (oldest queued time)
  AQC (attendant queued calls)
  AQT (attendant queued time)
- Ringer Cutoff

- Service Observing
- 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
  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
  Manual-In
  Auto-In
  Auxiliary Work
  After Call Work
  Assist
  Release
- Automatic Wakeup:
  Auto Wakeup Entry
  Failed Messages Wakeup
- Call Park and Call Park Answer Back
- Call Pickup
- Do Not Disturb
- Emergency Access to the Attendant
- 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 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 list of system maximum parameters for hardware and software items are listed below and on the following pages for the multi-carrier cabinet and the single-carrier cabinet DEFINITY Communications System Generic 1. These parameters apply to both the multi-carrier cabinet and single-carrier cabinet systems unless otherwise noted.

Table 3-1. Maximum System Parameters for Hardware and Software Items

ITEM	G1	G1.2	GD91
Abbreviated Dialing (AD)			
AD lists per system	1,600	2,000	2,400
AD list entry size	24	24	24
AD entries per system	8,000	10,000	12,000
Personal Lists	1,600	2,000	2,400
Max. entries	10	10	10
Per extension	3	3	3
Group Lists	100	100	100
Max. entries	90	90	90
Per extension	3	3	3
System Lists	1	1	1
Max. entries	90	90	90
Enhanced Lists	1	1	1
Max. entries	1,000	1,000	1,000
Applications Adjuncts			
BX.25 Links (SCC/MCC)	4/8	4/8	8
MSA adjuncts	1	1	1
AUDIX adjuncts	1	1	1
CMS adjuncts	1	1	1
CallVisorISDN Gateway*	1	1	1
CallVisor ASAI Adjuncts	None	8	8
Asynchronous links (RS232)	5	5	5
CDR output devices	2	2	2
Journal printer	2	2	2
System printer	1	1	1
Property Management Systems	1	1	1
BX.25 processor channels	64	64	64
Hop channels	64	64	64

* The system can have either an CallVisor ASAI or a CallVisor ISDN Gateway. Both cannot co-reside on the same switch.

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
ARS/AAR			
Patterns	254	254	254
ARS patterns for measurement	20	20	20
Preferences in an ARS pattern	6	6	6
Entries in RNX table	640	None	None
Entries in FNPA table	200	2,000	2,000
RHNPA tables	32	32	32
Toll tables	32	32	32
UDP (Entries)	240	240	240
Choices per RHNPA table	12	12	12
Entries in TOLL table	800	800	800
Entries in HNPA/RHNPA tables	800	1,000	1,000
FRLs	8	8	8
Inserted digit strings*	1,200	1,200	1,200
Digits inserted for ARS/AAR	36	36	36
Digits deleted for ARS/AAR	11	18	18
Routing plans	8	8	8
TOD charts	8	8	8
ARS/AAR Table Entries (NPA,NXX,RXX,HNPA,FNPA)	None	2,000	2,000
Digit Conversion Entries ARS/AAR	180	300	300
ASAI			
ASAI adjuncts	None	8†	8
Monitored splits/VPNs	None	170	170
Simultaneous active adjunct controlled calls	None	300	300
Switch to adjuncts associations	None	127	127
Maximum adjunct users	None	40	40
Active station controlling associations	None	2,000	2,000
Station controllers per station	None	2	2
Call controllers per call	None	1	1
Call monitors per call	None	14	14

* The number of 12 character digit strings available for AAR/ARS preferences.

† One link is sufficient to support Call Visor ASAI at full capacity.

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
Attendant Service			
Attendant positions (day/night)	6/1	6/1	15/1
Switched loops/console	6	6	6
Attendant controlled restriction groups	64	64	64
100’s groups/attendant console	20	20	20
Queue length	30	30	80
Emergency access queue length	50	50	50
Other Access Queues			
Maximum number of queues	1	1	12
Maximum number of queue slots	50	50	80
Size of reserved queue	None	None	2-75
Reserved queue default size	None	None	5
Centralized Attendant Service			
Release link trunk groups at branch	1	1	1
Release link trunks at branch	99	99	99
Release link trunk groups at main	99	99	99
Release link trunks at main	400	400	400
Branches per Main	99	99	99
Authorization			
Classes of Restriction	64	64	96
Classes of Service	16	16	16
Authorization codes	5,000	5,000	5,000
Length of authorization code	4-7	4-7	4-7
Remote access barrier codes	10	10	10
Length of barrier code	4-7	4-7	4-7
Toll Call Lists	None	1	1
Restricted Call Lists	None	1	1
CDR Forced Entry Account Code List	1	1	1
Length of Forced Entry Account Code digits	1-15		1-15
Unrestricted/Allowed Call Lists	1	10	10
Total Call List Entries	10	1,000	1,000
Automatic Callback Calls	160	160	240

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
Automatic Wakeup			
Wakeup requests per system	1,600	1,600	2,400
Wakeup request per extension	1	1	1
Wakeup requests per 15 min time interval	300	300	300
Advance Wakeup Request Time			
(Hours)	23	23	23
(Minutes)	55	55	55
Simultaneous display requests	10	10	10
Bridged call appearances	1,600	1,600	2,400
Cabinets			
MCC System			
Basic/PPN	1	1	1
EPN	1	2	2
SCC System			
Basic/PPN Control	1	1	1
Duplicated PPN Control	2	2	2
Port			
w/o Duplication	3	3	3
With Duplication	2	2	2
Basic/EPN Control	1	2	2
Duplicated EPN Control	2	4	4
Port			
w/o Duplication	6	3	3
with Duplication	4	4	4
Call Coverage			
Coverage paths	600	600	600
With Hospitality Parameter Reduction	5	5	5
Coverage points in a path	3	3	3
Coverage paths linked together	4	4	4
Coverage paths included in Call Coverage Report	100	100	100
Coverage Answer Group (CAG)	200	200	500
Members per CAG	8	8	8
Maximum users per coverage path	2,900	2,900	3,500

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
Call Detail Recording			
Number of CDRU adjuncts per system	None	1	1
CDRU trackable extensions	None	1,600	2,400
Intra-switch call trackable extensions	None	100	100
Call Forwarding			
Call forwarded numbers	1,600	1,600	2,400
Call forwarded digits	16	16	16
Call Park			
Attendant group common shared extension numbers	10	10	10
Number of Parked Calls	482	723	723
Call Pickup Groups			
Call pickup groups	800	800	800
With Hospitality Parameter Reduction	5	5	5
Call pickup members per system	1,600	1,600	2,400
Call pickup members per group	50	50	50
Call Vectoring/Call Prompting			
Vectors per System	None	256	256
Vector Directory Numbers	None	500	500
Steps per Vector	None	15	15
Priority Levels	None	4	4
Multiple Split Queuing Splits/Call	None	3	3
Call Classifier Circuit Packs	None	10	10
Ports per Call Classifier Circuit Packs	None	8	8
Carriers:			
Control (PPN Cabinet-Without Duplication)	1	1	1
Control (PPN Cabinet-With Duplication)	2	2	2
Port (PPN Cabinet-Without Duplication)	4*	4*	4*
Port (PPN Cabinet-With Duplication)	3†	3†	3†
Expansion Control (EPN Cabinet)	1	1	1
Port (EPN Cabinet)	4‡	4‡	4‡

* Limited by the switch capacity. The external CMS limits may differ.

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
CMS Reports			
Basic CMS Reports			
Simultaneous BCMS Sessions	3	5	5
Measured agents	30	200	200
Measured agents per Split	30	100	100
Measured splits	30	99	99
Measured trunk groups	30	32	32
Measured trunk group Members	400	400	400
Reporting periods (30 or 60 minutes)	25	25	25
Daily summary reports	7	7	7
External CMS *			
Measured ACD agents per system	400	400	400
Measured splits	32	99	99
Measured trunk groups	99	99	99
Agents Simultaneously Entering Call Work Codes	None	40	40
Communication Interface Links			
Multi-Carrier Cabinet	8	8	8
Single-Carrier Cabinet	4	4	4
Conference Parties	6	6	6
Simultaneous 3-way conf. calls	483	483	483
Simultaneous 6-way conf. calls	240	240	240
Data Parameters			
Access Endpoints	None	400**	400**
Administered Connections	18	128‡‡	128‡‡
Alphanumeric Dialing	No	Yes	Yes
Max. entries	None	200	200
Digital Data Endpoints (Note 1)	800	800	800

* This quantity is three for G1 single-carrier cabinets.

† This quantity is two for G1 single-carrier systems with duplication.

‡ This quantity is three for G1 single-carrier cabinets.

§ A prefixed extension number can be six digits.

** Access Endpoints consume the same resource that trunks use. The sum of Access Endpoints and trunks cannot exceed 400 (see Trunks section of table). The number of Access Endpoints is also subject to the Miscellaneous Extension limitation (see Dial Plan).

‡‡ Prior to G1.2, these were known as Permanent Switched Connections.

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
Dial Plan (Name/Number Database)			
Extensions (Note 2)	2,500	2,900	3,500
Dial access codes	70	70	70
Number of digits	1-3	1-3	1-3
Trunk access codes	197	197	197
Number of digits	1-3	1-3	1-3
Names			
Number of names	3,064	3,064	4,215
Name size in characters	15	15	15
Integrated Directory Entries	1,600	1,600	2,400
Minimum extension size	1	1	1
Maximum extension size	5§	5§	5§
Multiple Listed Directory Numbers	50	50	50
DID numbers	8	8	8
Prefix extensions	Yes	Yes	Yes
Miscellaneous extensions	900	900	900
Phantom Extensions (Admin Without Hardware)	None	1,600	1,600
Phantom Users	150	150	150
Do Not Disturb (DND)			
DND requests per system	1,600	1,600	2,400
Simultaneous display requests	10	10	10
Facility Busy Indicators	2,400	2,400	3,600
Buttons per tracked resource	100	100	100
Hunt Groups or Splits			
Groups/Splits	99	99	99
With Hospitality Parameter Reduction	5	5	5
Group members per system	500	500	500
Group members per group/split	200	200	200
Measured Groups and/or Splits	99	99	99
Queue slots per group	200	200	200
Queue slots per system	1,000	1,000	1,000
Announcements per group	2	2	2
Agents Logged in per System	400	400	400
ACD supervisors per system	99	99	99

§ A prefixed extension number can be six digits.

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
Intercom Translation Table (ICOM)			
ICOM groups per system	32	32	32
Members per ICOM group	32	32	32
Members per system	1,024	1,024	1,024
Leave Word Calling (SPE Based)			
Messages stored	2,000	2,000	2,000
Messages per user	10	10	10
Individual message retrievers	60	60	60
System-wide message retrievers	10	10	10
Remote message waiting indicator			
Per extension	80	80	80
Per system	80	80	80
Modem Pool Groups			
Mode 2/analog	5	5	5
Members per group	32	32	32
Group member per system	160	160	160
Networking			
CAS Nodes	99	99	99
DCS Nodes	63	63	63
UDP Nodes	240	240	240
Paging*			
Loudspeaker zones	9	9	9
Code calling Identifiers	125	125	125
Personal CO Line group (PCOL)			
PCOL groups	40	40	40
PCOL members in a group	4	4	4

* These maximum parameters do not apply if PagePac Paging Systems are used.

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
Port Circuit Pack Slots			
PPN			
MCC - Without Duplication	89	89	89
- With Duplication	78	78	78
SCC - Without Duplication	64	64	64
- With Duplication	56	56	56
PPN w/ 1 EPN			
MED w/MED - Without Duplication	186	186	186
MED w/MED - With Duplication	172	172	172
SCC w/SCC - Without Duplication	133	133	133
SCC w/SCC - With Duplication	122	122	122
PPN w/ 2 EPN			
MED w/MED - Without Duplication	None	281	281
MED w/MED - With Duplication	None	262	262
SCC w/SCC - Without Duplication	None	200	200
SCC w/SCC - With Duplication	None	184	184
Power Failure Transfer Extensions			
Model 574-5 Panel	35	35	35
Model 808A Panel	35	35	35
Recorded Announcements			
Recorded announcements	64	128	128
Analog queue slots per system	150	150	150
Analog queue slots per announcement	150	150	150
Integrated queue slots per system	50	50	50
Calls connected per announcement	5	5	5
Integrated annc. circuit packs	1	1	1
Channels per integrated annc. circuit pack	16	16	16
Integrated annc. recording time (min:sec)			
32 KB Recording	4:16	4:16	4:16
16 KB Recording	None	8:32	8:32
Speech Synthesis Circuit Packs	6	6	6
Channels per speech circuit pack	4	4	4
Call Detail Recording (CDR)			
CDR output device	2	2	2
Tracked Trunks	400	400	400
Tracked Stations	None	100	100
Buffered records	200	350	350

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
System Administration			
Asynchronous links (RS232)	7	8	8
Simultaneous Mgr I or G3-MT sessions	3	5	5
Simultaneous administration commands	1	1	1
Simultaneous maintenance commands	1	1	1
Scheduled report entries	50	50	50
Administration history file entries	250	250	250
Terminating Extension Groups (TEG)			
TEGs	32	32	32
Users that may share a TEG	4	4	4
Time Slots			
Total slots	1,024	1,536	1,536
Slots available for call switching	966	1,449	1,449
Simultaneous circuit switched calls	482*	723	723
Tone Classifiers			
Tone Detector circuit packs	20	10	20
General Purpose Tone Detectors	40	20	40
Touch Tone Receivers	80	40	80
Call Classifier circuit packs	None	10	10
Call Progress Tone/Touch Tone Receivers	None	80	80
R2-MFC Detector circuit packs	None	None	5
R2-MFC Receivers	None	None	40
TTR queue size	4	4	4

* Based on 241 simultaneous conversations using time slots on the PPN bus and 241 simultaneous conversations using time slots on the EPN bus.

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
Traffic Handling Capability (CCS)†	17,352	26,028	26,028
Basic System (BHCC)‡	7,200	10,000	10,000
ISDN System - 100 Percent PRI(BHCC)‡	5,000	6,000	6000
ISDN System - 100 Percent PRI/BRI(BHCC)‡	None	4,000	4,000
ACD System (BHCC)‡	5,500	7,000	7,000
ICM/IG (BHCC)‡	2,800	None	?
Call Visor			
ASAI-ICM (BHCC)‡	None	3,000	3,000
ASAI-OA (BHCC)‡	None	7,000	7,000
Trunks (see Note 3)			
Trunks in system	400	400	400
With Hospitality Parameter Reduction	50	50	50
Trunk members in a trunk group	99	99	99
Trunk groups in the system	99	99	99
Queue slots for trunks	198	198	198
Ringback Queue Slots	120	120	120
DS-1 circuit packs	30	30	30
PRI Interfaces (D-channels)			
MCC	8	8	8
SCC	4	4	4
PRI Interfaces (B-channels) (see Note 4)			
MCC	184	400	400
SCC	92	400	400
PRI Temporary Signaling Connections (TSCs)	None	656	656
Call-Associated	None	400	400
Non-Call-Associated	None	256	256

† Based on 241 simultaneous conversations using time slots on the PPN bus, 241 simultaneous conversations using time slots on the EPN 1 bus, and 241 simultaneous conversations using time slots on the EPN 2 bus.

‡ A call completion requires the terminals at both end points of a connection to be off hook with an established voice path between terminals. Nominal calculation based on a call distribution of 33% incoming, 33% outgoing, and 34% intercom calls—traffic capacities for individual configurations will vary.

Continued on next page.

Table 3-1. Maximum System Parameters for Hardware and Software Items (*continued*)

ITEM	G1	G1.2	GD91
Voice Terminals			
DCP Terminals			
(Combination of Digital, Hybrid, and Analog Terminals including 515 or 510D terminals, external alerts, and announcement machines.)			
MCC	1,600	1,600	2,400
SCC	1,200	1,600	2,400
Digital Terminals (Note 5)			
MCC	1,472	1,600	2,400
SCC	1,048	1,600	2,400
DCP Terminals With Display (Note 6)			
MCC	500	1,600	2,400
SCC	500	1,600	2,400
Phantom Users (Note 7)			
MCC	150	150	150
SCC	150	150	150
BRI Stations	None	1,000	1,000
BRI Display Stations	None	1,000	1,000

Notes:

1. Digital data endpoints terminate on a port of the switch. Up to 800 ports can terminate digital data endpoints and the remaining system ports are available for terminating digital voice terminals, trunks, etc.
2. For G1.2, the extensions are shared among the following applications: voice station, attendant, data endpoint, hunt group, recorded announcement, TEG, VDN, extension for failed automatic wakeup messages, attendant common shared extensions, code calling ID, and phantom extensions. (For example, it is possible to have 1600 stations plus 1300 phantom extensions, i.e., administration without hardware). In addition, there is another internal dial plan limit: PCOL groups, common shared extensions, access endpoints, code calling IDs, LDNs, hunt groups, announcements, and TEGs are limited to 860.
3. When more than 23 B-Channels for every one D-Channel are provisioned, Non-Facility Associated Signaling (NFAS) must be used.
4. Trunks and Access Endpoints (see Data Parameters entries for more on Access Endpoints) consume the same resource. The sum of trunks and Access Endpoints cannot exceed 400.
5. A fully equipped PPN cabinet and EPN cabinet without duplication (MCC system) has 184 port circuit pack slots available (after tone detectors and

expansion interfaces have been added in). A system using all these slots for digital line circuit packs has a maximum physical digital line capacity of 1472 lines (184 slots x 8 ports per slot). Replacing one of the digital line circuit packs with a trunk circuit pack or a circuit pack other than a digital line circuit pack reduces the digital line capacity by 8 lines. Therefore, trunking and analog and hybrid terminal requirements must be considered in determining the actual maximum for each system. determining the actual maximum for each system. When the second EPN is added in a G1.2 system, 1600 digital terminals can be configured in the system.

6. A button module is a portion of memory required to store button translations. Some digital voice terminals require no button modules while other terminals need one or more modules. Digital voice terminal button module and display requirements are given in the Station Allocation Characteristics table. These button module and display requirements limit the number of digital voice terminals with more than 10 feature buttons and/or displays that can be connected to a system. In G1.2, memory units are allocated by the system to each terminal based on the number of buttons actually translated. The button module scheme is no longer used in G1.2. There are 547,000 memory units available in G1.2 for terminals.

7. The system automatically initiates phantom user calls without any dialing from a voice terminal. An automatic wakeup call is an example of a phantom user call.

References

4

The following is an abbreviated listing of Generic 1 documents. Included is a brief description of each document in the list. User instructions are also available for all terminals used with the system.

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, and so on.

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 Hospitality Services **555-200-026**

Provides an overview of DEFINITY 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.

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 — 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.

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.

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 — Implementation 555-204-654

Provides the procedures and associated forms for collecting system and terminal software information for G1.1 systems. This information is used to initialize the system using the DEFINITY Manager I Terminal (G1) or G3 Management Terminal.

DEFINITY Communications System Generic 1 — Implementation (Generic 1.2) 555-204-655

Provides the procedures and associated forms for collecting system and terminal software information for G1.2 systems. Also describes various administration commands and error messages. This information is used to initialize the system using the System Access 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, and so on) required for efficient operation and use of the ACD 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.

DEFINITY® Communications System Generic 1 — System Management **555-230-500**

Describes Manager I terminal types, function keys, and other Manager I operations. Also describes various administrative tasks such as logon/logoff, changing of password, remote administration, and so on.

DEFINITY® Communications System Generic 1 — System Reports **555-230-510**

Explains switch-based measurement, traffic, performance, and summary reports. Descriptions include the overall purpose and uses for each report, complete definitions for each field, correlations with other reports, and possible actions that can be taken to further diagnose situations and remedy unsatisfactory conditions.

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.

Abbreviations

A

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

AWOH
Administration Without Hardware

AWT
Average Work Time

B

BCC
Bearer Capability Class

BCMS
Basic Call Management System

BCT
Business Communications Terminal

BHCC
Busy Hour Calls Completions

BLF
Busy Lamp Field

BN
Billing Number

BOS
Bit Oriented Signaling

BTU
British Thermal Unit

C

CACR
Cancellation of Authorization Code Request

CAMA
Centralized Automatic Message Accounting

CAS
Centralized Attendant Service

CBC
Call-by-Call

CCITT
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

CDR
Call Detail Recording

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

CPN
Calling Party Number

CPTR
Call Progress Tone Receiver

CRC
Cyclical Redundancy Checking

CSA
Canadian Safety Association

CSM
Centralized System Management

CSSO
Customer Services Support Organization

D

DC
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

DNIS
Dialed Number Identification Service

DOD
Direct Outward Dialing

DOSS
Delivery Operations Support System

DSU
Data Service Unit

DS1
Data Services Level 1

DTDM
Digital Terminal Data Module

DTE
Data Terminal Equipment

DTGS
Direct Trunk Group Select

DTMF
Dual Tone Multifrequency

DXS
Direct Extension Selection

E

EBCDIC
Extended Binary Coded Decimal Interexchange 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

E&M
Ear and Mouth (Receive and Transmit)

F

FAC
Feature Access Code

FAS
Facility Associated Signaling

FCC
Federal Communications Commission

FIC
Facility Interface Codes

FNPA
Foreign Numbering Plan Area Code

FRL
Facilities Restriction Level

FX
Foreign Exchange

G

GPTR
General Purpose Tone Receiver

GRS
Generalized Route Selection

H

HNPA
Home Numbering Plan Area Code

I

IAS
Inter-PBX Attendant Service

ICC
Inter Carrier Cable

ICI
Incoming Call Identifier

ICM
Inbound Call Management

IDDD
International Direct Distance Dialing

IE
Information Element

INADS
Initialization and Administration System

INS
ISDN Network Service

INWATS
Inward Wide Area Telephone Service

ISDN
Integrated Services Digital Network

ISN
Information Systems Network

IXC
Inter-Exchange carrier Code

K

KBPS
Kilo-Bits Per Second

L

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

M

MA-UUI
Message Associated User-to-User Signaling

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

MPDM
Modular Processor Data Module

MS
Message Server

MSA
Message Service Adjunct

MTDM
Modular Trunk Data Module

MTP
Maintenance Tape Processor

MWL
Message Waiting Lamp

M-Bus
Memory Bus

N

NANP
N. American Numbering Plan

NAU
Network Access Unit

NCOSS
Network Control Operations Support Center

NEC
National Engineering Center

NFAS
Non-Facility Associated Signaling

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

O

OCM
Outbound Call Management

OPS
Off-Premises Station

OQT
Oldest Queued Time

OSHA
Occupational Safety and Health Act

P

PBX
Private Branch Exchange

PC
Personal Computer

PCM
Pulse Code Modulated

PCOL
Personal Central Office Line

PCOLG
Personal Central Office Line Group

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

R

RAM
Random Access Memory

RCL
Restricted Call List

RHNPA
Remote Home Numbering Plan Area Code

RLT
Release Link Trunk

RNX
Private Network Office Code (Routing Number Exchange)

ROM
Random Access Memory

RPN
Routing Plan Number

RSC
Regional Support Center

S

SAKI
Sanity and Control Interface

SAT
System Access Terminal

SCC
Single Carrier Cabinet

SCI
Switch Communications Interface

SCO
System Control Office

SDDN
Software Defined Data Network

SDI
Switched Digital International

SDN
Software Defined Network

SID
Station Identification Number

SIT
Special Information Tones

SPE
Switch Processing Element

SSI
Standard Serial Interface

STARLAN
Star-based Local Area Network

T

TAAS
Trunk Answer From Any Station

TAC
Trunk Access Code

TCM
Traveling Class Mark

TDM
Time Division Multiplex

TEG
Terminating Extension Groups

TOD
Time of Day

TOP
Task Oriented Protocol

TTTN
Tandem Tie Trunk Network

TTY
Teletypewriter

U

UAP
Usage Allocation Plan

UCD
Uniform Call Distribution

UCL
Unrestricted Call List

UDP
Uniform Dial Plan

UPS
Uninterruptible Power Supply

V

VDN
Vector Directory Number

W

WATS
Wide Area Telecommunications Service

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