Lucent Technologies
Bell Labs Innovations

DEFINITY®
Business Communications System and GuestWorks®

Issue 6
Overview

555-231-208
Comcode 108596990
Issue 1
April 2000

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- Electromagnetic Compatibility (89/336/EEC)
- Low Voltage (73/23/EEC)
- Telecommunications Terminal Equipment (TTE) i-CTR3 BRI and i-CTR4 PRI

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Acknowledgment
This document was prepared jointly by the Customer Training and Information Products Organization and the Information Development Organization for Global Learning Solutions
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About This Document

The purpose of this overview

This document provides general information about the components and capabilities of the DEFINITY® Business Communications System (BCS) and GuestWorks® Issue 6 offers. You will learn how these systems provide practical and creative solutions to your business's needs.

The intended audiences

This document is written for those who have purchased or are considering the purchase of a DEFINITY BCS or GuestWorks, and for Lucent Technologies representatives and distributors who need high-level information about the system and how it can be used.

The content of this overview

This document discusses all system capabilities. It defines standard and practical solutions, but also suggests unusual or creative ones.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

This overview includes the following information:

- Chapter 1, "New Features for Issue 6," gives a list of the new enhancements for this release.
- Chapter 2, "Introduction," outlines basic advantages, capabilities, hardware and software components, and system configurations.
- Chapter 3, "Industry Applications," discusses how the system meets the communications requirements of several example industries.

- Chapter 4, "Desktop Solutions," discusses features that are available at your desktop computer or telephone.
- Chapter 5, "Mobility Solutions," discusses products and features that allow you to keep in touch with colleagues and clients while moving about freely inside and outside the workplace.
- Chapter 6, "Computer-Telephone Integration Solutions," discusses features that merge computer and telephone functions.
- Chapter 7, "Hospitality Solutions," discusses GuestWorks products and features useful for the hospitality and lodging industry.
- Chapter 8, "Data Management Solutions," discusses features that help you manage telecommunications information.
- Chapter 9, "Networking Solutions," discusses features that help you network your equipment and solutions.
- Chapter 10, "Voice Messaging Solutions," discusses features that use voice messaging to help you handle incoming and outgoing calls efficiently.
- Chapter 11, "Video Solutions," discusses features that allow you to send and receive synchronized voice and image information.
- Chapter 12, "Hunt Group Solutions," discusses features that help you set up and manage basic call management call groups.
- Chapter 13, "Telecommuting Solutions," discusses features that help you and your associates work effectively off-site.
- Chapter 14, "System Management Solutions," discusses the ways in which you can manage the switch and related systems.
- Appendix A, "Features," summarizes the features available with the system.
- Appendix B, "Features Not Supported," lists the features, adjuncts, and hardware not supported by DEFINITY BCS and GuestWorks.
- Appendix C, "Related Documents," lists and describes related documents.

A glossary with abbreviations and an index are also provided at the end of the document along with a feedback form.

How to use this document

Read Chapter 1, "New Features for Issue 6," to learn about the new features and hardware for DEFINITY BCS and GuestWorks Issue 6. Review Chapter 2, "Introduction," to get a basic understanding of the system. Next, review Chapter 3, "Industry Applications," since it discusses specific applications that may help you apply the system creatively. Although the applications may not describe your business or situation exactly, scanning the examples will help you to generate ideas that do apply to your business.

Read the more in-depth discussions of general applications in Chapters 4 through 14 selectively, focusing on the solutions that suit your circumstances.

Appendix A lists all of the system features and includes a short description of each feature. These feature descriptions help you understand the features as well as the scope of the system capabilities. Appendix B lists the DEFINITY Enterprise Communications Server (ECS) features that are not supported on the DEFINITY BCS or GuestWorks offers. The remainder of the book presents important reference material.

Conventions used in this document

The following conventions are used:

- The term *system* or *switch* is used to represent the DEFINITY BCS and GuestWorks products. The term *switch* is also used to represent other telecommunications switching products.
- DEFINITY BCS is used for general telecommunications applications, and GuestWorks is used for hospitality applications.
- Issue 6 of DEFINITY BCS and GuestWorks is part of the DEFINITY ECS Release 8 product.

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Where to find related documents

See Appendix C, "Related Documents," for a detailed list of documents related to the DEFINITY BCS and GuestWorks offers. Use these documents to help administer, maintain, and operate the system.

With each system that is shipped from the factory, you will receive a compact disc (CD-ROM) that contains most of the supporting documents listed in Appendix C. These documents can be viewed and printed from a personal computer. The order number for the CD-ROM is 555-233-813.

CAUTION:

Not all features in these reference documents will be available with DEFINITY BCS or GuestWorks. See Appendix A for a list of the supported features, and Appendix B for a list of the features not supported.

How to order related documents

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If the reader comment form is missing, fax your comments to +1-303-538-1741, and mention this document's name and number, *DEFINITY® BCS and GuestWorks® Issue 6 Overview* (555-231-208, Issue 1).

New Features for Issue 6

1

The following features and hardware are new for this release:

- 13-digit Authorization Codes, increased from a maximum of seven digits (see Page A-11)
- 6200 analog telephone native support (see Page 4-3)
- 64 bridged call appearances, increased from a maximum of 26 bridged appearances (see Page A-13)
- 6400 tip/ring module (see Page 4-4)
- Administrable loss plan (see Page A-6)
- Attendant Vectoring (see Page A-44)
- Auto Exclusion (see Page A-30)
- Automatic Selection of Direct Inward Dialing Numbers (DID) to Guest Rooms (see Page 7-5)
- Bellcore calling name ID and Caller ID telephone type (see Page A-12)
- CallMaster V native support

  The CallMaster V telephone can now be administered with a native telephone type.
- Centralized voice mail via interswitch mode codes (see Page A-47)
- Circular hunt groups (see Page A-33)
- Crisis Alert to Pager (see Page 7-5)
- E&M Continuous and Pulsed Signaling (see Page A-50)
- Fast analog modem support

  Analog ports on the switch have been tested to be compatible with the V.90 standard for 56 Kbps modems.
- Group call pickup (see Page A-15)
- ISDN Restricted Presentation (see Page A-52)
- Japanese national private networking support (see Page A-48)

- Long hold recall warning (see Page A-26)
- Multiple Public Network Calling/Connected Numbers/System (see Page A-52)
- Optical drive to replace tape drive for “r” systems

  A new read/write optical disk drive has replaced the cartridge tape system for all new “r” systems.
- Outgoing call no-answer by call type (see Page A-29)
- Pass Advice of Charge to World Class BRI Endpoints (see Page A-29)
- Reset shift call (see Page A-32)
- Special Dial Tone (see Page A-33)
- Station self-display (see Page A-34)
- Suite Check-in and Station Hunt Before Coverage (see Page 7-5)
- Time supervision and forced release (see Page A-35)
- TN793B/TN2793B Analog Line 2-wire, 24-port circuit pack with caller ID

  These circuit packs provide the capability to view, on an analog Caller ID display, the telephone number and name of the calling party, and time and date of the call. This circuit pack is used primarily with stand-alone Caller ID units and Caller ID hospitality telephones provided by Teledex and Telematrix.
- TN797 combination board (analog line/CO trunk)

  This board provides a new combination eight-port analog trunk and line circuit pack (TN797). With this board, you can administer any of the eight ports of this analog circuit pack as a:

  — Central Office trunk, either loop start or ground start

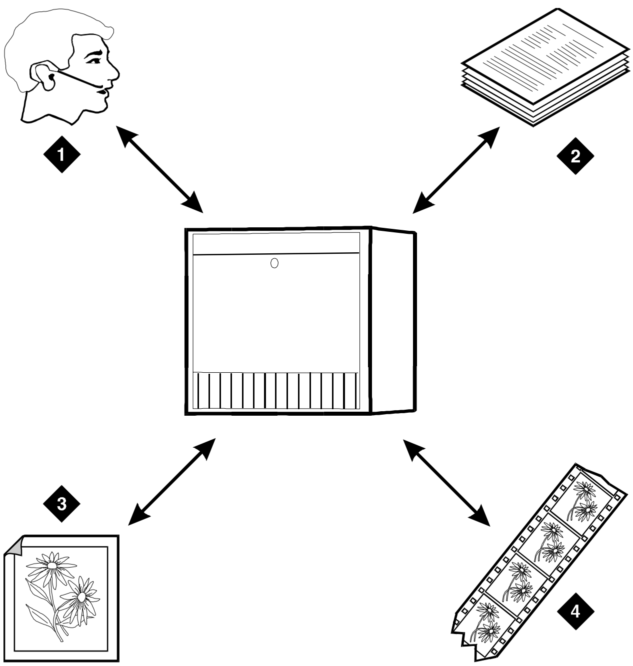
  — CAMA E911 trunk

  — Direct Inward Dialing trunk, either wink start or immediate start

  — An analog line, on or off-premises, with or without LED message waiting indication (MWI). However, this circuit pack does not support neon message waiting lamps.
- Transfer Abort (see Page 4-10)
- Transfer Recall (see Page A-36)
- X-station mobility (see Page 5-5).

Introduction 2

DEFINITY BCS and GuestWorks organizes and routes voice, data, image, and video transmissions (see Figure 2-1). The transmitted information is usually digitized (distilled into representative sequences) as it is switched (organized and routed), but the system can also receive and transmit analog information.



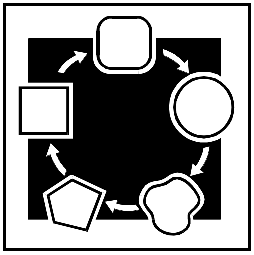
1) Voice
2) Data
3) Image/Facsimile
4) Video

Figure 2-1. DEFINITY BCS and GuestWorks

The system's advantages

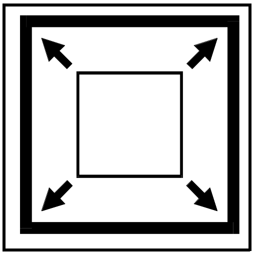
This is an affordable system that handles basic telephony traffic as efficiently as any system available. The system can accommodate most related equipment, and its modular design allows for system updates. These capabilities mean that the system offers your business an exciting array of practical, solutions-oriented features.

The system adapts to global protocols



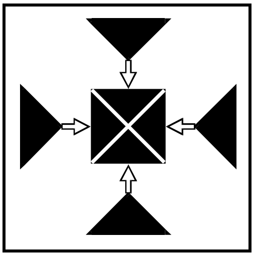
The system's open architecture and modular design make it compatible with a wide variety of hardware and software — both Lucent Technologies tools and tools from other vendors. These may include personal computers and shared servers, terminals, data access equipment, telephones, fax machines, and property management equipment. Multilingual options are available for messaging and telephones. The system was designed to accommodate both existing and future global communications protocols. It is adaptable to varying standards world-wide, providing efficient digital switching even when connected to conventional networks.

The system expands to match business growth



Modular port circuits, carriers (circuit shelves), and cabinets can be added to accommodate growth up to 25000 stations. Each system can also network to other systems (Lucent Technologies systems or other types) to service many simultaneous voice, data, image, and video transmissions.

The system integrates related tools

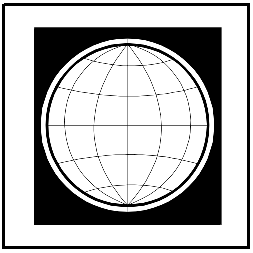


The system not only communicates with most networks and equipment throughout the world, but also unifies them by translating protocols as necessary. The system is designed to accommodate multimedia and network integration tools, in addition to features that integrate computer and telephone. The system's integration capabilities and its association with leading-edge tools enhance the value of your related telecommunications investments.

The system is reliable and recoverable

The system is reliable by design. If adjuncts connected to the system should fail, the system keeps working until those adjuncts are restored. If the disturbance is great enough that the system is disabled along with those other adjuncts, the modular design provides many options for getting your communications back into service quickly.

The system is a global platform



The system is compatible with video teleconferencing systems and desktop network management applications.

It accommodates existing and emerging worldwide standards and protocols.

It offers multilingual options for some applications.

Messaging services, world-class call features, multilanguage displays, and multilanguage voice messaging prompts augment your communications with associates and enhance business transactions around the globe.

NOTE:
Some applications and products are unavailable in some countries. Please check with your local distributor for further information about which features and applications are available to you.

The following section provides detailed information on why the system is the best choice for serving your global communications needs.

How the system communicates worldwide

The system adapts to the different telecommunication standards around the world, allowing you to use the same communications system in various countries.

The following capabilities are of particular interest to organizations that do business throughout the world:

- The system provides flexible language displays that allow you to administer the language in which messages are displayed on telephones.

- Terminal Translation Initialization (TTI) allows you to provide telephone service on demand as needs change.
- Music-on-Hold for Analog Ports allows the music-on-hold device to be connected to analog line ports.
- World Class Routing/Automatic Route Selection (ARS) allows flexible call routing for any type of national or international dialing plan. It consists of the following capabilities:
  - Flexible dialing
  - 18-digit routing
  - Automatic Route Selection (with International Direct Distance dialing calls and interexchange carrier access)
  - Automatic Alternate Routing for private networking
  - Digit conversion.
- QSIG Global Networking provides compliance with the European Computer Manufacturers Association Integrated Services Digital Network-Primary Rate Interface specifications. This interface supports voice and data basic call setup.
- Enhanced ISDN capabilities include the following:
  - Support for either Basic Rate Interface A-law or Basic Rate Interface Mu-law companding
  - Support of Integrated Services Digital Network (ISDN) slot maps to provide ISDN capabilities in countries that require them
  - QSIG Basic Global Networking.
- Digital signaling support is available for countries that require it.
- Generalized Multifrequency-Compelled Signaling allows 18 digits on Multifrequency-Compelled facilities for incoming calls. Multifrequency Russian is also supported.
- Multifrequency-Espanol interregister signaling needed in Spain for its E1 digital connectivity is supported. The protocols supported are the Public Network 2/5 and the IBERCOM 2/6.
- Cut-through on central office trunks provides connection to the central office immediately after the trunk access code is dialed and checks the digits for toll restriction.
- Added Restriction Checks allow you to block the connection of public network trunks to other public network trunks to allow compliance with local standards and regulations.
- Administrable Call Progress Tones allow you to select the dial tone, busy tone, ringback, reorder, and other tones that conform to local standards.
- Administrable Ring Cadence allows you to select the ring cadence for analog telephones to conform to local standards.

- Administrable Transmission allows you to select the transmission requirements that conform to local requirements.
- Administrable Timers support varied international trunk interface requirements, allowing you to change the timing according to local standards.
- Administrable Repetitive Call Waiting allows administration of the repetitive call waiting tone interval from 4 to 40 seconds in 1-second intervals.
- Attendant Serial Calling enables the attendant to transfer trunk calls returned to the attendant position once the called party has hung up, allowing the attendant to transfer the call to another party.
- Enhanced Attendant Queue, Display, and Misoperation allows attendants to see the exact number of calls and types in queue, and to prioritize calls via their different call types for countries that require it. In addition, in countries that require this, an attendant placing a call on hold and going on hook is considered a misoperation, and the attendant is alerted.
- Disconnect Supervision management avoids having system resources used indefinitely when far-end central office disconnect supervision is not provided. Resources used on the call are removed and made available for servicing new calls.
- When an internal user is the last person remaining off-hook on a call, that person's telephone will receive busy tone for 30 seconds or until the user hangs up the phone. This feature is called Busy Tone Forward Disconnect and can be enabled or disabled on a system-wide basis.
- International Toll/Code Restriction allows you to restrict calls using any international numbering plan.
- Call Detail Recording enhancements for periodic pulse metering provides periodic pulse metering pulse counts in the Call Detail Recording output record. The pulses transmitted over trunk lines from the serving central office are used to determine call charges.
- T1/E1 access and conversion allows simultaneous connection to both T1 (1.544 Mbps) and E1 (2.048 Mbps) facilities.

Most of these capabilities are described in greater detail throughout this document. See Appendix A, ''Features,'' for a listing of features available on the system. Appendix B lists the DEFINITY ECS features *not* supported by DEFINITY BCS and GuestWorks. For a complete description of the features used with the system, see the *DEFINITY® ECS Administrator's Guide*.

Software

All DEFINITY systems (and related Lucent Technologies systems) use similar software. For example, DEFINITY BCS and GuestWorks Issue 6 uses the same base software as DEFINITY ECS R8. To provide this commonality while still accommodating wide variations in configurations and options, the system dynamically allocates internal memory storage. Memory is sized when the system is initialized, selecting the proper software parameters based on the hardware configuration.

Hardware

Though the primary components are the same, your system can vary widely in size and appearance, depending on your capacity requirements. The system may be as small as a single, wall-mounted cabinet, or it may be as large as several tall cabinets linked together in the same room or even hundreds of kilometers apart. Regardless of configuration, however, the system's footprint is relatively small.

The system's main hardware components are port networks. Up to three port networks can be connected directly to each other. When there are more than three port networks, the connections are made through a Center-Stage switch.

Processor Port Network

Every system has one Processor Port Network; it is often the only component in small systems. The Processor Port Network houses the Switch Processing Element (SPE).

The SPE contains the central processing unit, which supervises system operation. It also contains a mass storage system for loading system software and saving system translations.

Because your application requirements may vary widely, the system has three types of SPEs available with proven capacities up to 100000 busy-hour calls. The performance you realize will depend on the call processing, administrative, and maintenance activities in which your system is engaged. See the *DEFINITY® ECS System Description* for more details.

Expansion Port Network

Expansion Port Networks are used when the system grows beyond the capacity of a single port network or must serve geographically-dispersed offices. They provide additional ports as needed. Depending on the model, a system can have up to 43 Expansion Port Networks.

Center-Stage Switch

The Center-Stage Switch (available only on an “r” system) is a connection hub that provides port network communication. It is an essential component of a system configuration if the system is composed of more than three port networks. Often it is incorporated in smaller configurations to allow for growth. The Center-Stage Switch consists of one to three switch nodes. Switch nodes are composed of one switch node carrier for both standard and high reliability. Each carrier can reside in the Processor Port Network cabinet or in an Expansion Port Network cabinet. One switch node can accommodate up to 15 Expansion Port Networks.

Fiber Link Administration

Port cabinets are connected via direct fiber links or through fiber links to a Center-Stage Switch to provide the connections required for voice and data information transfer. The Center-Stage Switch is composed of switch node carriers that are interconnected by fiber links. It provides both circuit-switched and packet-switched connections. Fiber Link Administration creates the translation data defining these links by identifying the end point pairs for each link. End points can be an expansion interface or a switch-node-interface circuit pack.

Carriers and Cabinets

Carriers are enclosed shelves composed of vertical slots that hold circuit packs. Circuit packs make up the logic, memory, and switching circuitry for the system. Port circuit packs connect to telephones, computers, and communications lines. The carriers are designed to accept any type of port circuit pack in each circuit pack position.

Each cabinet contains at least one carrier. The circuit packs fit into connectors attached to the rear of the slots. Every connector is connected to signal buses and power supplies in the cabinet. The cabinets also house equipment that supplies power backup, ringing signal voltage, and mass storage for software translations.

There are three types of cabinets:

- Compact Modular Cabinet (CMC). This cabinet functions like the compact single-carrier cabinet, but up to three of the cabinets can be connected together. The CMC can be mounted on the wall and is intended for smaller configurations.
- Single-Carrier Cabinet. These cabinets are modular, can be connected to Expansion Port Networks, and can be stacked up to four high. They are often used by small businesses that are growing or that expect to grow.
- Multicarrier Cabinet. A tall cabinet that contains up to five carriers and can be connected to Expansion Port Networks. Multicarrier cabinets are used by businesses that require larger configurations.

Compact Modular Cabinets

Figure 2-2 shows a Compact Modular Cabinet.

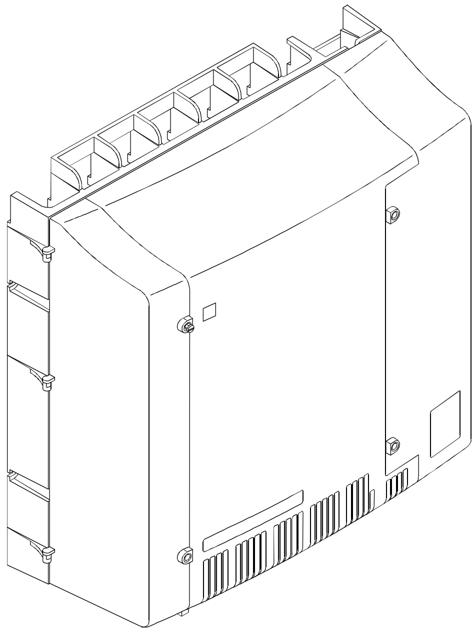


Figure 2-2. Compact Modular Cabinet (CMC)

The Compact Modular Cabinet has the following characteristics:

- Up to three cabinets can be connected together.
- It allows small organizations to expand while keeping the initial investment moderate.
- It can be mounted on a wall.
- The first two universal port slots in the first cabinet (cabinet 1) are dedicated to the processor complex; therefore, there are eight slots available in the first cabinet for general use.
- The other two cabinets have 10 universal port slots per cabinet.

The CMC is used as a Processor Port Network only. It does not support duplication, and it requires AC power.

Single-Carrier Cabinets

Figure 2-3 shows a typical Single-Carrier Cabinet.

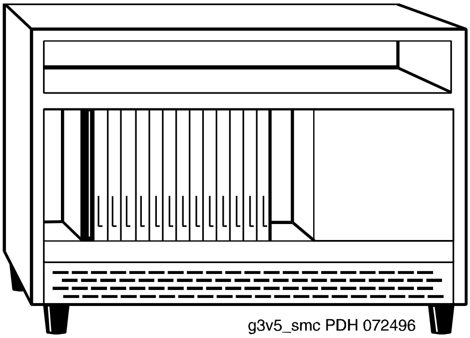


Figure 2-3. Typical Single-Carrier Cabinet (SCC)

A maximum of four single-carrier cabinets can be stacked on top of each other to form a single Processor Port Network or Expansion Port Network. There are four types of single-carrier cabinets:

- Control cabinet (located in the Processor Port Network only), which contains ports and a control complex (for call processing). Each control cabinet contains 16 universal slots.
- Port cabinet (located in the Processor Port Network and in Expansion Port Networks), which contains ports. Each port cabinet contains 18 universal slots.
- Duplicated control cabinet (PPN only), contains duplicate SPE circuit packs to perform call processing, maintenance, and administration identical to the control carrier. The duplicated control carrier also contains 16 port circuit pack slots. Only G3si and G3r support duplication.
- Expansion control cabinet (optional and located only in an Expansion Port Network), which contains ports, a tone-clock, an interface to a Processor Port Network cabinet, and a maintenance interface. Each expansion control cabinet contains 17 universal slots.

Figure 2-4 shows a typical cabinet stack.

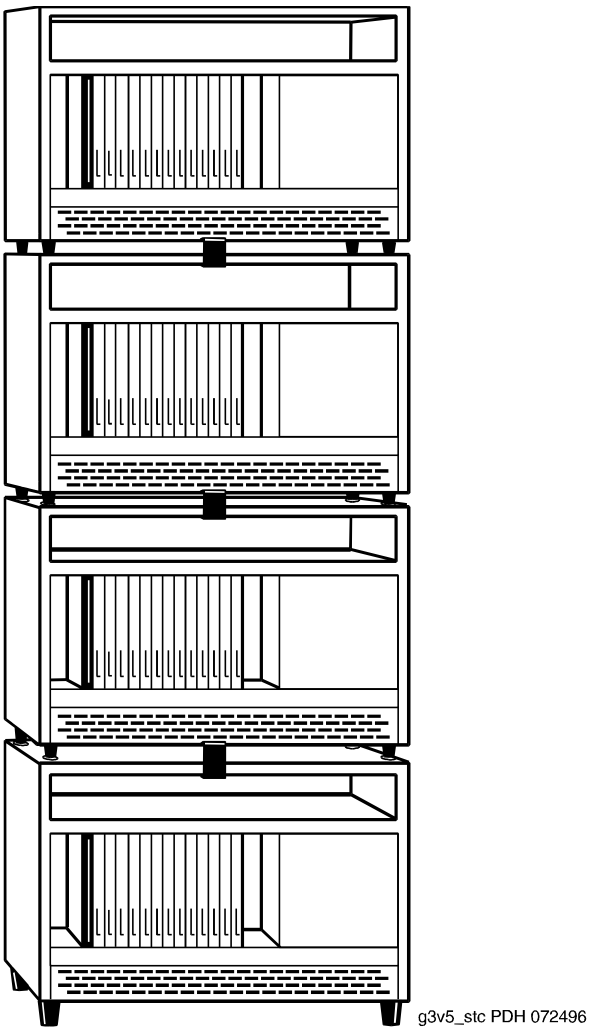


Figure 2-4. Typical Cabinet Stack (four cabinets maximum)

Multicarrier Cabinets

Figure 2-5 shows a typical Multicarrier Cabinet. The power arrangement shown at the bottom of the figure will be different depending upon the country where the cabinet is installed.

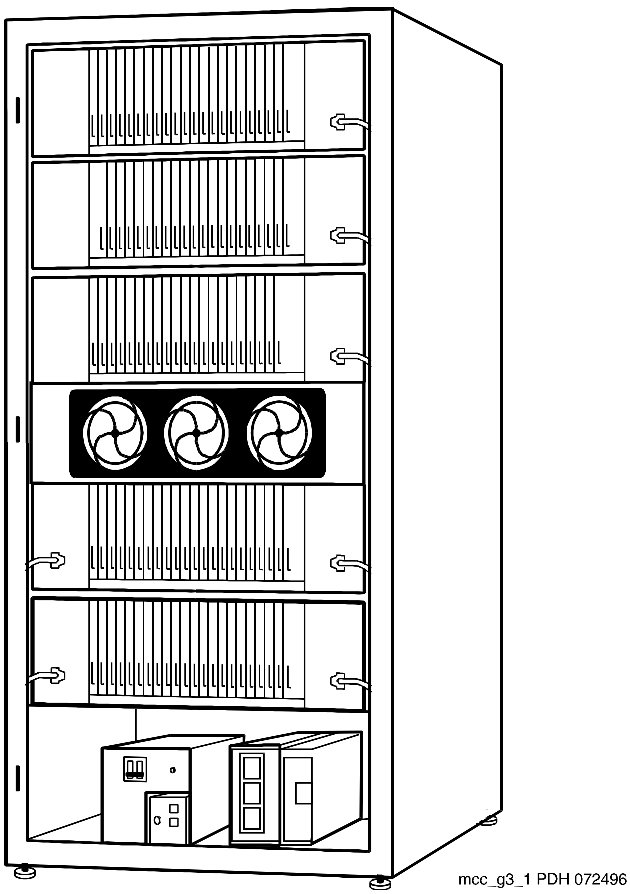


Figure 2-5. Typical Multicarrier Cabinet (MCC)

There are two types of multicarrier cabinets:

- The Processor Port Network cabinet, which contains the following:
  - — The processor that performs call processing
  - — Ports
  - — An interface to an Expansion Port Network cabinet (optional)
  - — A Center-Stage Switch (optional; for G3r only).

- The Expansion Port Network cabinet, which contains the following:
  - Additional ports
  - Interfaces to the Processor Port Network cabinet and other Expansion Port Network cabinets
  - Maintenance interface
  - Components of a Center Stage Switch (optional; for G3r only).

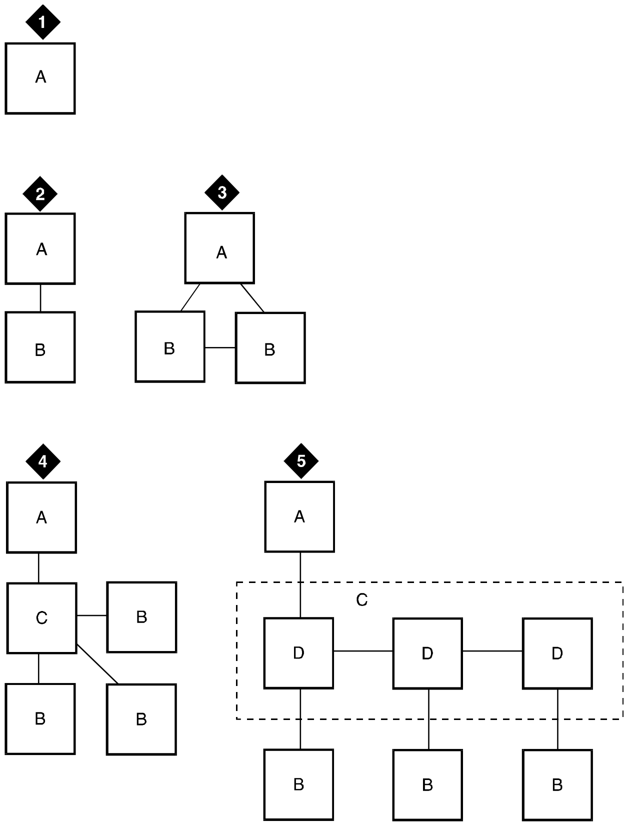
Control, duplicated control, expansion control, port, and switch node carriers can be installed in multicarrier Processor Port Network and Expansion Port Network cabinets. The slots for the carriers differ based on the type of system as follows:

- Control Carrier and Duplicated Control Carrier for G3si have nine universal slots.
- Control Carrier and Duplicated Control Carrier for G3r have no universal slots.
- Expansion Control Carrier has 19 universal slots.
- Port Carrier has 20 universal slots.

For more detailed hardware information, see the *DEFINITY® ECS System Description*.

Standard configurations

The system hardware can be configured in a variety of ways, depending on the number of end points the switch serves and the number of circuit packs required to connect the end points. Figure 2-6 shows the five main system configurations.



1)	Basic System	A)	Processor Port Network
2)	Directly Connected System	B)	Expansion Port Network
3)	Directly Connected System with Two EPNs	C)	Center-Stage Switch
4)	CSS-Connected System with up to 15 EPNs	D)	Switch Node
5)	CSS-Connected System with up to 43 EPNs		

Figure 2-6. Standard Configurations

The main configurations are as follows:

1. Basic system consisting of a Processor Port Network (PPN) only
2. Directly-connected system consisting of two Port Networks (PNs): one PPN and one Expansion Port Network (EPN) connected directly together
3. Directly-connected system consisting of three PNs (one PPN and two EPNs) connected directly together
4. Center-Stage Switch connected system consisting of up to 15 EPNs interconnected by one Switch Node (SN) to the PPN
5. Center-Stage Switch connected system consisting of up to 21 EPNs interconnected by two SNs to the PPN, and up to 43 EPNs interconnected by three SNs to the PPN.

Direct-Connect configurations

Direct-connect configurations have these distinguishing characteristics:

- Every port network is connected to every other port network via an expansion interface circuit pack and a fiber-optic cable.
- Each fiber is connected to a fiber transceiver that can transmit great distances.

For G3r systems, a port network can be hundreds of kilometers away from the central site. These remote port networks are connected to the other port networks via a Digital Signal Level 1 (DS1 — T1 or E1) link attached to a converter board, which in turn is connected to the expansion interface. The converter board converts the fiber-optic signals between DS1 protocol and the internal expansion interface protocol so the signal can travel over dedicated public or private lines.

Center-Stage Switch configurations

Center-Stage Switch configurations have these distinguishing characteristics:

- An expansion interface in every port network is connected to a switch-node interface in the Center-Stage Switch.
- DS1 Remote Expansion Port Networks require T1/E1 Converter pairs at the remote end and switch node T1/E1 Converter pairs at the switch node. In the pairs, the T1/E1 Converter board converts the fiber-optic signals between T1/E1 protocol and the internal expansion interface protocol so the signal can travel over dedicated public or private lines.
- Switch-node interfaces and fiber-optic cables are also required for communications between switch-node carriers. The number of switch-node interfaces required depends on the call traffic between port networks whose switch-node interfaces reside in different carriers.

Reliability and recoverability

Much of the system's reliability and recoverability can be attributed to the switch architecture and the power of the system software. The distributed processor architecture provides subsystem processors on each circuit pack. A standard maintenance routine is conducted automatically by the system, as are periodic backups of translations.

Based on the needs of your organization, two redundancy configurations are available:

- Standard reliability
- High reliability.

Standard reliability

The built-in duplication of many of the system's parts makes it inherently reliable. In addition to the dual bus, the system includes the following:

- One control carrier
- One tone-clock circuit pack per port network
- Port networks interconnected by single fiber cables (SCC and MCC hardware only).

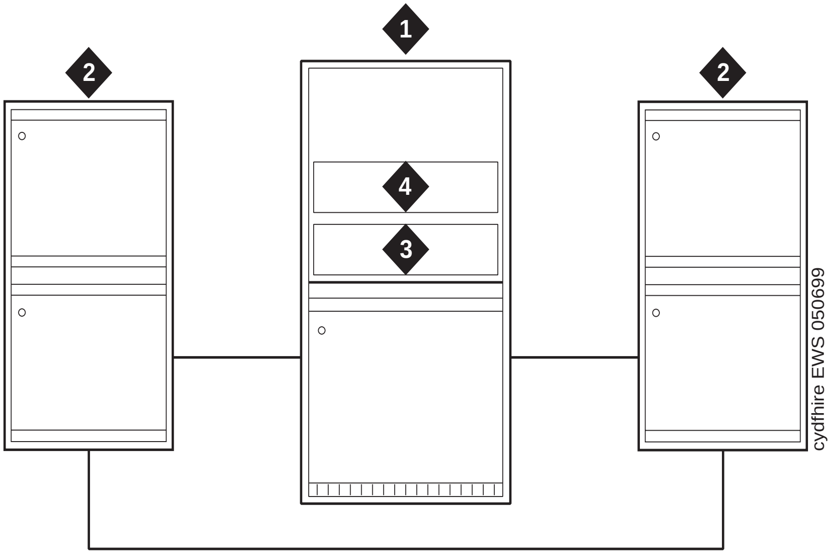
High reliability

NOTE:
The high-reliability option (also known as duplicated systems) may not be available in your region. Please contact your local account manager or authorized Lucent Technologies representative for further information about reliability options.

High-reliability systems include the following:

- Two control carriers (located in the Processor Port Network cabinet), which contain duplicate processor and tone-clock circuit packs (one is active and the other is in standby mode) (G3si and G3r only; not available on G3csi)
- One tone-clock circuit pack per Expansion Port Network
- Duplicate connections between the Center-Stage Switch and the Processor Port Network (G3r only)
- Expansion port networks connected by single fiber cables
- Duplicate switch-node clock circuit packs (one is active and the other is in standby) in each switch-node carrier (G3r only).

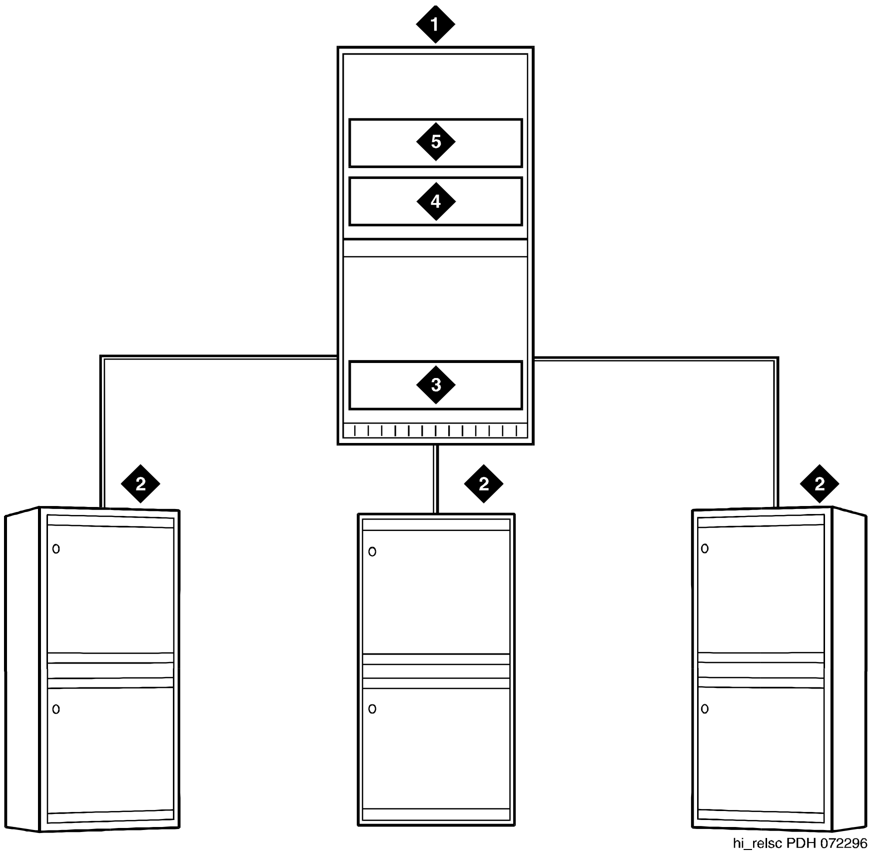
Figure 2-7 shows an example of a high-reliability, directly-connected system.



1) Processor Port Network
2) Expansion Port Network
3) Control Carrier
4) Duplicate Control Carrier

Figure 2-7. High-Reliability, Directly-Connected System

Figure 2-8 shows an example of a high reliability center stage system, where the Center-Stage Switch is connected to both the active and standby control carriers.



1) Processor Port Network
2) Expansion Port Network
3) Center-Stage Switch
4) Control Carrier
5) Duplicate Control Carrier

Figure 2-8. High-Reliability, Center Stage System

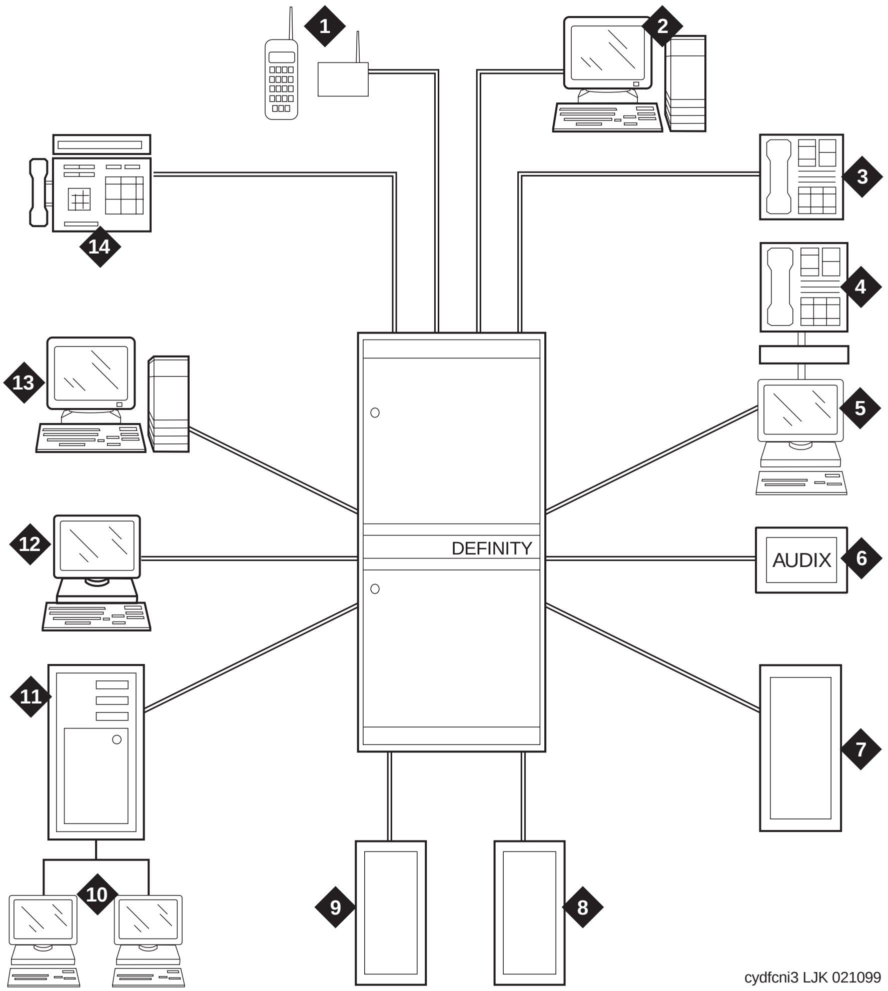
Connections to the system

The system can be connected to communications paths that transmit voice and data signals between the system and a Central Office and/or other systems. The system can also be connected to public and private networks. Other possible connections are as follows:

- Data Communications Equipment, such as a data module, which translates transmitted data to a form compatible with the communications channel
- Data Terminal Equipment, such as a workstation, which generates or receives data
- Other peripherals for administering and maintaining the system and auxiliary equipment for features such as Loudspeaker Paging and Music-On-Hold.

Figure 2-9 shows typical system connections.

NOTE:
Actual equipment may look different than the equipment shown.



1) Wireless System
2) Property Management System
3) Business Telephone
4) Telephone with Data Module
5) Data Terminal
6) Voice Messaging System
7) Outside Private Line Data Transmission Equipment
8) Digital Facilities
9) Analog Facilities
10) Data Terminals
11) Host Computer
12) Data Terminal
13) Management Terminal
14) Attendant Console

Figure 2-9. System Connections

Adjunct connections

In addition to station connections, the system includes many connections for adjunct (subordinate, related) equipment. The system provides a TCP/IP link adjunct interface. This interface supports a variety of adjuncts, including the INTUITY AUDIX Messaging System.

The system provides two Electronic Industries Association (EIA) RS-232 ports: one for a PC using the DEFINITY Site Administration management tool, and one for a spare connection. In addition, a tip/ring connector with a built-in modem is provided for remote administration on the ESCC and MCC cabinets. The CMC cabinet requires a separate modem.

The spare RS-232 port can support one of the following:

- Call Detail Recording Utilities
- Call Detail Recording printer
- System printer
- Property Management System (PMS)
- Basic Call Management System (BCMS) Terminal.

NOTE:
Normally the PMS and BCMS terminals will be connected into the switch via data modules and not connected through the spare RS-232 port.

The system uses an analog line circuit to support the following voice adjunct and interface functions:

- Loudspeaker paging
- Music-on-hold
- Queue status indications
- Recorded announcement
- External alarm inputs.

The system supports an auxiliary trunk interface that connects to equipment supporting the following features:

- Recorded announcement
- Music-on-hold
- Loudspeaker paging.

The system supports the following network interfaces:

- Electronic Tandem Network
- Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI).

Telephone connections

All signals between analog telephones and the system are in analog form over a pair of wires. Digital telephones (such as the 6400-series telephones) using the Digital Communications Protocol (DCP) employ digital transmission for integrated voice and data signals and control signals. Transmission is over a connection consisting of one or two pairs of wires. Each connection supports one signaling channel and two information (voice and data) channels.

Like the digital DCP telephones, ISDN telephones transmit voice, data, and control signals digitally. With the ISDN telephones however, the transmission employs the worldwide standard BRI protocol between the system and the telephone.

Network connections

Lucent Technologies is the first vendor to provide compatibility with the QSIG global networking protocol. This means you can connect the system with other switches throughout the world. QSIG Global Networking was developed to comply with the QSIG standards developed by the European Computer Manufacturer's Association and the International Standardization Organization. It supports the ISDN-PRI connection from system to system as long as both systems support the same protocol.

The system supports both E1 and DS1 facilities. As industry standards around the world, E1 and DS1 provide the latest alternative to analog trunking. T1/E1 access and conversion allows simultaneous connection to both T1 (1.544 Mbps) and E1 (2.048 Mbps).

The system's support of ISDN-PRI, ISDN-Basic Rate Interface, and available public network services means that you can achieve full end-to-end ISDN connectivity and take advantage of ISDN services and features. The system provides ISDN support for up to 84 (R8csi), 1000 (R8s), and 7000 (R8r) telephones.

The system also supports connection to an Electronic Tandem Network. Different Electronic Tandem Network locations are connected via analog or digital tie trunks. For example, a Digital Signal Level 1 interface can act as a high-speed (1.544 Mbps) digital backbone for voice and data communications between Electronic Tandem Network locations.

For more information, see Chapter 9, ''Networking Solutions.''

Power

Depending upon the cabinet style, the system can accept a variety of AC or DC power depending on your region. The system can operate without requiring a power transformer in almost any part of the world.

During a power outage, battery backup differs depending on the cabinet style. See the *DEFINITY ECS System Description* for more information.

Industry Applications

3

Overview

The applications presented in this chapter explain how DEFINITY BCS and GuestWorks meets communications challenges in various industries. Though the specific requirements of the industries vary, the general information presented here can help to generate ideas. Even if none of the applications precisely match your situation, the examples may suggest creative solutions you can apply to suit your needs.

In the financial services industry, for example, banks, brokerage houses and insurance companies now offer many of the same services. In this chapter, industries are presented in the broadest terms, with little regard for overlap. For example, the insurance industry can be considered under both the "Health Care" and "Financial services" headings.

In most cases it is difficult to consider the system without also considering its array of options. Many of the solutions discussed in this chapter are enabled by optional hardware and software. The system is the essential integrating platform that coordinates and enhances these specialized tools. Even if your intention is to purchase a basic system, it is important to gain some understanding of the many options the system provides so you can eventually capitalize on those advantages.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

Education (K – 12 and small colleges)

Municipal school districts and small colleges seek to:

- Ensure reliable telephone service
- Coordinate information and services
- Communicate easily with the outside world
- Reduce costs but still serve student needs
- Plan for expansion and innovation.

Ensure reliable telephone service

The system can service up to 25000 telephones. The size and efficiency of the system allows small colleges to generate revenue from student phone service, which offsets the cost of other services.

The reliability of the system is without equal. The system's automatic backup features, maintenance tests, and line-monitoring functions work proactively to protect your investment. These and related features identify potential difficulties well before the system's operations might be compromised, further enhancing the high reliability inherent in the system architecture.

Coordinate information and services

Many colleges have large campuses or are composed of a network of scattered buildings and offices. Efficient connections among the many elements are essential to the integrity of the institution. A variety of options can help coordinate information and services from many locations:

- Wireless and cordless telephones allow librarians, technicians, and clerks to easily search for things while talking to the person requesting the search.
- A single voice messaging system can be accessed by several sites using the Centralized Voice Mail via Mode Code Interface feature. This allows satellite campuses or offices to access common directories and handle messages as if they were all on the same campus.
- Audio conferencing equipment allows teachers and managers to easily participate in policy-making meetings, regardless of location.

- The security of all buildings can be coordinated and enhanced in the following ways:
  - The system efficiently routes emergency calls to security staff.
  - PassageWay® Direct Connection logs incoming calls and pinpoints the location of the telephone making the call, using the system's name/number display capability.
  - Call management software logs the speed of the response so that response times can be measured and improved.

Communicate easily with the outside world

Most schools receive many incoming calls. The number of calls also fluctuates a great deal — increasing just prior to the start of a semester, for example. Often, the callers are unsure which department or to which individual they need to talk. The communications system must, therefore, be able to handle fluctuating call volume while satisfying each caller's particular needs. Here is how the Basic Call Management System's tools meet these needs:

- Automatic Call Distribution routes incoming calls to agents in a hunt group. As additional calls come in, they are routed to an available agent or placed in queue if an agent is not available. If there are too many calls in queue, the calls can be automatically routed to standby agents during periods of high-call volume. Display telephones alert the standby agents that they are handling overflow calls.
- Call management software keeps statistics on the number of abandoned calls, average length of call, average wait time, and other activities so you can manage your hunt groups and track productivity.

Reduce costs while meeting student needs

Educators now have many options for making the most of their resources while providing a top quality education for many students. The system provides efficient, integrated access to both the school and to worldwide resources:

- Speakerphones allow distant experts to share knowledge with students in the classroom.
- Desktop Conferencing Systems enable students to see and speak with one another and to collaborate on documents. They can create and jointly edit documents that may reside on only one computer in one location. The students can work together as if they were all seated at the same table.
- Wireless telephones can help teachers and students solve software problems because access to technical experts is easier.

These tools allow schools to form partnerships with each other world-wide, enhancing the overall quality of education they offer while spawning new revenue-generating opportunities.

Plan for expansion and innovation

Schools must be at the forefront of communications innovation, so it is important to use a platform that can accommodate rapidly-evolving requirements. The system is:

- Capable of handling multimedia calls
- Compatible with many different products from many different vendors so that it fully integrates all of your tools and options
- Designed to accommodate existing and emerging standards and protocols.

Financial services

The financial services industry may include banking institutions, credit unions, insurance companies, mutual funds companies, and brokerage firms. These types of businesses are nearly indistinguishable from one another in some areas. Deregulation, technological advances, and strong competition induce each business to offer a broad range of financial services. Many of these services are automated to improve customer service and to make the most of available resources.

Financial service providers worldwide seek to do the following:

- Control costs
- Automate routine transactions
- Network regional and global offices
- Upgrade customer service.

Control costs

Cost savings are inherent in many DEFINITY solutions. Using an automated attendant in place of an employee to answer routine calls reduces payroll expenses. Beyond the day-to-day savings that automation and networking provide however, the system includes some capabilities that directly affect your operating costs.

Automate routine transactions

In many countries, as much as a quarter of all bank transactions are conducted by telephone. For related businesses such as brokerage houses, the percentage can be much higher. Typically, at least half of these calls are from callers requesting routine information. The Call Vectoring feature and recorded announcements allows you to set up an automated attendant that screens calls for your busy customer service representatives. For example, the attendant may handle incoming calls by offering the following options to the caller:

- If you know your parties' extension, press 1.
- For business hours, press 2.
- For interest rates, press 3.
- To receive a loan application, press 4.
- To speak to a customer service representative, press 0.

For the calls channeled to your customer service representatives, the system holds overflow calls in queue for the next available representative. It generates reports that identify peak calling periods, how much time representatives are spending on calls, and which lines are being used. This allows you to maintain high quality customer service while adjusting the size and working hours of your staff.

Network regional and global offices

If your company has offices scattered throughout different regions or countries, it is probably important to you that your procedures are the same everywhere. Your customers probably expect consistent service wherever they go and however they choose to interact with your firm. Networking the offices together is an obvious solution, because it also allows the offices to share information. To do this, your system must be flexible enough to accommodate a variety of requirements and equipment.

As part of this network, a single voice messaging system can be accessed by several sites using the Centralized Voice Mail via Mode Code Interface feature. This allows satellite offices to access common directories and handle messages as if they were all on the same site.

QSIG global networking

Lucent Technologies has been a leader in providing equipment compatible with QSIG, a standard for vendor-independent networking. QSIG has been adopted by the International Standardization Organization, ensuring its acceptance worldwide. Lucent's QSIG Global Networking allows you to network different types of systems throughout the world. If, for example, you have acquired an office in another country that uses non-Lucent equipment, QSIG Global Networking allows you to incorporate that equipment into a DEFINITY network. The systems can work seamlessly together, through shared features, flexible numbering plans, and simplified network operations and management. This interface supports voice and data basic call setup, which includes Number Identification and Transit Counter.

Here are some additional ways the system can help you serve your customers:

- The system's open architecture allows you to easily change features to meet the changing needs of your customers. For example, setting up a small telemarketing group often requires making only minor modifications to your measured hunt group.
- Some organizations have calls received after business hours relayed to an office that is still open for business in another part of the world. This saves the cost of an around-the-clock staff and keeps your customers in touch with your best people.

Government

Government agencies must project a professional image to their constituents while controlling costs. Government agencies want to do the following:

- Provide valuable service to the public.
- Keep in contact with various offices.
- Provide flexible telephone services to employees.

Provide valuable service to the public

The Call Vectoring feature provides an auto-attendant to callers that need to connect directly to specific governmental agencies. The feature allows employees to spend time doing valuable work, not answering calls for other departments. Call Vectoring can also be used as a help line to provide information about services provided by the local agency.

Keep in contact with various offices

Government offices tend to be spread about in various locations. With a system at each location, features such as uniform numbering allow easy access between locations. Speed Dialing is another feature that is valuable and can save time when calls are made regularly between groups.

As part of this network, a single voice messaging system can be accessed by several sites using the Centralized Voice Mail via Mode Code Interface feature. This allows satellite offices to access common directories and handle messages as if they were all on the same site.

With the wireless mobility solutions offered with the system, workers can move freely around their department but still remain in contact with calls from the public or from associates.

Should an emergency occur at the office, the Crisis Alert feature allows employees to contact local emergency agencies quickly. When this call is made, the attendant as well as digital stations and up to three pagers are notified of the call so that when emergency personnel arrive, someone can help them find the exact location of the emergency. This feature is critical at government locations that tend to have several different buildings.

Provide flexible telephone services to employees

Since many government agencies are now requiring private industries to telecommute, government agencies can also participate by using features such as the DEFINITY Extender and Call Forwarding. Government employees can work from their homes, thereby saving energy and the cost of a permanent office. The call coverage feature also routes unanswered calls to either an assistant or to voice messaging, ensuring that calls are always answered.

Health Care

The health care industry may include providers, insurance companies, employers, patients, researchers, pharmaceutical companies, and the government.

Health care administrators worldwide seek to do the following:

- Maximize resources to reduce or contain costs.
- Improve response time in a busy urban environment.
- Maximize productivity and efficiency of high-salaried professionals.
- Provide highly efficient service, without losing the human touch.
- Promote wellness and satisfaction with easy access to information within the community.
- Improve accessibility to specialized medical care.
- Maintain skills and collaborative relationships regardless of location.

Maximize resources to reduce costs

For individual health care providers, cost containment and reduction is the key to survival and growth. The rules of health care payment are changing, and providers must keep the costs of care down without sacrificing quality.

Beyond providing quality care — always an overriding concern — health care's primary goal is to maximize resources through efficient operation. Savings can be realized in reexamining everything from staff size and operations to the number and type of rooms provided.

The system can provide a variety of options to fully use available resources. It can turn the telecommunications investment into a seamless network for managing and monitoring incoming calls and voice messaging.

Improve response in a busy environment

Hospitals deal with a high percentage of emergencies, both in the hospital and in the outside community. Hospitals can improve their patient services and emergency response by:

- Mobilizing staff during disasters or emergencies outside the hospital
- Improving response to emergencies inside the hospital
- Improving emergency room response for the critical cases arriving by ambulance.

The system provides the following services to hospitals:

- Paging systems provide an effective way to broadcast information about emergency situations throughout an entire department or facility. Visual paging ensures that the hearing-impaired are also notified of emergencies.
- Wireless telephones help nurses stay in touch with doctors and technical experts while carrying out their duties.

The system helps hospitals improve emergency services without adding staff.

Maximize productivity and efficiency

Many health care facilities participate in an integrated health network consisting of numerous hospitals, clinics, doctors' offices, laboratories, and other medical facilities. Although they are often autonomously managed, these multiple sites have to function as a single organization to keep costs down and enable the facilities to be financially successful.

The staff of an integrated health network includes administrators, nurses, technicians, physicians, and support personnel. Many members of the staff are active on multiple shifts, and are seldom confined to an office.

Health care facilities need to do the following:

- Maintain close communication links between distant facilities, and include related organizations such as suppliers and clinics.
- Reduce unnecessary overhead paging.
- Improve response to emergencies.
- Provide an efficient way to communicate non-emergency information to busy mobile staff.

The system can help health care facilities maintain productivity and efficiency with the following products and features:

- Standardized systems, networked with four-digit dialing between locations, can ensure that the staff wastes no time adapting to the communications system as they go from location to location.

- Voice Messaging systems can reduce personal paging and eliminate telephone tag when the staff must continuously leave messages and wait for returned calls.
- Basic call management system packages can support the facility's busiest offices such as the business office, hotline groups, clinics, and admissions offices.
- Lucent Technologies Call Accounting System for *Windows*® allows health care facilities to chargeback telephone equipment and usage to doctors, clinics, and offices.
- Lucent Technologies offers an array of wireless solutions that provide an effective way to communicate with nurses, doctors, and others who must be mobile.
- Outside labs, pharmacies, physicians' practices, vendors, and other organizations who frequently deal with the health care facility can obtain guest mailboxes on the voice messaging system. The health care institutions can thus avoid toll charges that should be paid by others.

By using Lucent Technologies products, health care facilities can reap the following benefits:

- Improved communication between staff members
- Better response to true emergencies
- Improved staff efficiency and satisfaction.

Provide highly efficient phone service

Many health care facilities encounter problems responding to the large number of incoming calls. Callers are frequently put on hold for long periods of time before representatives are available to help them.

Health care facilities need to do the following:

- Eliminate the frustration and negative perceptions of the facility that are experienced by callers.
- Improve the quality of service, without increasing costs.
- Using staff to do the jobs for which they were trained.

The system can provide the following capabilities to the health care industry:

- Recorded announcements and the Call Vectoring feature give callers access to basic information 24 hours a day, seven days a week.
- PassageWay products allow a caller's record to appear on the agent's screen as the call rings on the phone, based on caller input or Calling Line (or number) Identification. This eliminates the need for the agent to ask

identifying questions and enables him or her to locate the records more easily. It also improves service by enabling the agent to greet the caller by name and to address the issues more quickly.

- Basic Call Management System allows the business office supervisor to assign the appropriate number of representatives and analyze call volume to identify opportunities for improvement. The system can also be used by the supervisor to determine whether representatives are responding quickly to callers.

By using Lucent Technologies products, health care facilities can provide more efficient phone service and in return, reap the following benefits:

- Faster response to callers
- Accurate staffing
- More personal service
- Higher productivity
- Improved image of the health care facility.

Promote wellness and satisfaction with easy access to information within the community

Health care facilities measure their success by the satisfaction level of their services. Facilities need to provide the best “first impression” of the hospital. In most cases, it is in the best interest of the health care provider and insurer to promote wellness to keep hospitalization costs down.

Health care facilities need to do the following:

- Provide easy access to wellness information.
- Educate the public about preventative measures.
- Encourage the public to take control of their health issues in a timely manner.
- Provide referrals for health care professionals and specialists.

The system offers an easy way to help the health care industry. Voice messaging allows callers to leave non-emergency questions or messages for later callbacks, so that callers can get personal attention.

Lucent Technologies products help health care facilities to provide first-rate personal care in a cost-efficient manner.

Improve accessibility to specialists

Medical professionals often need to contact specialists in a particular field but are restricted because of time, distance, and expense. They provide better medical care by doing the following:

- Consulting with experts, sometimes during surgery
- Overcoming boundaries of distance — by consulting with any physician, no matter the location.

Desktop conferencing systems can be used in patients' homes by home health nurses to confer with physicians about patient conditions. This enables more patients to be cared for outside the hospital, and reduces the need for the very ill to travel to the hospital or physician's office.

Maintain skills and collaborative relationships regardless of location

In the health care industry, there is an urgent need for multiple sites to operate as one and for medical professionals to collaborate remotely, so they can provide top-quality health care to patients in rural areas. Doctors and nurses must also stay abreast of technological innovations in the field and continue their educations.

The system can play a critical role in connecting remote and sparsely populated communities with the advanced centers in health care. This technology enables the same level of sophistication in the rural settings as that available in the urban medical centers by doing the following:

- Improving communications
- Improving staff satisfaction
- Increasing personnel skills
- Providing improved patient care
- Reducing time and travel expense.

The Lucent Technologies Desktop Conferencing System can help with the following:

- Continuing medical education

  Doctors can learn at their desktops, without having to pay for expensive travel bills and time away from their offices and homes.

  Medical students can be educated at remote sites. Distance learning can help medical students assigned to rural clinics learn from doctors in hub hospitals and medical centers.

- Remote consultations by non-physician medical staff, which are often difficult to arrange in rural areas.

Nutritionists, for example, are particularly scarce in remote settings. A nutritionist can use video to communicate with a patient in a distant facility, showing food models of healthy portions and being face-to-face with the patient for better understanding.

Hospitality

The hospitality industry is composed primarily of hotels, motels, and restaurants.

Hospitality facilities worldwide seek to do the following:

- Control costs.
- Improve operating efficiency and safety.
- Enhance guest services.

Control costs

Hospitality providers must contain costs to maintain a profit and to stay competitive in the industry.

Two ways to help control costs are as follows:

- Separate long-distance calling privileges.

  Hotel and motel guests frequently place long-distance phone calls from their rooms, while providers do not allow staff members from accessing long-distance phone service.

- Charge guests more accurately for terminated calls.

  Hospitality providers need the ability to detect short duration calls (that is, calls that terminate before the specified answer detection time-out), enabling hotels to more accurately charge guests for these calls.

GuestWorks provides the following capabilities to the hospitality industry to help control costs:

- World Class Routing features that allow hotels to separate long-distance calling privileges for guests and administrative staff.
- Answer Detection enhances the DEFINITY system's ability to detect short duration calls.
- INTUITY Lodging Call Accounting (a co-resident application developed by Homisco for North America) and Xiox stand-alone call accounting provide accurate and flexible call accounting for guest room billing.

Improve operating efficiency and safety

Hospitality service facilities continuously deal with fluctuating economies, and must maintain maximum efficiency to ensure smooth operations and productive employees.

Three ways hotels can improve operating efficiency and safety are as follows:

- Simplify guest billing for phone expenses.

  Hotels and motels need simplified guest billing, along with the ability to generate guest phone records.

- Powerful voice-messaging service.

  Guests and administrative staff need to be able to leave voice mail or faxes for other guests and staff members. Guests can have callers leave messages or faxes for them privately, without having to involve the front desk.

- If a guest makes an emergency call, the system automatically notifies the desk attendant, identifying the room that placed the call.

DEFINITY products can provide the following capabilities to the hospitality industry to maintain maximum operating efficiency:

- The Call Detail Recording feature works in combination with system adjuncts to generate guest records and call costs records.
- INTUITY Lodging allows guests and the administrative staff to create, store, send, and receive voice or fax messages. Spoken prompts guide the user through each step of the procedure. The system can be administered for a variety of languages.

Enhance guest services

Hospitality providers must constantly find ways to enhance guest services. Staff must work hard to make guests feel comfortable and to maintain and uphold a reputation for outstanding service. Today’s harried consumers want to get top-quality service for their hard-earned income.

Hotels can enhance guest services as follows:

- Review guest requests for services.

  Hotels and motels need a way to review guest requests and ensure that guest’s needs and requests are met in an efficient manner by the staff.

- Connect to internal computer systems.

  Staff can provide better customer service by linking the telephone system to the hotel’s internal computer system for registration information and voice messaging features.

- Provide phones with modem hookups and conference call capabilities.

  The 6416D+M and 6424D+M digital telephones provide simultaneous voice and analog data capabilities over a single pair of wires.

- Provide voice and fax messaging services.

  The INTUITY Lodging system allows guests to receive voice messages and fax transmissions. Guests can retrieve the voice messages from any location and print fax transmissions at a centralized fax machine.

DEFINITY products can provide the following capabilities to the hospitality industry to enhance guest services:

- Guest activity reports containing information on items such as requests for wake-up calls and delivery of these calls. These reports can be printed in hard-copy form or can be viewed at the Administration terminal. These reports help the administrative staff to ensure that guest requests for services are not overlooked, and that guests get prompt and efficient service from the staff.
- INTUITY Lodging allows guests and the administrative staff to create, store, send, and receive voice or fax messages. Spoken prompts guide the user through each step of the procedure. The system can be administered for a variety of languages.

Specialized solutions

The following features for hospitality services are also available (see "Appendix A, Features" for more information):

- Analog Caller ID
- Attendant Backup
- Attendant Vectoring
- Automatic Selection of DID Numbers
- Automatic Wakeup
- Controlled Restrictions
- Crisis Alert to Attendant, Digital Station, Pager
- Daily Wakeup
- Dial by Name
- Do Not Disturb
- Dual Wakeup
- Emergency Access to the Attendant
- Housekeeping Status
- Integration of voice/fax messaging with property management systems

- Mixed Numbering
- Names Registration
- Room Change/Swap
- Suite Check-In
- VIP Wakeup.

For more information about hospitality solutions, see Chapter 7, "Hospitality Solutions."

Legal/Professional

In the legal and professional business (such as consultants and advisors), you must juggle a variety of client types while keeping track of time spent on projects. Professional businesses seek to do the following:

- Keep track of client costs.
- Stay in contact at different locations.
- Provide a high level of service.

Keep track of client costs

Call detail records sent to Call Accounting systems keep track of client calls so that employees can keep track of time spent on cases or consultations. Long-distance calls made by lawyers or assistants can use authorization codes to track calls by account numbers.

Stay in contact at different locations

In many cases, lawyers and consultants have meetings at different locations where audio teleconferencing is valuable for resolving issues. Voice messaging allows users to get their messages from any location so they can keep up on recent developments while away from the office.

Provide a high level of service

Many legal offices now provide free access to legal advice using recorded announcements via an auto-attendant procedure. This projects a professional image to potential clients. Clients can also use the Dial-by-Name feature to contact personnel directly if the caller knows a name, but not an extension at the office.

Manufacturing

Manufacturing is typically a no-nonsense business that requires an exact and accurate bottom line. Manufacturing industries want to do the following:

- Keep in contact with vendors and suppliers.
- Remain mobile within a factory location.
- Provide a safe environment for employees.
- Expand telephony services as the business grows.

Keep in contact with vendors and suppliers

As vendors and suppliers change, factories must stay in contact to keep costs of goods at the lowest level. Least-cost routing (World Class Routing) means that your employees will save money while making important long distance calls. Telephone features such as Speed Dialing allow employees to place calls quickly and efficiently.

DEFINITY AUDIX voice messaging ensures that calls from vendors and suppliers will not be missed, especially since critical components in the manufacturing process change so quickly. Voice messages can be retrieved from any telephone on the system, or from remote locations.

Remain mobile anywhere in the factory

With the wireless mobility solutions offered with the system, workers can move freely around the factory, but still remain in contact with calls from the suppliers and vendors, or from associates.

Provide a safe environment for employees

With the machinery used in the manufacturing industry, accidents do happen. Should an emergency occur in the factory, the Crisis Alert feature allows employees to contact local emergency agencies quickly. When this call is made, the attendant as well as digital stations and up to three pagers are notified of the call so that when emergency personnel arrive, someone can help them find the exact location of the emergency. This feature is critical in a large factory.

Expand telephony services

With support for up to 25000 stations, the system can be expanded as your business grows. You can add more telephones for your engineering and purchasing departments as new opportunities arise.

Real estate

The real estate industry is a fast-moving and mobile business that requires snap decisions. The real estate industry wants to do the following:

- Be flexible with locations and personnel.
- Provide a professional image to clients.
- Be available at a moments notice.

Be flexible with locations and personnel

Real estate agents often have two offices: one at their main corporate location and one at their home. Features such as the DEFINITY Extender and Call Forwarding allow agents to keep up with their phone calls. At the main office, paging is a key to locating an agent quickly to close a deal.

Provide a professional image to clients

The telephone system is often an agent's biggest ally. Agents are often on the phone when other calls come in. With voice messaging, calls are not missed; clients can leave messages for the agent and be guaranteed that the message is delivered.

When clients call in to an office, the Dial-by-Name feature can be offered to route calls to the correct agents. All the client has to do is enter the agent's name using a touch-tone keypad, and the call is routed immediately.

Be available at a moments notice

When conference rooms are unavailable for closing deals, you can use the terminal translation initialization feature to keep some telephone ports in reserve to serve as a temporary conference line. You plug in a telephone, enter a feature access code and security code, and the telephone is activated.

Retail

The retail industry is a fast-moving, high-pressure business that requires employees to produce at a high level. Retail industries want to do the following:

- Improve sales while containing costs.
- Provide a professional image to customers.
- Expand resources as opportunities arise.
- Stay in contact with corporate locations to keep up to date with current trends.

Improve sales while containing costs

More and more, sales employees must make long-distance calls to gather information when making sales. Least-cost routing (World Class Routing) means that your employees will save money while making important long distance calls.

With the wireless mobility solutions offered with the system, sales associates are not confined to their immediate locations, but can move easily from department to department to answer sales call questions. The system also provides loudspeaker paging so that employees can be contacted from any location in the building.

Provide a professional image to customers

Retailers are finding that customers are doing more shopping over the telephone than in the past. Businesses must provide an increased level of service through their communications system. Features such as Call Vectoring and Recorded Announcements to route calls to specific departments or to a group of agents where orders are taken provide callers with quick access to what they need.

Since the employee base in the retail industry has a high turnover, the system provides video tape training for your employees. Your employees can quickly become proficient at using the telephone system.

The voice messaging system gives sales associates a professional way to keep up with all of their calls, both from customers and vendors. Sales associates can also use mailing lists to send out broadcast messages to other associates to notify them of new sales procedures or events.

Should an emergency situation occur at your store, the Crisis Alert feature allows employees to contact local emergency agencies quickly. When this call is made, the attendant as well as digital stations and up to three pagers are notified of the call so that when emergency personnel arrive, someone can help them find the exact location of the emergency. This feature is critical in a large department store.

Expand resources as opportunities arise

With support for up to 25000 stations, the system can be expanded as your business grows. You can add more telephones for your telephone orders department, or add new telephones for new sales departments.

Stay in contact with corporate locations

The system provides audio teleconferencing so that executives and sales associates can stay in contact with other corporate locations. This is a cost-effective way to quickly communicate new information to many locations.

Systems located in different geographical regions can be connected using the QSIG Basic Networking service and the Uniform Dialing Plan (UDP) software.

As part of this network, a single voice messaging system can be accessed by several sites using the Centralized Voice Mail via Mode Code Interface feature. This allows satellite offices to access common directories and handle messages as if they were all on the same site.

Telephone features such as speed dialing give employees easy access to other stores. This is valuable when trying to locate merchandise for customers.

Wholesale distribution

The wholesale distribution industry includes both merchants and agents. Merchants buy and sell merchandise, while agents limit themselves to presenting the merchandise and negotiating its sale. Some wholesale distribution companies serve both functions, depending on the circumstances. Most wholesale distribution companies are relatively small, and face increasing competition from larger firms and even from manufacturers themselves. Therefore, most wholesalers cannot easily raise the prices of their products. Continued success requires that they reduce costs and offer more services to both suppliers and customers.

Wholesale distributors seek to do the following:

- Provide convenient access to product information.
- Automate or streamline ordering procedures.

Provide convenient access to product information

DEFINITY AUDIX allows retailers to get product information at the touch of a button. For example, when a clothing retailer calls the wholesaler's product information number, an auto-attendant procedure presents the caller with the following options:

- For information about women's clothing, press 1.
- For information about men's clothing, press 2.
- For information about children's and young adult's clothing, press 3.
- For information about shoes, press 4.
- To speak to a representative, press 5.
- Or, simply enter the extension number of the person you are trying to reach.

You can also have DEFINITY AUDIX call customer service representatives to notify them when they receive voice messages from special customers.

Automate or streamline ordering procedures

The system offers a wide range of features which allow customers to order via phone call, fax, or automated voice messaging.

Desktop Solutions

4

Overview

The communications needs of the people in your company may vary widely. Some may need only basic telephone service. Others may need effective messaging services to save valuable time. Still others may require data communications and access to a variety of host and personal computers.

DEFINITY BCS and GuestWorks brings voice communications, data communications, visual communications, and messaging together on the desktop, and lets you customize types of service for various individuals.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

Telephones and workstations

A wide variety of telephones are available with the system, ranging from basic single-line telephones to sophisticated workstations that integrate voice, data, image, and video communications. Your configuration might incorporate a mixture of telephone types based on the various users' job functions. The telephones and workstations are easy to use and attractive while giving you the ability to tap into the power of the system.

The telephones fall into three basic families — analog, Digital Communications Protocol (DCP), and ISDN-BRI. These terms describe how each type of telephone communicates with the system. These families of telephones are designed to accommodate the types of communications various users require. All telephones have touch-tone dialing and the message-waiting lamp for notification of messages.

Analog (single-line) telephones

Single-line telephones are an economical choice for users who do not handle many calls and do not use modems and fax machines extensively.

All signals between analog telephones and the system are in analog form over a pair of wires. Only one incoming call can ring at a time, but the telephone can actually handle two calls — one active and one on hold. Depending on the particular telephone, you can alternate between two calls or set up a three-way conference using the switchhook or flash button. You can access voice features by either entering access codes from your touch-tone keypad or pressing feature buttons. Several models of analog telephones are available.

DCP telephones

Digital telephones using digital communications protocol (DCP) employ digital transmission for integrated voice and data signals and control signals. Transmission is over a connection consisting of one or two pairs of wires. Each connection supports one signaling channel and two information (voice and data) channels.

DCP telephones are used most effectively by those who have a high volume of calls, require access to multiple applications or databases, use system features heavily, or require messaging services. These telephones can be used with personal computers to expand their capabilities.

These telephones provide the full range of features on your desktop. In addition to multiline and multifunction capabilities, they provide access to integrated voice and data applications and messaging services. Some models include displays. DCP telephones can actually save you money by reducing the number of lines, modems, and ports that would normally be needed for analog facilities.

ISDN BRI telephones

Like the digital DCP telephones, ISDN telephones transmit voice, data, and control signals digitally. With the ISDN telephones however, the transmission employs the world-wide standard BRI protocol between the system and the telephone.

Also, like the DCP telephones, these telephones can be used with personal computers to expand their digital capabilities. The system’s family of ISDN telephones include several models which include unique features such as call logs and personal directories.

Telephones for the global marketplace

With help from our many global customers, Lucent Technologies has developed telephones, the 6200-series and the 6400-series, to meet the demand for two-wire telephones in the global marketplace.

6200-Series telephones

The 6200-series of analog telephones provides a high-quality, low cost, Lucent-branded analog telephone solution. These telephones include a range of options that meet the business customers' need for higher feature/function analog telephones, and the business customers' demand for more basic feature/function telephones. The 6200-series provides a native analog solution that supports all Lucent Technologies switches. These new telephones have been designed in anticipation of the new Federal Communications Commission (FCC) ruling that mandates volume control on handsets and speakers by the year 2000.

Table 4-1 lists the features available with the new 6200-series analog telephones.

Table 4-1. 6200-Series Analog Telephone Features

Feature	Model 6210	Model 6218	Model 6220
Colors	Deep Gray, White	Deep Gray, White	Deep Gray, White
Desk/Wall Mount	Yes	Yes	Yes
Data Jack	Yes	Yes	Yes
Flash Button	Yes	Yes	Yes
Message Waiting Lamp	Yes	Yes	Yes
Mute Button	No	No	Yes
Set Hold (with LED)	Yes	Yes	Yes
System Hold	No	Yes	Yes
Positive Disconnect	Yes	Yes	Yes
Redial	Yes	Yes	Yes
Enhanced Redial	No	Yes	Yes
Repertory Dialing	No	Yes (10)	Yes (10)
Ringer Volume Control	Yes	Yes	Yes
Handset Volume Control	Yes	Yes	Yes
Personalized Ringing	No	Yes	Yes

Continued on next page

Table 4-1. 6200-Series Analog Telephone Features — *Continued*

Feature	Model 6210	Model 6218	Model 6220
Speakerphone	No	No	Yes
Tone Dialing	Yes	Yes	Yes
Program Keylock	No	Yes	Yes
Local Display	No	No	No
Active Dial Pad	No	No	No
External Power Supply	No	No	No

6400-Series telephones

The two-wire, DCP 6400-series digital telephones feature new styling and a pullout instruction card. The 6400 telephones also include the following additional features:

- Date and time display.
- A feature button which allows switchhook control of a headset.
- *Group Listen* capability, which allows you to use your handset or headset normally while others in the room listen in via speakerphone. This two-way handset, one-way speaker mode allows you to serve as a spokesperson for a group.
- *Telephone Self Administration* capability, which allows you to program feature buttons on the telephone yourself.

There are several models of the 6400-series telephones:

- 6402 – Automatic management of two call appearances, one-way speaker with group listen, and fixed features plus shifted dial pad for 12 additional features. It is ideal for areas where there is minimum use, such as reception areas, copy rooms, file rooms, or warehouse locations. It has a built-in one-way (listen-only) speakerphone that facilitates on-hook dialing and listening to voice mail or broadcast messages.
- 6408+ – Eight dual-indicator call appearances, speakerphone with group listen option, three-way volume control, message waiting lamp, and fixed buttons for hold, conference, transfer, and redial. This telephone is for employees with call coverage responsibilities who need multiple line appearances and extensive features.
- 6408D+ – All of the features of 6408+, two-line, 24-character display, and four softkeys (menu, exit, previous, and next).

- 6416D+ – Sixteen dual-indicator call appearances, jack for XM24 expansion module. This telephone uses the I2 channel of the analog adjunct connected to the Tip/Ring port of the module.
- 6416D+M – With all of the features of the 6416D+ telephone, this telephone also provides a built-in RJ11C jack as an interface to analog telephone devices (such as a telecopier or a modem in a laptop personal computer), and an RS232 data interface to a PassageWay Direct Connection. These sets cannot be wall mounted.

  The analog port on the 6416D+M is an important feature for business people that use laptop computers with analog modems. Users can connect their laptops to the 6416D+M for data, and use the telephone for simultaneous voice, all through a single pair of wires.
- 6424D+ – Twenty-four dual-indicator call appearances and a jack for XM24 expansion module. This telephone uses the I2 channel of the analog adjunct connected to the Tip/Ring port of the module. This telephone is for the busy executive or executive assistant when extensive call handling and call coverage flexibility are vital.
- 6424D+M – With all the features of the 6424D+ telephone, this telephone also provides a built-in RJ11C jack as an interface to analog telephone devices (such as a telecopier or a modem in a laptop personal computer), and an RS232 data interface to a PassageWay Direct Connection. These sets cannot be wall mounted.

  The analog port on the 6424D+M is an important feature for business people that use laptop computers with analog modems. Users can connect their laptops to the 6424D+M for data, and use the telephone for simultaneous voice, all through a single pair of wires.
- XM24 – Expansion module that provides 24 dual-indicator call appearances. One version of the XM24 supports the 6416D+ and the 6424D+ telephones, and another version supports the 6416D+M and the 6424D+M telephones.

International icons and languages. International icons are used on the telephones, and buttons are available in several languages, as are the messages on display sets. You can also use a user-defined table to customize the translations. Additional international portability is provided with downloadable handset transmission parameters.

Voice features

With the system’s voice features, the employees in your company can easily place a simple telephone call while still having access to powerful features. These features range from the basics (such as Call Forwarding, Hold, Transfer, and Conference) to more sophisticated features intended for particular situations or users.

These features can be accessed in a variety of ways. For example, some can be accessed by pressing a fixed-feature button on the telephone. Many others can be accessed by dialing an access code or by pressing a programmed button on the telephone. The following sections show a few examples of how particular voice features can help your employees to handle calls more efficiently.

Abbreviated Dialing

Provides lists of stored numbers users can use to do the following:

- Place local, long-distance, and international calls.
- Activate features.
- Access remote computer equipment.

Users dial the list number and the one-, two-, or three-digit number associated with the phone number the user wants. The number is then automatically dialed by the system. A frequently-called number can be stored on an abbreviated dialing button that users need only press once to make the call.

Automated Attendant

Automated Attendant uses call vectoring commands to allow callers to enter the extension of the party they wish to reach. The call is routed to that extension by the vector.

Bridged Call Appearance

Allows calls to be handled from more than one telephone. A bridged call appearance is set up by administering a primary extension and the button number associated with it on a two-lamp button on another telephone. One way this feature is most often used is by secretaries or assistants who answer or handle calls to the primary extension (an executive, for example). When the primary extension receives a call, the bridged call appearance flashes or rings and the call can be handled as if the primary extension user was answering it. You can have up to 64 bridge call appearances.

Call Coverage

The Call Coverage feature ensures that your calls are always answered and that callers rarely, if ever, receive a busy signal. Call Coverage is so flexible that external calls can be routed to one group of attendants and internal calls to an entirely different group.

In some respects, Call Coverage serves as an assistant who screens your calls. It automatically redirects calls to other telephones and messaging services, allowing you to delegate or defer calls as needed.

You can redirect calls according to five status conditions: Active, Busy, Don't Answer, Cover All, and Send All Calls. If you are using one of your several call appearances, the system considers you to be "active." If you are using all your available call appearances, the system considers you to be "busy." If the call goes unanswered, the status is "don't answer." Sometimes you might need to assign a secretary or other colleague to "cover all calls," or you may "send all calls" to a permanent voice messaging system or to an assistant.

Call Coverage lets you redirect calls to suit any or all of these criteria. For each telephone, you can have up to four coverage paths. A path is a set of alternate extensions to which a call can be sequentially transferred. Each path can be composed of as many as three extensions, arranged in order of preference. A redirected call immediately goes to the first choice extension. If the first choice is not available, the system tries the second choice and then the third choice, if necessary.

Many people prefer to redirect all of their calls to the same answering points under all conditions, and need only one coverage path. If a secretary is available to cover all calls, even if you are available, the other criteria can be ignored. If you prefer to answer your own calls however, you will probably require Busy, Don't Answer, and Send All Calls coverage. Send All Calls lets you redirect your calls by pressing a single button or by dialing an access code.

Time-of-Day call coverage allows you to redirect calls to different lead-coverage paths at different times of the day and on different days of the week.

For example, you may want to be available in the evening hours during a special project. You might also want calls directed to the office during the day, and have all other calls directed to DEFINITY AUDIX. By specifying the appropriate lead-coverage paths, you can have the call redirection flexibility you need.

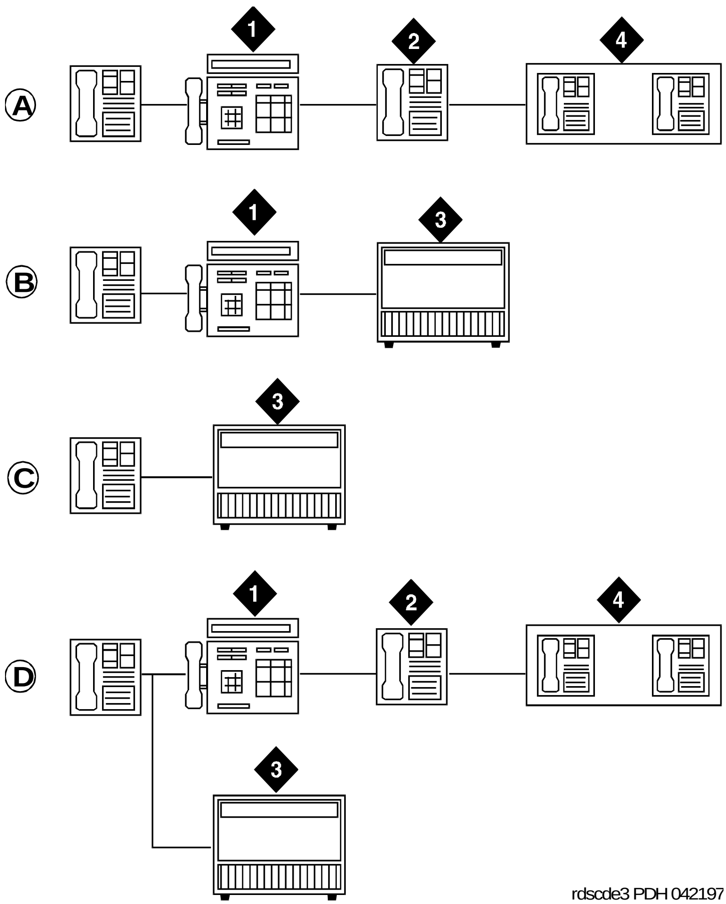
Telecommuting features, such as Coverage of Calls Redirected Off-Net, allow you to have call coverage redirected to a remote site. This is useful if you have a home office to which you want calls sent. For more information on remote call coverage/forwarding, see the *DEFINITY® ECS Administrator's Guide*.

Coverage paths for a manager

Figure 4-1 shows four coverage paths you might need as a manager. The example assumes that you do the following:

- Receive many external calls.
- Share an assistant with two other managers.
- Prefer to answer your own calls when available.
- Travel frequently.

External calls are important because they are usually from customers and require personal attention as they arrive. Internal calls are also important, but often need not be dealt with immediately by you or an assistant. In either case, Send All Calls is useful because it allows you to redirect all calls immediately when you are not available. This saves the caller the annoyance of waiting for several rings before being able to talk to someone or leave a message. The Call Coverage arrangement shown works well for many managers. Note that the same coverage path is used for all external calls because these calls need special attention even when you are unavailable.



A)	External Calls: Active, Busy, Don't Answer	1)	Assistant
B)	Internal Calls: Cover All	2)	Clerk
C)	Internal Calls: Active, Busy, Don't Answer	3)	Voice Messaging
D)	Internal Calls: Send All Calls	4)	Message Center Group

Figure 4-1. Typical Call Coverage Options

Conference

Allows multi-appearance telephone users to set up six-party conference calls without attendant assistance. Single-line telephone users can set up three-party conference calls without attendant assistance.

Directory

Allows users with display-equipped telephones to access the system database, use the touch-tone buttons to enter a name, and retrieve an extension number from the system directory. The directory contains the names and extensions assigned to all telephones on the system.

Group Listen

Simultaneously activates the speakerphone in listen-only mode and the handset or headset in listen-and-speak mode on 6400-series telephones. This allows a user to serve as spokesperson for a group. The user can participate in a conversation while everyone else in the room is listening to what is said.

Integrated Announcements

The system allows you to store recorded announcements (messages) internally on a circuit pack. The system's integrated announcements have the following characteristics:

- Easy to use. Announcements can be recorded and updated from any telephone. All announcement configuration is performed from the management interface (usually a PC using the DEFINITY Site Administration tool). Whenever someone changes an announcement, a log of the activity is kept. This can be useful in tracking unauthorized changes.
- Reliable. Even a power failure will not affect the integrity of your announcements. Because the announcements are stored digitally, voice quality does not degrade over time. There are no external boxes, messy cabling, or separate power supplies, and there are no tapes to jam or break.
- Flexible. Since the announcements are integrated within the system, the applications are almost endless. Announcements can be played to callers waiting for connection. They can be inserted into coverage paths to give out your hours of business. Applications like Call Vectoring were designed to take advantage of the power of integrated announcements.
- Ideal for a global market. Since you record your own announcements, any language that you are able to speak can be provided — even multiple languages on the same system. For example, your hotel guests can receive wakeup greetings in their native language.

Last Number Dialed

Allows a user to automatically redial the last number dialed. The system saves the first 24 digits of the last number dialed, whether the call attempt was manually dialed or dialed using Abbreviated Dialing. When the user presses the Last Number Dialed button or dials the Last Number dialed feature access code, the system places the call again.

Leave Word Calling

Allows internal system users to leave a short preprogammed message (usually “Call” with the calling user’s name, extension number, and the time of the call) for other internal users. When the message is stored on the system, the message lamp on the called telephone automatically lights. Leave Word Calling messages can be retrieved using a telephone display, Voice Message Retrieval, or AUDIX. Messages may be retrieved in English, French, Italian, Spanish, or a user-defined language.

Transfer Abort

Transfer Abort allows you to abort a transfer when you select another call appearance in the middle of the transfer operation or hang up. If you decide to cancel the transfer for any reason (that is, you get an important call, you dialed the wrong number), all you need do is select another call appearance or hang up. The original call you were transferring is put on hold, and you can then go back to that call at your convenience and re-initiate the transfer.

NOTE:
You cannot have the Transfer Upon Hangup feature enabled if you want to abort the transfer when hanging up.

This feature applies to DCP, Hybrid, ISDN-BRI and wireless telephones, but not to analog telephones.

Whisper Page

Allows an assistant or colleague to bridge onto a telephone conversation and give a user a message without being heard by the other party or parties on the call.

Messaging services

The system offers a variety of voice messaging services that allow you to leave, send, and receive messages quickly, accurately, and conveniently. The messaging services include the following:

- DEFINITY AUDIX (used for non-hospitality applications)
- INTUITY AUDIX
- INTUITY Lodging (used for hospitality applications)
- Octel 100 Messaging (formerly Messaging 2000).

These messaging services can be purchased with the system and can be fully integrated with the Leave Word Calling and Call Coverage features. A message-waiting lamp on your telephone lets you know when messages are waiting from any of the messaging services.

The AUDIX system and call coverage

Often a DEFINITY AUDIX and/or INTUITY AUDIX system is set up as the last point on a call-coverage path, as in Figure 4-1. A secretary or colleague who answers a redirected call intended for you can also transfer the caller to your AUDIX mailbox. The caller may prefer to leave voice mail for you if the message is personal, lengthy, or technical.

Many other options are available:

- A caller can redirect a call from the AUDIX system to an attendant.
- A caller can transfer to another extension instead of leaving a message. This feature is dependent on the type of integration used with the AUDIX system (control link or display set).
- The AUDIX automated attendant can answer all calls to the company.
- The AUDIX automated attendant can send calls to various extensions. In this case, callers are instructed to enter keypad commands to direct the call.

Message-Retrieval options

With the message-waiting lamp on their telephones, employees always know when they have messages. Messages can be retrieved using the Display Retrieval feature. This feature allows users having digital telephones with displays or a personal computer integrated with a telephone to display messages.

Teleconferencing products

How much of your time do you spend in meetings — or traveling across the building, across town, or across hundreds of miles to get to a meeting? How often was time lost because vital information was left in someone's office? Meeting by phone or teleconferencing offers an attractive alternative. Meetings are suddenly more convenient and easier to schedule, and travel expenses are greatly reduced. The Lucent Technologies SoundStation® products provide you with all the benefits of voice conferencing.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

The Lucent Technologies SoundStation and SoundStation EX® Audioconferencing Systems enable a group of people in a conference room to share their conversation with others through a telephone connection. The SoundStation equipment permits natural conversation among many people—whether the conversation is loud or soft, or from a standing or sitting position.

The equipment's full-duplex technology allows conferees to speak at the same time, thus eliminating the tendency conventional speakerphones have of *clipping* — failing to transmit the beginning or ending sounds made in conversation. The SoundStation systems adapt automatically to changing room and telephone line conditions to permit natural, two-way conversations without distortion. This allows you to be heard without straining to hear what others are saying.

Integrated components and a stylish tripod design make the console an attractive yet unobtrusive conference table centerpiece (Figure 4-2).

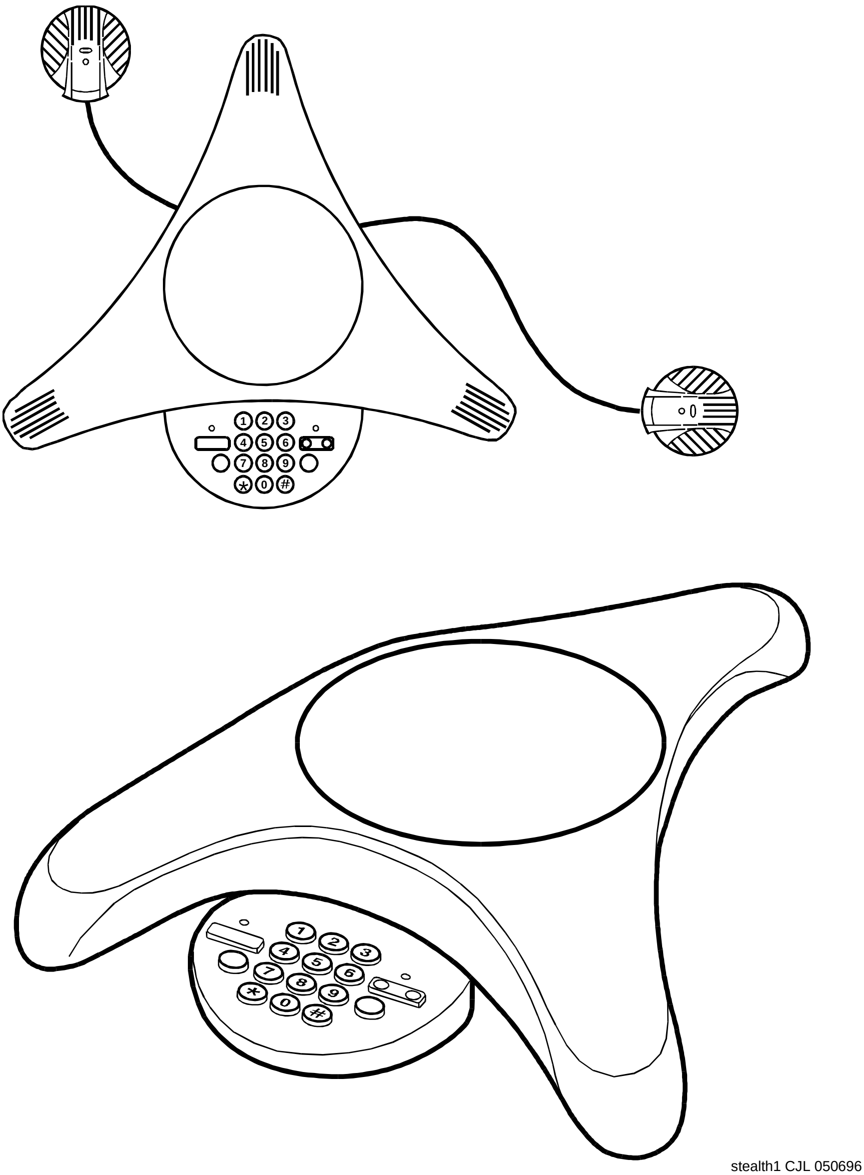


Figure 4-2. SoundStation EX (with external microphones) and SoundStation

SoundStation speakerphone

The Lucent Technologies SoundStation has three microphones and a digitally-tuned speaker that provide 360-degree coverage whether you use the system in an office or in a conference room. The built-in keypad includes a mute button and a flash key. An additional port allows you to connect the speakerphone to a tape recorder.

The speakerphone is available in both analog and digital models. The analog model connects directly to an analog telephone line. The digital model can be used with any digital telephone (DCP or BRI) that has an adjunct speakerphone jack.

The system is simple to install and use. You plug the phone line into a small wall module plugged into a power outlet. A single cable from the wall module to the console reduces tabletop clutter. The console works like a regular telephone.

SoundStation EX speakerphone

The SoundStation EX includes all the features and functions of the SoundStation. It accommodates larger conferences by including two palm-size external microphones that can be positioned up to 6 feet (1.8 meters) on either side of the center console. An optional lapel microphone is available for stand-up presenters.

Mobility Solutions

5

Overview

Many businesses strive to improve customer service and increase profits by controlling costs and staff size. That means employees have to be more productive, more responsive, and often more mobile. Wireless solutions allow you to control costs by reducing time and resources spent on paging employees, interrupting work to find a phone, rushing to answer calls, or being tethered to the desk waiting for an important call. Reliable wireless tools remove the fear of losing customers who could not wait to reach you directly.

Lucent Technologies is a major provider of wireless solutions for business. Lucent offers a range of options from single-zone to multi-zone cellular business systems that greatly enhance the flexibility of wireless telephones.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

Single-Zone mobility solution

TransTalk® (Figure 5-1) is a multiline, single-zone solution using the 900 MHz wireless technology. This allows you to roam up to 700 feet (230 meters) from the base station. It effectively covers up to 500,000 square feet (150,000 square meters) in most business environments. Either a DCP or hybrid line circuit pack provides the interface for the TransTalk telephones.

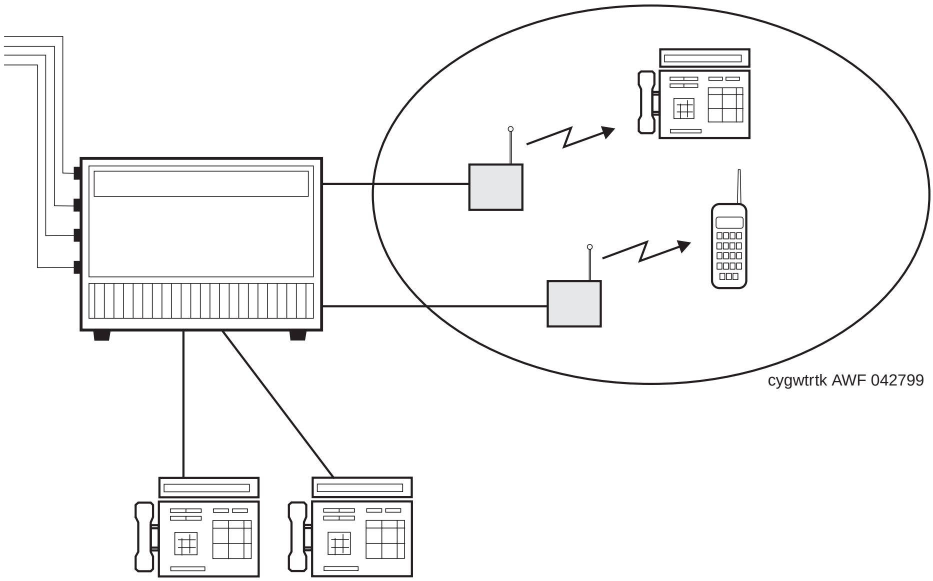


Figure 5-1. TransTalk

TransTalk is available in two configurations:

- Complete system, consisting of a carrier that holds up to six radio modules, wireless telephones, and corresponding charging cradles, radio modules, and holsters
- Stand-alone, consisting of a single radio module, a wireless telephone, charging cradle, and holster.

TransTalk telephones have the following features:

- Clear voice quality
- Consistent privacy and secure operation
- Intercom, conference, and transfer
- 10-line capacity
- 10 programmable buttons
- 30 handsets per zone
- Automatic registration
- Trouble lights
- Extended battery life
- Battery pack and optional battery backup
- Battery charger (1 1/2 hours)

- Dynamic power adjustment
- Mute button
- Mobility range test capabilities
- 3 hours talk time
- 22 hours of standby time
- Noise cancellation/sound enhancements
- Vibrator alert.

Dual-Zone mobility solution

TransTalk Dual Zone provides an easily installed, easily maintained solution that doubles the coverage (two full 500,000 square feet zones) of a multizone system with a **manual hand off** and allows you to support a small number of users.

The standard TransTalk Dual Zone system consists of the following:

- Two carrier assemblies
- Twelve radio modules
- Multiline 9031DZ pocket phones with holster and charging cradle.

A smaller, stand-alone TransTalk Dual Zone system consists of the following:

- Two radio modules
- One charging cradle
- Multiline 9031DZ pocket phones with holster and charging cradle.

Transtalk Dual Zone telephones have the following features:

- Clear voice quality
- Consistent privacy and secure operation
- Intercom, conference, and transfer
- 10-line capacity
- 10 programmable buttons
- 30 handsets per zone
- Automatic registration
- Trouble lights
- Extended battery life
- Battery pack and optional battery backup
- Battery charger (1 1/2 hours)
- Dynamic power adjustment

- Mute button
- Mobility range test capabilities
- 3 hours of talk time
- 22 hours of standby.
- Noise cancellation/sound enhancements
- Vibrator alert.

NOTE:
Dual-Zone and Single-Zone Transtalk radio modules can operate in the same coverage area and can be installed in the same carrier.

Multi-Zone mobility solutions

Lucent Technologies offers two robust systems that will keep you in touch with customers, coworkers, and suppliers wherever you go in your office complex—desk-to-desk, office-to-office, or office-to-warehouse. In both systems, overlapping zones allow you to move about freely without changing phones (Figure 5-2). The phone connection is "handed off" from one transmitter to another as necessary.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

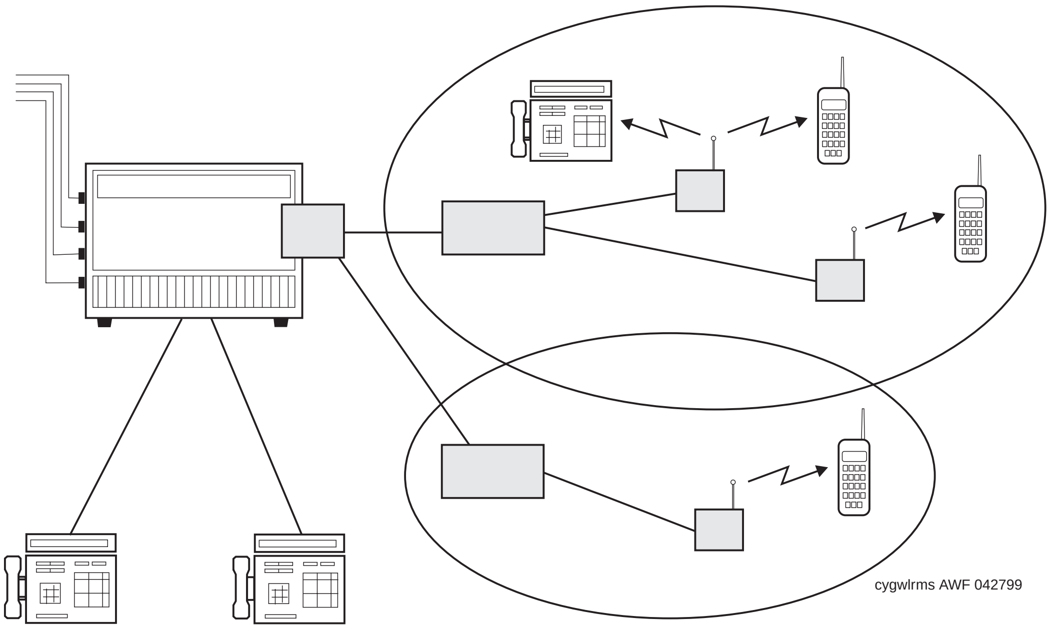


Figure 5-2. Multi-Zone Mobility Solutions

The DEFINITY Wireless Business System (DECT Release 2) connects to the system as adjunct devices, thus providing flexibility in setting up applications. It uses an international industry standard that is more common in some parts of the world.

This system features Lucent's Wireless System Engineering Expert Design System. This patented software, which is unique in the wireless industry, analyzes the building or campus space and determines how the wireless system should be configured. It precisely recommends where the base stations should be located within the structure or structures. The software effectively eliminates the most difficult aspect of wireless implementation and ensures maximum efficiency and lower life cycle costs.

DEFINITY Wireless Business System

The DEFINITY Wireless Business System (international only) uses the latest Digital Enhanced Cordless Telecommunications (DECT Release 2) technology — the leading European standard adopted in over 70 countries worldwide. It operates in the allocated 1880-1900 MHz spectrum. Since this is an unlicensed frequency range, there is no charge for air time when operating in this band.

The DEFINITY Wireless Terminal WT 9610 is used with this system. The phones have superb voice quality, an alphanumeric display, and can access most DEFINITY features through use of a small, lightweight handset. The battery has 8 hours of talk time, with 80 hours of standby power. A menu offers nine different languages.

The DEFINITY Wireless Business System has the following features:

- Telephone communications with on-site mobility
- No air time charges
- Multi-zone seamless handover between calls
- Secure encrypted speech
- Capacity for up to 16320 users
- Capacity for up to 128 base stations
- Processing power for 7000 to 40000 calls per busy hour (depending on the system configuration)
- Coverage of up to 64.6 million square feet (6 million square meters)
- Advanced ISDN Display features
- X-Station Mobility (wireless phones which are remote over a trunk interface are controlled by the system as if they are directly connected to the system).

DEFINITY Wireless Business System

The DEFINITY Wireless Business System (DWBS) is an in-building wireless communications system. It offers mobility to medium and large business customers by integrating wireless capabilities into the switch and provides a pocket size wireless terminal. DWBS offers solutions for today's in-building mobility needs in the United States, Canada, and Central and South America as follows:

- Increase Customer Satisfaction

  Calls can be answered immediately, and customer needs can be addressed more quickly, no matter where the employee is located.

- Increase Employee Productivity

  Employees can be more productive when they can be mobile without cutting their communications links. Clerks may need to access information in a nearby file cabinet; warehouse workers can check inventory to fill a customer order; supervisors can oversee operations without missing important calls.

- Overcome Communications Obstacles

  People can communicate quickly and easily. Previously it may have been difficult or impossible to provide telephones in some areas of a building or a temporary location. For example, a building may be difficult or costly to wire, a disaster recovery situation requires a communications system that must be set up and be running immediately, or a temporary project such as consulting or audits need telephone access.

- Cost Savings

  Mobility solutions can reduce the “hard” expense associated with returning calls, as well as the “soft” expense related to inefficient customer service and lack of employee productivity. It can save the cost of frequent moves, additions, and changes.

DWBS supports exceptional voice quality to a maximum of 1500 wireless users and mobility throughout a premise covering up to 2.4 million square feet. DWBS is an option in DEFINITY BCS and GuestWorks Issue 4 and later. It is not backward compatible. However, adjunct and networked alternatives are offered.

The 9631 wireless terminal (modest in size and weight) features two call appearances and a 4x16 display for soft-key and calling-party information. A messaging icon provides message waiting notification for the user. The 9361 ensures critical communication with its 8 hours of talk time and 100 hours of standby time.

The low-wattage, high-security, digital DWBS system offers the following features:

- Telephone communications with on-site mobility
- No air time charges
- Multi-zone seamless hand-over between calls
- Secure encrypted speech
- Capacity for up to 1500 users
- Digital transmission
- Remote maintenance
- Conformance to FCC and UTAM standards. No license is required to purchase and operate the system.

Computer-Telephone Integration Solutions

6

Overview

Telecommunications and information systems are the fundamental building blocks of most businesses. Whether a sale is being made, a question is being answered, or an order is being placed, the telephone is the primary communications medium. However, the information to make the sale, answer the question, or fulfill the order is stored in a computer.

If these two building blocks are closely integrated, your business will realize benefits that will redefine your standards for success and customer satisfaction. DEFINITY BCS and GuestWorks can integrate data processing, data communications, and voice communications.

The following products work with the switch to unite your computer and telephone in powerful ways:

- DEFINITY PC Console
- PassageWay Direct Connect.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

DEFINITY PC Console

Lucent Technologies' DEFINITY PC Console allows your DEFINITY BCS call attendants to handle incoming calls efficiently from their personal computer. Using the familiar Microsoft® Windows graphical interface, the attendants can easily keep track of how long callers have been on hold and for whom they are waiting. Attendants can monitor up to six calls at once. They need not fumble with pen and paper when handling calls, since they can make notes on their computers about what each caller needs. All this contributes to making a favorable first impression with your customers. Having the call processing software on the same computer with spreadsheet, word processing, or other software allows the attendants to stay productive between calls.

Your company directory is displayed on screen with busy extensions shaded. A variety of search functions are available, so attendants can find names and extensions easily. On-line photo identification allows attendants to quickly identify employees. Calls are transferred with the press of a button. On-line help makes it easy for attendants to remind themselves how to use the PC Console.

The PC Console is easily customized, so even if attendants from different shifts share the same computer, they can each preserve their preferences in the call processing environment. The PC Console is available in English, Dutch, Spanish, French, German, Portuguese, and Italian. The system also accommodates any language that uses the Roman alphabet and ASCII 128-character set. For example, if a Spanish-speaking attendant takes over for a French-speaking attendant, a single press of a button converts all labels, error messages, and on-line help to Spanish.

PassageWay Direct Connection Solution

Lucent Technologies' PassageWay products bring the telephone and the personal computer together into an integrated voice and data workstation that can greatly enhance communications. With PassageWay, you can efficiently process calls while accessing powerful voice and data features. It also permits you to connect to a variety of host computers and other PCs through the networking strengths of the system. PassageWay provides error-free data transfer between your personal computers and other shared resources. You can even create your own applications to take advantage of the PassageWay connection.

Combine the Power of the PC and the Telephone

The PassageWay Direct Connection Solution is ideal if you do the following:

- Spend more than an hour a day on the telephone generating, supporting, and servicing customers
- Rely heavily on your desktop PC to manage important contacts
- Want to boost productivity and customer service.

The PassageWay Direct Connection Solution links a desktop Windows PC with the switch. The easy-to-use Windows interface provides greater business communications capabilities than either the telephone or PC offers alone. This solution gives you valuable computer-telephony integration (CTI) benefits right out of the box, plus it is a bridge to a wealth of other great CTI applications. Microsoft

Windows Dynamic Data Exchange (DDE) and Telephone Applications Programming Interface (TAPI) support ensures that you can capitalize on future CTI technologies.

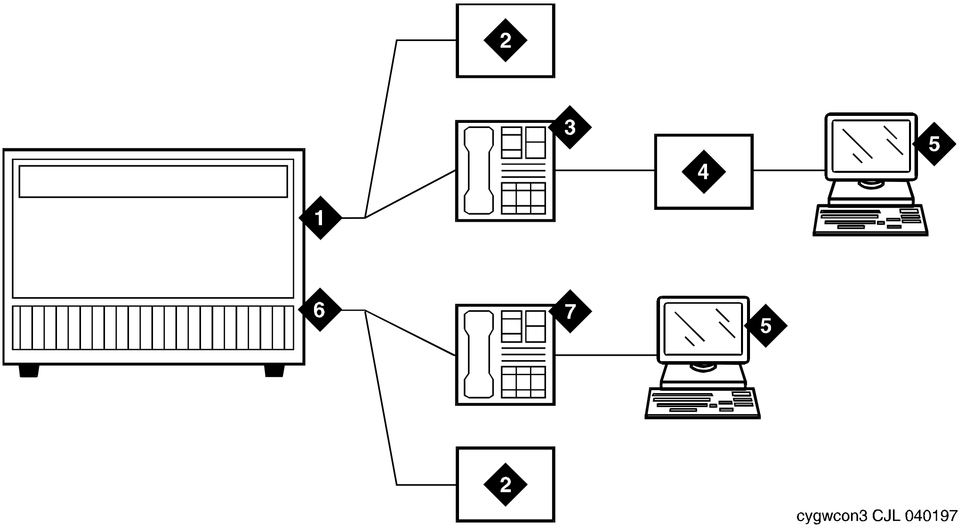
Make your PC work for you by having it do the following:

- Access and autodial phone numbers while you do other tasks,
- Instantly retrieve and display PC data about incoming callers. You save time digging for important numbers.
- Be better prepared to meet callers' needs and let your professionalism really shine through.

Turn your call center into profit centers. Departments that live on the phone such as sales, customer service, and purchasing, will work more efficiently, make more contacts, and produce better results because of CTI.

The PassageWay Direct Connection software consists of PC software (Service Provider and/or call control) and special Lucent Technologies PC-to-telephone hardware modules (DEFINITY 8411 telephones have built-in hardware).

You can choose to purchase the PassageWay modes with only our TAPI Service Provider software or you can add the Telephony Manager call control software.



1) 2- or 4-wire DCP Port
2) Auxiliary Power
3) DCP Telephone
4) Passageway
5) Personal Computer
6) 2-wire DCP Port
7) 8411 DCP Telephone w/ PassageWay

Figure 6-1. PassageWay Direct Connection Configurations

PassageWay Direct Connection Telephony Manager

The Telephony Manager software for Windows 95 is an application that allows you to control telephone calls (both incoming and outgoing) directly from your PC. You can make calls, answer calls, place calls on hold, hang up, transfer, set up conference calls, and view calling/called party information. With Caller ID, as the phone starts ringing, customer information can “screen-pop” on your PC. You can retrieve data stored in your database contact manager or your PassageWay Phonebook (included with the PassageWay Direction Connection Telephony Manager solution). When using the Phonebook, you can also automatically dial phone numbers with a click of your mouse. To automatically track every call you make or receive, PassageWay Direct Connection Telephony Manager solutions provides Log Manager. This displays information such as caller name; phone number; and duration, time and subject matter of call. Telephony Manager is not supported on Windows 98.

PassageWay Direction Connections and CTI Applications

You can convert your existing windows-based software into CTI applications by writing DDE links to PassageWay Direct Connection Solution. Or, add a "middleware" software application to dial and screen-pop from any windows application. If your application supports the Microsoft Windows TAPI standard, the PassageWay Direct Connections solution will probably work right out of the box since the PassageWay Direct connection also supports the TAPI standard. The following list contains a few of the CTI applications currently available from Lucent Technologies that enhance your Direct Connections Solution.

- FastCall® Middleware
  - allows existing applications to be quickly "telephony enabled"
  - provides rules for inbound "screen-pop" and call handling
  - previews outbound dialing
  - Windows 3.1/3.11, or 3.x, 95, 98, and NT
- SNAP™ Middleware with Enhanced Call Control
  - easy-to-install with your existing applications
  - provides "screen-pop" to your databases
  - dials any number from any windows application using an icon that is always on the current windows screen
  - Windows 3.1, 3.11, 95, 98, and NT
- PhoneLine® Corporate Directory
  - allows access to the latest telephone directly
  - quickly finds a phone number
  - automatically dials with one click
  - transfers and makes conference calls
  - stores pictures of individuals for security desks to validate employees
  - Windows 3.1/3.11, or 3.x, 95, and NT.

System Requirements

The system requires the following:

- IBM-compatible PC with a 486 or higher processor
- 8 MB RAM (16 MB recommended; 16 MB required with Telephony Manager R2)
- available serial port
- 3.5”, 1.44-MB, high-density disk drive (CD-ROM recommended)
- hard disk with 15 MB of space available (25 MB required with Telephony Manager R2)
- VGA or higher monitor
- A windows-compatible pointing device (mouse or trackball is recommended)
- Windows 95, 98, Windows NT (4.0). Note that Telephony Manager and PhoneLine are not supported on Windows 98.

Hospitality Solutions

7

Overview

GuestWorks, the hospitality offer of the DEFINITY product line, offers an array of features that enhance guest services and keep guests happy. You can thus enjoy robust hospitality functions on a state-of-the-art communications system.

NOTE:
In some countries, DEFINITY BCS is sold with the hospitality features instead of selling GuestWorks.

For example, GuestWorks can provide the following:

- Automatic wake-up for guest rooms. The wake-up call can be as simple as silence, or as elaborate as a custom sales message in the native language of the guest, tailored to the time of day and day of the week. There are several ways wake-up calls can be requested and delivered:
  - Assisted by voice prompts, guests can request their own wake-up calls.
  - Auto Wake-Up via confirmation tones allows a user to enter the request for a wake-up call using the telephone dial pad.
  - Dual Wake-Up allows two wake-up times to be entered.
  - Guests can also request a daily wake-up call if they need to be awakened at the same time every day during their stay.
  - The system also supports a VIP wake-up that alerts front desk personnel that they need to place a personalized wake-up call instead of letting the system do it automatically.
- PMS Insert/Delete digit streamlines dialing within a hotel that has multiple extensions sharing an extra leading digit in front of the room number.
- A check-in and check-out button on the attendant console. When a guest is checked in, the desk clerk presses the check-in button; the system prompts for an extension number, marks the room as occupied, and turns the telephone on. At check-out, the reverse happens.

- Suite check-in. This feature provides the capability to have the system automatically check-in several related extensions with one check-in command. This feature allows hotels that offer “suite” rooms with several phones the ability to check in all the phones associated with that “suite” at one time.
- Automatic selection of DID numbers. This feature allows the system to automatically choose a number from a list of available Direct Inward Dialing (DID) numbers that will be assigned to a guest’s room extension when checking in.
- Feature access codes to signify certain conditions. For example, maids can use the telephones in the rooms to change the room status from “dirty” to “clean and ready for occupancy.”
- TCP/IP connectivity. The switch uses TCP/IP to connect to the INTUITY AUDIX system.
- A Do-Not-Disturb feature that turns off ringing in a room, except for designated priority calls and automatic wake-up calls.
- Guest voice messaging, which unburdens attendants and provides guests with an important convenience.
- Crisis alert. Whenever someone in the hotel places a call to an emergency service agency (for example, 911), the attendant console, designated digital telephones, and up to three pagers are notified of the call. When emergency personnel arrive, someone can help them find the exact location of the emergency.
- Controlled Toll Restriction, which allows you to restrict some telephones from making toll calls. In this way hotels can provide free local calls, while still restricting toll calls.

For more information about GuestWorks features, see *GuestWorks®* and *DEFINITY® ECS Hospitality Operations*.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

Lodging establishments often use the following systems together:

- GuestWorks
- A property management system (PMS)
- INTUITY Lodging Voice Messaging
- INTUITY Lodging call accounting or Xiox call accounting.

Property management systems are used for making guest reservations, checking guests in and out, printing guest bills, and other accounting functions. INTUITY Lodging provides a variety of voice messaging and fax functions for guests and administrative staff, and includes flexible administration capabilities that simplify moves and changes. The call accounting system takes call records from the system and applies cost structures used for billing guests.

As the centerpiece of the hospitality communications network, GuestWorks continues to refine its integrating capabilities. For example, recent message tandeming enhancements make it unnecessary for the INTUITY Lodging voice messaging and the property management system to be directly connected to support the voice messaging application (See Figure 7-1). With this link, guest room updates for voice messaging are “tandemed” through the GuestWorks system between the INTUITY Lodging voice messaging system and the property management system. These systems are constantly exchanging and updating information to provide a seamless integration between the systems.

NOTE:
This integration feature does not affect the link between the INTUITY Lodging Call Accounting and the property management system. This link must remain intact so that the call accounting information is exchanged between the INTUITY system and the property management system.

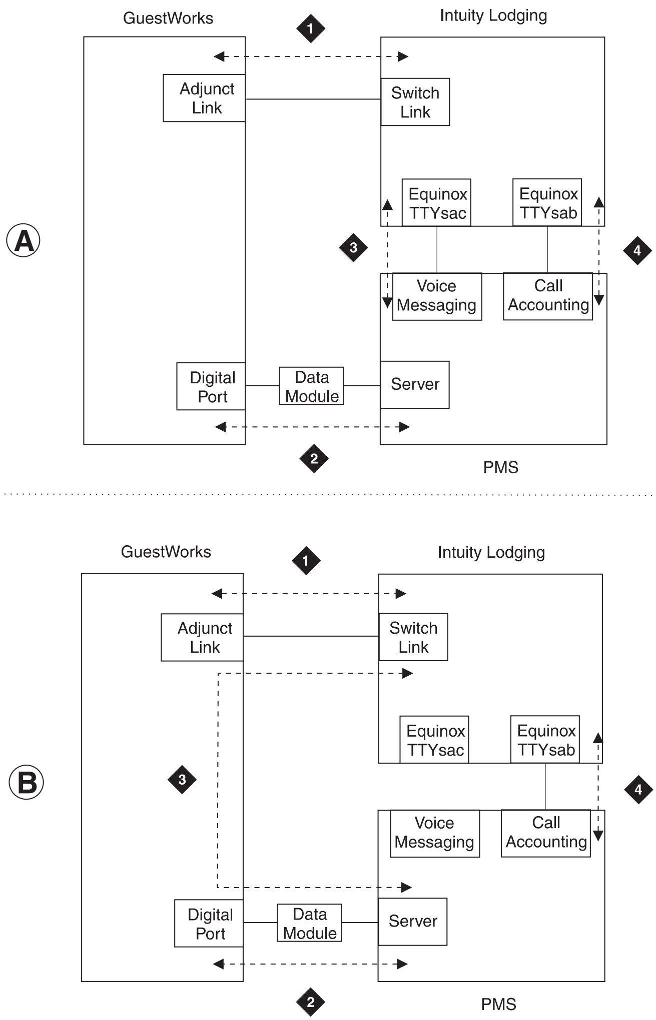
The general advantages of using GuestWorks in the Hospitality industry are presented in Chapter 3, "Industry Applications." The following sections provide a closer look at INTUITY Lodging, GuestWorks, the system’s communications with property management systems, INTUITY Lodging call accounting, and Xiox call accounting.

Switch/INTUITY/PMS link integration

GuestWorks can tandem messages between the property management system and INTUITY Lodging voice messaging so the systems need not be connected to each other. The following property management system messages are tandemed through GuestWorks:

- Check-In
- Check-Out
- Room Data Image (Database Synchronization)
- Modify (Guest Information)
- Add/Remove Text/Fax Notification Message (Message Waiting)
- Transfer/Merge Mailbox (Room Change/Swap).

NOTE:
This integration feature does not affect the link between the INTUITY Lodging Call Accounting and the property management system. This link must remain intact so that the call accounting information is exchanged between the INTUITY system and the property management system.



A) Standalone Link

B) GuestWorks Link

1) Switch-to-INTUITY Lodging Link (administrative voice messaging link)

2) Switch-to-PMS Link (control messages link)

3) INTUITY Lodging-to-PMS Link (mailbox control link)

4) INTUITY Lodging Call Accounting-to-PMS Link (call accounting records link)

Figure 7-1. Switch/INTUITY/PMS Link Integration

Hospitality enhancements

Recent enhancements to GuestWorks provide additional hospitality features and options. The primary enhancements are the following:

- Automatic Selection of Direct Inward Dialing Numbers for Guest Rooms
- Crisis Alert to Pager
- Suite Check-In
- Station Hunt Before Coverage.

Automatic Selection of Direct Inward Dialing Numbers for Guest Rooms

This feature allows the system to automatically choose a number from a list of available Direct Inward Dialing (DID) numbers that will be assigned to a guest's room extension when checking in.

With this feature, hotels can give a guest a phone number that is different from their room number, thereby protecting the guest's privacy. When a particular DID number is called, the call routes to the guest's room extension, and covers as if the room was called directly. Besides improving guest security, this eliminates the need for an attendant or front desk staff to extend a call to a guest room.

Crisis Alert to Pager

The Crisis Alert to Pager feature provides the capability to receive Crisis Alert messages on up to three digital pages. The pager call may include various information including information indicating that the call is an emergency and information about the caller. This information helps the person receiving the Crisis Alert page to know where to send the emergency team.

CAUTION:
This feature should be used in conjunction with the Crisis Alert to an attendant or digital station. It is not recommended as a replacement for the attendant or digital station because the paging message may not get an immediate response.

Suite Check-In

This feature provides the capability to have the system automatically check-in several related extensions with one check-in command. This feature allows hotels that offer "suite" rooms with several phones the ability to check in all the phones associated with that "suite" at one time.

Station Hunt Before Coverage

This option works with the Suite Check-In feature. With Station Hunt Before Coverage, the call routes to the other phones in the "suite" of rooms before going to coverage, if the primary number called is busy.

INTUITY Lodging

INTUITY Lodging is a messaging system designed especially for lodging establishments such as hotels or other lodging providers such as hospitals or colleges. The system supplies guests with electronic mailboxes that store voice or fax messages. INTUITY Lodging serves as a private answering machine for each extension.

Users are greeted with spoken prompts that guide them in pressing keypad buttons to make choices. Because touch tones are not needed to leave a message for a guest, outside callers may use rotary phones.

Hotel guests can leave messages for each other without going through the attendant. For incoming calls, an attendant transfers the call to the appropriate room. If the guest does not answer the call or if the line is busy, the call is automatically transferred to the guest's voice mailbox, where the caller can leave a voice message.

A message-waiting indicator on the guest's phone notifies the guest that the voice mailbox contains messages. Guests may assign a password for accessing messages remotely. They can retrieve and save messages from any telephone, on or off premises.

Calls are transferred to an attendant when any caller does the following:

- Presses **0** at any time (for assistance)
- Leaves a maximum-length message
- Stays on the line after leaving a message
- Is silent when prompted to leave a message.

Fax Messaging

With the Fax Messaging option, the caller can leave a fax by simply pressing a key when prompted and starting the fax transmission. The fax is stored until the guest, instructed by the system's voice prompts, does one of the following:

- Sends it to the Guest Services fax machine
- Prints it on an in-room fax machine
- Retrieves it into a portable computer
- Forwards it to another location.

Faxes can also be stored in the administrator's mailbox for later delivery to a guest. This occurs when someone sends a fax to the hotel, but not directly to a guest's mailbox. The administrator can either print the fax or send it to the guest's mailbox. Guests or administrators can also send faxes to multiple locations simultaneously.

Language options

Guests can hear voice mail prompts and menus in one of several languages. The current set of available languages includes the following:

- American English
- Arabic (female voice)
- Brazilian Portuguese
- British English
- Canadian French
- German
- Greek
- Japanese
- Latin American Spanish
- Mandarin Chinese
- Parisian French
- Russian.

Any or all of these languages may be installed, but only **nine** can be made available at any one time. The attendant enters the guest's desired language at check-in time. The guests will hear menus and prompts in their chosen languages after logging in to retrieve messages. Contact your account representative for language options.

Call accounting

The INTUITY Lodging Call Accounting package (an integrated offering from Homisco) takes call records supplied by the system, puts the records into a standard bill format, and sends the billing information to the property management system. When guests check out, their long distance calling charges are printed automatically on their bill. This gives you better control over telephone usage revenue.

Additional features

INTUITY Lodging includes many features similar to those of DEFINITY AUDIX and INTUITY AUDIX. (For more information, see Chapter 10, "Voice Messaging Solutions.") Guests may record their own personal greetings, for example, and administrators can send broadcast messages to many recipients simultaneously. Mailing lists can be created for specific groups staying at a hotel (for example, a convention) so that special messages can be sent to only that group of guests.

When guests change rooms, their voice mailboxes can move with them. Attendants can change room A with room B, transfer room A to room B, or merge room A with room B so messages are not missed. Passwords and backup features protect privacy and ensure that information is not lost.

System administrators have many options for controlling the operation of INTUITY Lodging. For example, they can do the following:

- Set fax options.
- Customize the voice prompts.
- Designate call coverage paths.
- Define conditions under which callers are automatically sent to an attendant.

Xiox Call Accounting

The Xiox Call Accounting works with GuestWorks as an adjunct. Xiox call accounting allows hotel management to use their property's telephone system as a major source of revenue by generating the information they need to make important decisions about their network and usage. The benefits of Xiox call accounting are as follows:

- Instant automatic bill back of guest room telephone charges
- The ability to interface with most property management systems for instant posting of call charges to guest folios
- A surcharge on a cost per call or percentage basis can be automatically added to guest room calls
- Standard or custom reports allocate administrative phone costs by department
- Analysis of traffic to assure the proper mix of lines, trunks, and services
- An increase in efficient employee telephone usage.

Xiox call accounting focuses on the following areas:

- Revenue generation
  - — Bill back call costs to guests
  - — Automatically mark up call costs by percentage and/or flat fee on a cost per call and/or cost-per-minute basis.
- Cost allocation
  - — Accurately charge telecommunications costs to the proper department
  - — Monitor misuse and abuse
  - — Improve employee productivity.
- Traffic engineering integration, analyzing lines, trunks, and services
  - — Spot traffic trends and define future projections
  - — Instantly redesign your telecommunications network and recommend cost saving opportunities at the press of a key
  - — Establish the most cost effective telecommunications network of lines and services.

Xiox call accounting supports up to 6000 extensions at a single location.

Data Management Solutions

8

Overview

DEFINITY BCS and GuestWorks are designed for fast, efficient, and reliable movement and management of data. All information transmitted through the digital system is carried in a digital format. Analog signals — both voice and data — are converted to digital form before being switched. Analog data compatible with data modules and fax machines can be transmitted through the system at speeds up to 28.8 kbps. Digital data can be transmitted at speeds up to 64 kbps per channel.

Data communications capabilities

Whether your data environment is asynchronous, synchronous, or a combination of both, the system's data-switching capabilities can greatly enhance your company's data communications. Using the system to switch your company's data has many possible benefits:

- It can greatly reduce the number of terminals and amount of cabling required.
- It enables employees to gain needed access to host computers, applications, and databases.
- It provides connectivity between different data environments that your company may have — asynchronous, synchronous, and personal computer environments.
- Voice and data are integrated and transmitted over the same wires; employees can exchange data and discuss it over the phone at the same time.
- Your data communications system will benefit from many of the system's capabilities. For example, voice features such as Abbreviated Dialing, Queuing, and Automatic Route Selection can also be applied to data communications. The system's networking strengths can expand data connectivity to wider areas. The system's management capabilities can monitor and control your data communications.

The system can be used in a variety of data applications. The applications listed below are just a few examples of the many ways in which you can use the system to improve your data communications:

- Switched asynchronous host computer access
- Switched synchronous host computer access
- Personal computer networking
- Switched video conferencing
- Fax networking.

See your local distributor for information on how you can make the system's data communications capabilities work for you.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

Data management features

The system offers a number of data management features to help control your data environment and to allow users quick and convenient access to data. Appendix A, ''Features'', contains a list of these data management features. The following list introduces a few of these features:

- Administered Connections automatically establishes an end-to-end connection between two data end points. An administered connection can be either permanent or scheduled. The feature supports Auto Restoration (preserving the active session) for connections routed over Software-Defined Data Network trunks and an administrable retry interval (from 1 to 60 minutes) to reestablish a connection. The resulting benefits are increased reliability of your data networks and improved disaster recovery.
- Alphanumeric Dialing enhances computer dialing by allowing you to place a data call by entering an alphanumeric name, making dialing both convenient and user-friendly. When an alphanumeric name is entered from your terminal, the system 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 you had entered the digits.
- Default Dialing enhances computer dialing by allowing you to place a data call to a preadministered destination by simply entering a carriage return at the "DIAL:" prompt. This gives you a simple method of dialing that number.
- Data Call Setup enables you to set up data calls (at any of the industry-standard rates) using a telephone or a computer keyboard.

- Data Hot Line enables you to administer a data module so that when the module goes off-hook the data call is immediately placed to the preassigned number. This feature may also be used to restrict a data module to the assigned number only.
- Data Protection prevents disruption of data transmissions by the system's other features or tones. Both the originating and terminating ends of the call are protected.
- Data Communications Access allows you to communicate with a computer via analog trunks.
- Host-Computer Access allows data endpoints with data modules to access a computer directly.

Digital interfaces

The system offers powerful digital interfaces for voice, data, and integrated voice/data transmission. Digital Communications Protocol, a key part of the system's digital architecture, provides integrated voice and data communications between terminals and the system.

The system supports a wide variety of bit-oriented signaling formats on Digital Signal Level 1 (1.544 Mbps) facilities, compatible with local CO services, nodal network services (such as AT&T MEGACOM* services), and services conforming to European Conference of Postal and Telecommunications standards in the international marketplace.

The system also implements both standard ISDN interfaces: ISDN Consultative Committee for International Telephone and Telegraph Primary Rate Interface and Basic Rate Interface. With both interfaces, the system delivers the advantages of full end-to-end ISDN connectivity to every desktop.

Digital Communications Protocol

Digital Communications Protocol (DCP), similar to ISDN-Basic Rate Interface, has been the architectural foundation for both of Lucent Technologies' digital systems and has for many years provided advanced ISDN-like functions to Lucent Technologies systems by integrating voice and high-speed data. DCP continues to serve as a key digital interface for the system.

Like ISDN-Basic Rate Interface, DCP defines the communications interface between a terminal and the system. It consists of two 64 kbps information (or bearer) channels and a separate 8 kbps channel (referred to as a data channel) for signaling and control information. Out-of-band signaling via the data channel allows the information channels to be used for clear-channel transmission.

* Registered trademark of AT&T.

DCP's framing structure allows voice, data, and signaling information to be transmitted with low overhead and be virtually free of errors. DCP transmits at a rate of 8000 frames per second or 160 kbps. DCP allows data and digitized voice to be multiplexed on two twisted pairs, terminating in a standard telephone jack.

The 6416D+M and 6424D+M DCP telephones are of special interest for data management. These telephones uses DCP to provide the full capabilities of digital voice and data, but also provides an analog port for data connectivity. This is important for business people who use laptop computers with analog modems. Users can connect their laptop computers to the telephone for data, and use the telephone for simultaneous voice calls, all through a single pair of wires.

ISDN-PRI

ISDN-PRI delivers ISDN service to the system for high-speed connectivity to the public switched telephone network and to other systems in a private or public network. It can also be used to connect to host computers that support the interface. PRI provides 24 64 kbps channels arranged in the North American ISDN standard of 23B plus D. That is, the 24 channels are divided into 23 bearer (B) channels at 64 kbps for information transmission and one signaling (D) channel at 64 kbps for control and signaling. Outside the United States, the system supports ISDN-PRI using the international E1 format, which provides 30B plus D.

The system offers applications that use the ISDN-PRI. See the Network Solutions section for information on these applications.

World-Class BRI

World-Class BRI provides an international BRI platform that offers multiple protocol options to meet specific country and application requirements. It provides access to Video Conferencing, Desktop Video Conferencing, Data Transmission, and other nonvoice-based applications that use BRI as a communication interface. Voice access is not supported, though voice features are not blocked for World Class BRI terminals.

World class BRI devices must be administered as the type "wcbri." You select a country protocol for each terminal that will use the feature. This selection determines both the code set modifications required to meet the national standards, as well as the terminal initialization procedures, if required.

World class BRI supports the following country protocols:

- Bellcore National ISDN-1 protocol in the United States (TR268)
- National protocols in Australia (AUSTEL TS013, Telecom Australia TPH 1962), Japan (NTT BRI) and Singapore (FETEX 150 TIF 218)
- ETSI NET 3 protocol (ETS 300 102) for use in most of Europe.

World class BRI supports multipoint (up to two devices per port) only for the Bellcore National ISDN-1 Country Protocol option.

Data modules

Data modules connect the system with other communications equipment, changing protocol, connections, and timing as necessary. The system supports the following data modules:

- 8400B Plus DCP data module
- 7400A DCP data module
- 7500B ISDN-BRI data module.

All of these data modules support industry standards and include options for setting the operating profile to match that of the data equipment.

8400B Plus data module

The 8400B Plus data module is a two-wire version of the 7400B Plus data module (no longer offered). The 8400B Plus dual-function data module provides full-duplex, asynchronous connectivity for DCP applications. The 8400B Plus emulates the industry-standard Hayes modems and works with host-connection software packages that use the Hayes command set. The data module gives you a choice of transmission speeds ranging from 300 bps to 19.2 kbps.

7400A Data Module

The 7400A dual-function data module provides full-duplex, asynchronous connectivity for DCP applications. It emulates the industry-standard Hayes® modems and works with host-connection software packages that use the Hayes command set. The data module gives you a choice of transmission speeds ranging from 300 bps to 19.2 kbps.

7500B Data Module

The 7500B data module gives you synchronous or asynchronous connectivity for ISDN-Basic Rate Interface applications such as video conferencing, fax, and personal computers at speeds up to 64 kbps. The 7500B features three connections: one to the Basic Rate Interface line to the system, one to a 7500-series telephone, and one (RS-232) to the computer. The module may be used stand-alone or in conjunction with a 7500-series telephone.

Networking Solutions

9

Overview

DEFINITY BCS and GuestWorks provide not only powerful voice and data capabilities, but connections to a variety of voice and data networks as well. Lucent Technologies has long been a leader in networking. The system continues to build on those established networking strengths to offer you network management features, network interfaces, a variety of private network configurations, and end-to-end ISDN capabilities. Lucent Technologies' leadership in developing and supporting open international networking standards is also apparent in the system's compatibility with the QSIG global standard.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

Centralized Voice Mail via Interswitch Mode Codes

Provides the capability to share a voice mail system among several switches using the Mode Code - Voice Mail System Interface. This feature provides a cost effective choice for multiple sites by eliminating the need for a voice mail system at each site.

A Centralized Voice Mail network can consist of DEFINITY BCS Issue 6 or later, DEFINITY ECS R8 or later, ProLogix™ R3 or later, Merlin Legend® R6.1 or later, and Merlin Magix™ systems. Either a DEFINITY BCS, ProLogix, or DEFINITY ECS switch must be the host for the voice mail system, and UDP and ISDN-PRI software are required at each site. See *DEFINITY ECS Administration for Network Connectivity* and *Merlin Legend Network Reference* for more information.

QSIG global networking

Lucent Technologies is the first vendor to provide compatibility with the QSIG global networking protocol. This means you can connect the system with other systems throughout the world. QSIG Global Networking was developed to comply with the QSIG standards developed by the European Computer Manufacturer's Association and the International Standardization Organization. It supports the ISDN-PRI connection from system to system as long as both systems support the same protocol.

QSIG is the generic name for a family of signaling protocols. The Q-reference point or interface is the logical point where signaling is passed between two peer entities in a private network.

DEFINITY BCS and GuestWorks supports only the QSIG Basic features, which are voice and data basic call setup with Number Identification and Transit Counter. QSIG Number Identification allows a switch to send and receive the calling number, and displays up to 15 digits for the calling and connected numbers across ISDN-PRI interfaces.

World Class Routing

The system has been designed to be a world-class system that meets the needs of all customers. One capability essential in meeting those needs is the ability to dial any location in the world, regardless of the dial plan used at that location. In recognition of this requirement, the system has been designed with World Class Routing.

World Class Routing is a powerful enhancement to the system's call-routing capabilities. ARS links several call-routing features to build a communications network capable of providing flexible call routing for any type of dialing plan, while accommodating changes in both international and local dialing plans.

The following are key components of World Class Routing:

- Digit Conversion converts a dialed public network number into a private network number and vice versa. Dialed numbers matching entries in the digit conversion tables are treated and converted. Converted calls can be routed via the most optimum route, resulting in reduced network charges and appropriate use of the private network.
- Toll Analysis compares a dialed number to entries in the system's list. Based on the results, calls may be restricted from completion.
- Automatic Route Selection digit analysis compares a dialed public network number with entries in the system's tables, mapping the number to a selected public network routing pattern.

- Automatic Alternate Routing digit analysis compares a dialed private network number with entries in the system's tables, mapping the number to a selected private network routing pattern.

World Class Routing supports the Automatic Route Selection and Automatic Alternate Routing as separate features, but through generalized administration applicable to both features, provides both of them with the same routing abilities. In addition, there are a number of capabilities that enhance the flexibility of routing in supporting your local and/or global calling requirements.

For example, 18-digit routing allows the system to determine call routing by analyzing up to 18 digits with no restriction on the grouping or format of the digits, thereby eliminating any assumptions about the use of a particular dialing plan.

International Direct Distance Dialed calls generally consist of an international access code, a country code, and a national number. Both codes may vary in length. Support for International Direct Distance Dialed calls eliminates any restriction on the grouping and formatting of digits on Automatic Route Selection numbers. Call routing is determined by the digits and the length of the dialed number.

Multinational World Class Automatic Alternate Routing allows the Automatic Alternate Routing number (Electronic Tandem Network number) to be any number of digits in length.

Digit conversion can be used to reroute numbers that are initially dialed to use ARS to be converted to use Automatic Alternate Routing and vice versa. This utility can analyze a maximum of 18 digits. In this way, destinations in a customer's network can be called using the public network number. This feature can also be used to reroute certain Direct Distance Dialed destinations to specified alternate destinations (such as intercept, attendant, or another Direct Distance Dialed number).

Network management features

The system has a variety of features that enable you to manage your network resources effectively. The following are just a few examples of features that can be used to manage your network:

- Automatic Route Selection
- Automatic Alternate Routing
- Time-of-Day Routing
- Subnetwork Trunking
- Generalized Route Selection
- Facility Restriction Level

- Bearer-Capability Class
- Authorization Codes.

Automatic Route Selection

Automatic Route Selection (ARS) routes public network calls on the most desirable (usually the most economical) trunking facilities available on your system when the call destinations are accessible through your public network.

The system supports up to 40 routing patterns. Each routing pattern consists of up to 16 routing preferences (types of facilities) set up in the order in which you want them checked when a call is placed. Typically, the least expensive facility will be first on the list; the most expensive will be last.

If Generalized Route Selection is not being used when a call is made, the system selects a routing pattern based on the digits dialed. The routing preferences in that pattern are checked in the order in which they were listed, and the first available facility is used to place the call. If no facility is available, the call can be queued until a facility becomes available.

Automatic Alternate Routing

Automatic Alternate Routing (AAR) allows private network calls to originate and terminate at one or many locations without accessing the public network. When a user dials an access code and phone number, AAR selects the most desirable route for the call and performs digit conversion as necessary. If the first choice route is unavailable, another route is chosen automatically.

The numbers called using AAR are normally private-network numbers. However, users can call a public-network number, a service code, an international number, operator access code, or an operator-assisted dialing number. With AAR and Subnet Trunking, users have a convenient way to place international calls to frequently-called foreign cities. Such calls route as far as possible over the private network, and then access the public network. This saves toll charges and allows users to use your private network as much as possible.

Time-of-Day Routing

Time-of-Day Routing allows you to select the most economical routing of ARS and AAR calls based on the time of day and the day of the week that a call is made. Up to eight Time-of-Day routing plans may be administered, each scheduled to change up to six times a day for each day in the week.

With Time-of-Day Routing, your company can take advantage of lower calling rates during specific times. If your company has locations in different time zones, you can maximize the use of your public or private network facilities by utilizing those facilities in the location that has the lowest calling rates at the particular time a call is made. You can also use this feature to change the routing patterns when an office is closed and to eliminate unauthorized calls.

Subnetwork Trunking

Subnet Trunking modifies the number dialed so an AAR or ARS call can route over different trunk groups that may terminate in switches with different dial plans. Subnet Trunking inserts digits, deletes digits, pauses, and/or waits for dial tone in digit outpulsing, as required, so calls route as follows:

- To or through a remote switch
- Over Tie trunks to a private network switch
- Over CO trunks to the serving CO.

Subnet Trunking is required on calls routing to or through a remote switch, regardless of the call's destination.

Generalized Route Selection

Generalized Route Selection gives you the capability to not only select the optimal call routing based on the dialed number, but also to select the appropriate facility based on the type of call. Generalized Route Selection enhances Automatic Route Selection and Automatic Alternate Routing by incorporating additional parameters such as the type of call to be used in deciding how a call is to be routed.

Different types of calls require the use of different types of facilities. For example, high-speed data calls must use digital facilities, whereas voice and voice-grade data calls can use either analog or digital facilities. The system uses Generalized Route Selection to differentiate between these and other types of calls and route them on the appropriate trunks. Based on the call types and available trunk facilities, voice and data calls may be routed over different trunk types or integrated on the same trunk group. The system also provides the capability to route calls based on the data format and the need for restricted or unrestricted facilities.

To select the appropriate trunking facility for a call, the system must know the type of call being made. To do this, each originating facility such as a telephone or data module has a bearer-capability class assigned. Some originating facilities, such as data modules, may have multiple bearer-capability classes. Each trunk group in the routing pattern is assigned a list of allowed bearer-capability classes. When a user makes a call, the system queries the originating facility for its bearer-capability class and then tries to route the call on a trunk group with a bearer-capability class that matches the bearer-capability class of the originating facility. If an exact match is not found, the system then tries to find a trunk group with a compatible bearer-capability class.

Since the system automatically chooses the right trunk based on the administration, the system's dial plan can be independent of the type of call being dialed. Users do not have to worry about dialing a different access number for different call types.

Facility Restriction Levels

Facility Restriction Levels are used to limit user calling privileges for incoming and outgoing calls. The Facility Restriction Level determines if a call attempt is permitted and which routes can be used or denied in the routing process. Through use of the system's management tools, eight Facility Restriction Levels can be assigned to telephones, computers, and trunk groups. The system does not require the Facility Restriction Level to be in ascending order when administered in the patterns or preferences through system management.

When a call is attempted, the system compares the Facility Restriction Level of the telephone with the Facility Restriction Level of the trunk routes available to complete the call. If the Facility Restriction Level of the telephone is equal to or higher than the Facility Restriction Level of trunks, the call is completed; if it is lower, the call is blocked on that preference and compared to the Facility Restriction Level of the next route available. If the call fails to match the Facility Restriction Level on the available preferences, the call may queue for the first available and compatible trunk group.

The system also provides a feature called Alternate Facility Restriction Levels that allows the attendant to temporarily change the Facility Restriction Levels on originating facilities to a different set of Facility Restriction Levels. It is used to grant users greater access to trunking facilities than is normally provided, such as when charges are lower during evening hours.

Bearer-Capability Class

Bearer-capability class uses information available in the system to match the calling requirements of a specific call with the best available resources to support that call. Bearer capability applies to all calls and support facilities, but is of primary significance for data calls. Each call has a bearer requirement — that is, a set or range of requirements needed to support that call. For data calls, these requirements include data rate, synchronization, and channel type.

Authorization Codes

Authorization codes are used on certain calls to temporarily raise a telephone's Facility Restriction Level. This is useful for those who make calls from telephones other than their own or from outside the network. If a call you dial is blocked because the telephone's Facility Restriction Level is too low, you can enter your authorization code. If the Facility Restriction Level associated with the authorization code is equal to or higher than the Facility Restriction Level of the trunk facilities required to place the call, the call is then completed. Up to 5000 (csi/si) or 90000 (r) different authorization codes can be provided for your system at any one time. Authorization codes can be from 4 to 13 digits long. Through the use of the system's management tools, you can assign authorization codes and change their associated Facility Restriction Level and network access permissions.

Network interfaces and equipment

The system supports a variety of interfaces to voice and data networks. Trunks supply links between the system, the public network, and other systems. Digital Signal Level 1 interfaces offer high-speed digital connectivity between systems. For a complete listing of the trunk and line interfaces available in different countries, see the *DEFINITY® ECS System Description*.

Trunk group circuits

Trunks provide the communications links between systems, including central office switches and other premises switches. Trunks that perform the same function are grouped together and administered as trunk groups. Trunks interface with the system via port circuit packs. Trunk group circuit types include the following:

- Local exchange trunks
- Tie trunks
- Auxiliary trunks
- Digital trunks.

Local exchange trunks

Local exchange trunks connect the system to a central office. The following are some of the types available:

- Central office trunks which connect the system to the local central office for incoming and outgoing calls
- Foreign exchange trunks which connect the system to a central office other than the local one
- Wide Area Telecommunications Service trunks which allow you to place long-distance outgoing voice-grade calls to telephones in defined service areas; these are priced according to distance in the service area, length of the call, time of day, and the day of the week
- Toll-free service trunks (such as 800 and 888) which let your business pay the charges for inbound long-distance calls so that callers can reach you
- Direct Inward Dialing trunks which connect the system to the local central office for incoming calls dialed directly to stations without attendant assistance
- Digital Service 1 trunks which can be used to provide T1 or ISDN Primary Rate Interface service.

Tie trunks

Tie trunks carry communications between the systems in a private network. Several types of trunks can be used, depending on the type of private network you establish. Tie trunks use a variety of signaling types such as ear and mouth (E&M), A-law companding, Mu-law companding, Type 1, and Type 5.

Auxiliary trunks

Auxiliary trunks connect devices in auxiliary cabinets with the system. Some of the features that are supported with this type of trunk are recorded announcements, telephone dictation service, malicious call trace, and loudspeaker paging.

Digital trunks

The system supports both E1 and Digital Signal Level 1 facilities. As industry standards around the world, E1 and Digital Signal Level 1 provide the latest alternative to analog trunking.

E1 interface

The system also supports E1 connections. T1/E1 access and conversion allows simultaneous connection to both T1 (1.544 Mbps) and E1 (2.048 Mbps) facilities (using separate circuit packs).

T1 interfaces

When planning your networking requirements, one of the options you should consider is multiplexing over Digital Services 1 (DS1) facilities. As the industry standard for interconnecting digital systems, DS1 is an economical alternative to analog trunking arrangements. Multiplexing up to 24 digitized voice/data communications paths onto a single T1 carrier or other high-speed digital facility (such as fibre or microwave) can reduce your network trunking and equipment costs.

Used to connect systems to the public network or to other systems in a private network, Digital Signal Level 1 also delivers high-speed, end-to-end digital connectivity. Voice and data calls are completed at transmission speeds of up to 64 kbps.

The system offers several options in supporting the Digital Signal Level 1 interface. The options include support for voice-grade Digital Signal Level 1 and alternate voice/data. The voice-grade Digital Signal Level 1 interface is a T1 D4 channel-bank-compatible interface that does the following:

- Uses in-band bit-robbed signaling to provide 24 voice-grade-only tie trunks consisting of 56 kbps channels for voice and voice-grade data transmission
- Interconnects the system with other systems with an external D4 channel bank or with other systems (analog or digital) having the appropriate interfaces

- Interconnects the system with central offices such as AT&T's 4ESS switch (where services such as MEGACOM and Software Defined Network can be accessed) and 5ESS®-2000 switches
- Interconnects the system with private networks by connection with DS1 facilities
- Can be used with the same Automatic Alternate Routing capabilities as normal analog E&M lead tie trunks.

Configuring your system with an alternate voice/data DS1 interface does the following:

- Uses out-of-band signaling in which signaling information is multiplexed onto one of the 64 kbps digital channels
- Permits end-to-end voice and digital data connections between systems
- Delivers 23 clear 64 kbps digital channels plus one signaling channel multiplexed onto a 1.544 Mbps Digital Signal Level 1 line with provisions for framing, maintenance, and signaling
- Delivers 8 kbps timing and slip information for a synchronization subsystem
- Supports ground-start and loop-start switch-central office, foreign exchange, and Wide Area Telecommunications Service (inbound/outbound) trunks, as well as direct inward dial trunks, off-premises stations, and dedicated voice/data system connections.

To achieve even greater benefits than those just listed, you can combine the DS1 interfaces and ISDN-PRI to give you additional capabilities. ISDN-PRI is a DS1-compatible direct-connect access service that links the intelligence inherent in the network with the intelligence provided by your system.

For example, with ISDN-PRI, the Software-Defined Data Network service may be accessed. Software Defined Data Network provides virtual private-line connectivity, via a switched network, for voice, data, and video applications. Software-Defined Data Network services compliment the Software-Defined Network voice services.

The system delivers Automatic Restoration capability with Software-Defined Data Network, which restores disrupted connections between access end points (non-signaling trunk) and data end points (devices that connect the system to computers and data communications equipment). This restoration is achieved within seconds of a service disruption so that critical data applications can remain operational.

ISDN

The system provides a complete set of ISDN features. Demonstrating its role as a leader in making ISDN a universal reality, Lucent Technologies makes it possible for anyone connected to the system to benefit from ISDN capabilities and features.

ISDN eliminates the need for multiple, separate access arrangements for voice, data, facsimile, and video services and networks. Using the same pair of wires that now carry simple telephone calls, ISDN can deliver voice, data, and video services in digital format.

ISDN is a global access standard established by the Consultative Committee for International Telephone and Telegraph designed to help you move and manage information with unprecedented ease and productivity — anywhere in the world. ISDN uses a layered protocol that conforms to layers one, two, and three (physical, link, and network layers) of the seven-layer Open Systems Interconnect Reference Model of the International Standards Organization.

The system supports the two major interfaces specified in the ISDN standards — PRI and BRI.

- PRI is used for connecting premises equipment to the network, and acts as a powerful interface between intelligent equipment such as systems and computers.
- BRI is used for connecting telephones, computers, personal computers, and other desktop devices to other computer equipment.

The system also supports an optional adjunct that converts ISDN-PRI lines to a trunk-side ISDN-BRI. A single PRI is converted to up to eight BRIs plus a proprietary 2 Mbps expansion interface. See your Lucent Technologies representative for more information about this adjunct.

Both PRI and BRI are based on the same common building blocks — the use of a common interface to a transmission path that is divided into channels. Both PRI and BRI use two types of channels for communication:

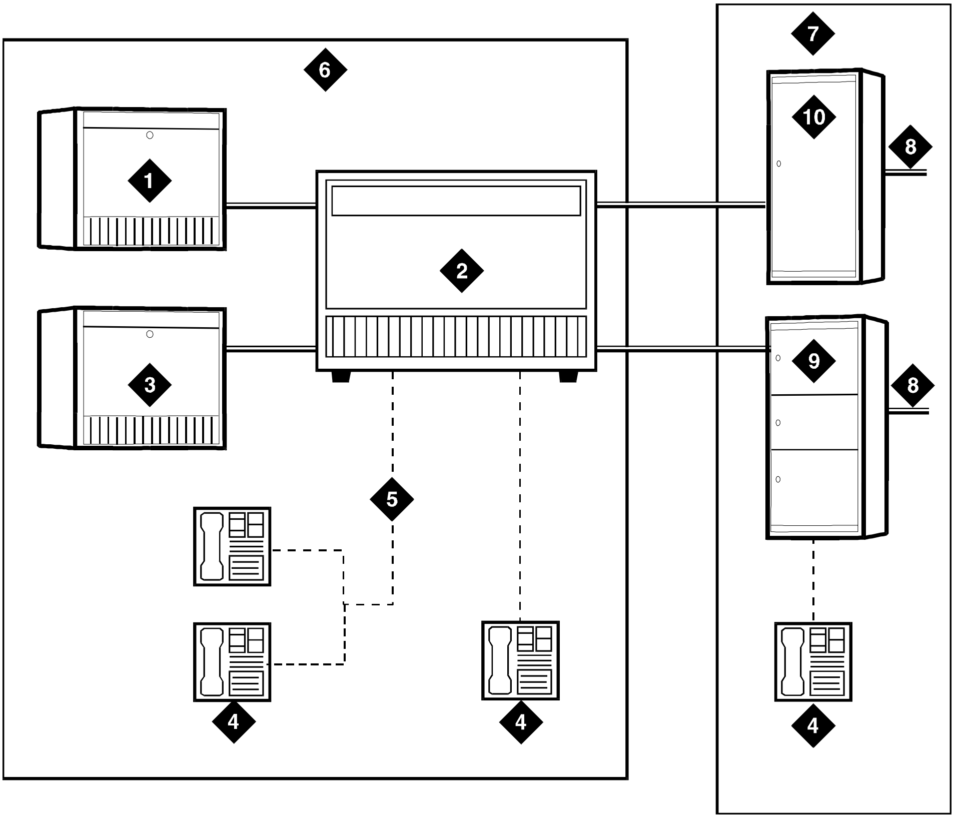
- Bearer channels are the communications links in ISDN. They provide 64 kbps digital communications service for voice, data, video, and other information transmission.
- Delta channels, sometimes known as data channels, are the signaling links in ISDN. They carry call-control and call-related information, such as caller ID, between ISDN endpoints.

PRI, referred to as 23B + D or 30B + D on an E1 interface, uses 23 or 30 64 kbps B channels and one 64 kbps D channel. The 23 or 30 B channels can be used for 23 or 30 individual voice or data calls. BRI, referred to as 2B + D, uses two 64 kbps B channels and one 16 kbps D channel. The B channels give the user simultaneous voice and data transmission over the same connection. This channel architecture allows full and complete use of the 64 kbps B channels from end point to end point for information movement managed by signaling messages, called Q.931 messages, in the D channel.

To help your business achieve maximum benefits from ISDN and the public network, the following features reside within the ISDN service nodes:

- Call-by-Call Service Selection lets you reach multiple services via the same ISDN B channel. Therefore, a channel can be allocated among MEGACOM Wide Area Telecommunications Service, MEGACOM 800/888 Service, and other services on a dynamic basis, eliminating the need for dedicating each truck or channel to a specific service.
- Automatic Number Identification, marketed as Information Forwarding-2 (INFO-2), is available on MEGACOM 800/888 Service. INFO-2 delivers the originating calling party's billing number to your system.
- Station Identification Number, similar to INFO-2, identifies the calling party number behind the system. Station Identification delivers the originating caller's telephone number to the network where it is sent to the terminating location.
- User-to-User Information sends user information from one endpoint to another using the D channel. Three forms are available: message associated data, sent within Q.931 call control messages during call establishment and call clearing; call-associated data, sent during call setup on a B channel; and noncall-associated data, sent with no related call-setup activity on the B channel. Applications for this feature include display of calling party name and number.

The system's support of ISDN-PRI, ISDN-BRI, and available public network services means that you can achieve full end-to-end ISDN connectivity and take advantage of ISDN services and features. For example, two systems connected by PRI can exchange calling party name and/or number information. The information is displayed on the called party's telephone. In addition, the called party's ID is also displayed at the calling party's telephone. This lets users identify the source of an incoming call before answering. Computer telephone integration interfaces can also use the information provided by the network to integrate your communications and data-processing systems.



1)	DEFINITY BCS/GuestWorks	6)	Private ISDN
2)	DEFINITY BCS/GuestWorks	7)	Public ISDN
3)	DEFINITY BCS/GuestWorks	8)	Public and Private Networks
4)	Basic Rate Interface Telephone	9)	Central Office Switch
5)	Passive Bus	10)	Tandem Switch

Figure 9-1. DEFINITY BCS/GuestWorks and ISDN

The system also adds the following capabilities to the basic ISDN services, depending on local availability of support.

- Call-by-Call Service Selection, in addition to the services provided by this feature on the network, allows each trunk in a PRI link from your system to the local central office to be designated on a per-call basis as Direct Inward Dial, incoming Wide Area Telecommunications Service, outgoing Wide Area Telecommunications Service, and so forth. This eliminates the need for dedicating each trunk or channel to a specific service, although they can still be dedicated, if desired.
- ISDN flow control monitors message activity on the Primary Rate Interface D channel.
- Non-Facility-Associated Signaling allows a PRI D channel to supply signaling for B channels (voice and data) located on PRI interfaces other than the one where the D channel is found. As a result, one D channel can support call control and signaling for up to 20 Primary Rate Interfaces.
- D Channel backup, when administered, improves reliability in the event of a signaling link failure on a Non-Facility-Associated Signaling D channel group. A primary D channel provides signaling for the Non-Facility-Associated Signaling D channel group (two or more Primary Rate Interface facilities). A second D channel, located on a separate Primary Rate Interface facility of the same Non-Facility-Associated Signaling D Channel group, is designated as a backup. If the primary D channel fails, call-control signaling automatically transfers to the backup D channel.

By combining public network services and ISDN features with the system's ISDN and other features, you can differentiate your business from your competitors, both in improved customer satisfaction and in greater operating efficiency. The result is improved profits and reduced costs. Here is a brief glance at a few of the possible ISDN applications:

- Dealer locator
- Product sourcing and fulfillment
- Consumer-to-business and business-to-business data retrieval
- Logging for callback.

IP Trunks

IP trunks allow you to route voice and fax calls over Internet Protocol (IP) networks such as the Internet and private intranets, reducing long-distance charges and giving you added flexibility in routing traffic between sites. Both the originating and destination switches must have the DEFINITY Internet Protocol Trunk (DEFINITY IP Trunk) application or Lucent's Internet Telephony Server-Enterprise (ITS-E) Release 1.2. The DEFINITY IP Trunk feature consists of the following components:

- An IP Trunk circuit pack, which contains a Windows NT server
- The DEFINITY IP Trunk software, which routes telephone calls and faxes over the Internet or your company's intranet
- Configuration Manager software, which lets you administer the operation and performance of DEFINITY IP Trunk service.

Both the IP trunk software and Configuration Manager reside on the Windows NT server on the IP Trunk circuit pack. For information about Internet Telephony Server-Enterprise, contact your Lucent representative.

NOTE:
DEFINITY BCS and GuestWorks do not support the full IP Solutions feature of the DEFINITY ECS, only IP trunks.

Electronic Tandem Network

If your company requires a medium to large network spanning a large geographic area, nationwide or even worldwide, Electronic Tandem Network is the answer. An Electronic Tandem Network is a wide-area private network that tandems calls through one or more systems to route the calls to their destinations.

An Electronic Tandem Network consists of tandem systems, inter-tandem tie trunks that interconnect them, access or bypass trunks from tandem systems to main systems, and the software and equipment to support call routing over the trunking facilities. Different Electronic Tandem Network locations are connected via analog or digital tie trunks. For example, a DS1 interface can act as a high-speed (1.544 Mbps) digital backbone for voice and data communications between Electronic Tandem Network locations.

An Electronic Tandem Network can be configured hierarchically. An Electronic Tandem Network can connect individual systems; it can also connect other private networks together.

Within an Electronic Tandem Network, each location is identified by a unique private network location code, similar to the public network office codes that exist within an area code. When accessing the Electronic Tandem Network, a user simply dials the network office code plus the desired extension number, for a total of seven digits.

In an Electronic Tandem Network, the system provides a variety of features on a network-wide basis. Here are a few examples:

- Uniform Dial Plan — A unique four- or five-digit number assigned to each station on the network. Uniform numbering gives each station a unique number (location code plus extension) that can be used at any location in the Electronic Tandem Network. To access that station, the system enhances the standard uniform dial plan with the unrestricted five-digit uniform dial plan, which allows up to five digits to be parsed for call routing.
- Automatic Alternate Conditional Routing — A feature used to control the routing of particular calls using conditional routing. For example, you can limit the number of communications satellite hops (communications satellite links used as trunks) in any end-to-end private network routing pattern. Limiting the number of satellite hops may be desirable for controlling transmission quality or call delay in both voice and data calls.
- Automatic Transmission Measurement System — A feature used to perform routine and on-demand maintenance tests on facilities in the Electronic Tandem Network.
- Enhanced Trunk Signaling and Error Recovery — A feature that improves the reliability of Electronic Tandem Network calls by allowing a trunk call to be retried on another circuit when signaling failures occur.

Voice Messaging Solutions

10

Overview

Less than 30 percent of person-to-person business calls reach the intended party on the first attempt. Integration with Lucent Technologies voice messaging products can help ensure that important calls are not lost.

For nearly a decade, the Lucent Technologies voice messaging systems have provided businesses with the voice processing tools to communicate more efficiently and to make time spent on the job more productive. Whether companies have ten employees or hundreds, the dilemma of how to do more with less is driving them toward innovative multimedia processing solutions.

Within an organization, voice messaging is much more than just an answering machine. It bypasses idle chatter to promote a communications mode that can be much more efficient than two-way calling. Lucent Technologies studies show that voice messages average 30 seconds, whereas 2-way calls run much longer and are devoted to business only 50 percent of the time.

The Lucent Technologies voice messaging systems available with DEFINITY BCS and GuestWorks include the following:

- DEFINITY AUDIX (for non-hospitality offers)
- Octel 100 Messaging (formerly Messaging 2000)
- INTUITY AUDIX
- INTUITY Lodging (for GuestWorks hospitality offers).

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

DEFINITY AUDIX messaging system

While many voice messaging systems require separate equipment and connections, the DEFINITY AUDIX Release 4.0 system easily installs directly into a DEFINITY BCS cabinet to support advanced multimedia voice messaging capabilities without the need for an adjunct processor.

The DEFINITY AUDIX system gives small- to medium-sized businesses full voice messaging performance in a streamlined, cost-effective package. The result is high-performance voice messaging no matter what your business size.

Each DEFINITY AUDIX system supports up to 2000 mailboxes and stores up to a maximum of 100 hours of recorded messages using a maximum of 12 ports (in two-port increments). With each DEFINITY AUDIX package, you also receive a complete set of end-user and product-support documentation.

The system includes such features as multiple personal greetings, full-functioned automated attendants, outcalling for message notification, and multiple language support. The DEFINITY AUDIX system includes both analog (Audio Messaging Interchange Specification [AMIS]) and proprietary digital networking software, which allows the system to exchange voice messages, subscriber profiles, and message status information with other voice messaging systems.

By embedding the voice messaging hardware within the switch, DEFINITY AUDIX provides the following advantages:

- Because it is integrated within the switch, separate review and approval by government agencies for compliance with electrical requirements and other technical specifications often are not required.
- Connecting to the backplane provides direct access to interfaces such as time slots, signaling mechanisms, and power feeds. DEFINITY AUDIX uses either display set (DS) integration or X.25 integration. TCP/IP integration is not available.
- Bypassing analog ports and digital conversions provides a more efficient, higher quality call storage process.
- You can use the system’s maintenance strategy with DEFINITY AUDIX to allow remote maintenance by the same team that maintains the switch.

The entire DEFINITY AUDIX system is contained on one circuit pack (two slot positions are required in most cases; only one slot position is used if the circuit pack is installed in slot 6 of a CMC cabinet). The components mounted on the circuit pack include: the central processing unit, the hard disk for real-time storage, a magneto-optical disk drive for software updating and backup, the digital signal processor complexes that do speech processing, and the time slot interfaces for connecting the system to the DEFINITY switch.

Reliability and security

The TN568 alarm circuit pack on the DEFINITY AUDIX system has its own processor that allows maintenance and diagnostic access if the main processor fails. A series of LEDs on the faceplate lets on-site technicians check system status. There also is a robust set of built-in diagnostics that technicians can access either on-site or remotely through a separate modem. A special alarm-originating feature helps speed problem diagnosing and correction.

The system routinely performs self diagnostics. If it detects a problem, the system automatically dials a remote maintenance center and produces a detailed alarm message with diagnostic specifics. The remote maintenance center staff responds quickly via the remote access port and external modem to perform further diagnostics, isolates the problem, and takes corrective action. As a back-up, the DEFINITY AUDIX system can send an alarm message to the switch.

Easy installation and expansion

Lucent Technologies specifically designed the DEFINITY AUDIX system for easy installation. There are no special power or cabling adjustments required.

System expansion is simple. All the hardware required for the full 12 ports is included in the initial DEFINITY AUDIX system. If you start with fewer than 12 ports initially, Lucent Technologies can then activate additional ports through a simple software change that technicians download remotely. There is no need to modify the basic hardware.

Improved clarity

A speech processing algorithm developed at Bell Laboratories encodes at 16 kbps, giving the DEFINITY AUDIX system a major advantage over its competitors in that it can store more messages in a smaller space.

The algorithm also improves the speech quality; it provides superior voice quality in system prompts, users' personalized greetings, and the voice messages themselves. Bell Laboratories' listening studies show that the message playback clarity is unsurpassed in the voice messaging industry.

Enhanced speed-up/slow-down of message playback is now offered because of this new algorithm with no loss of clarity. DEFINITY AUDIX system users can play back messages twice as fast or at half speed with no distortion in pitch.

The best solution worldwide

Lucent Technologies offers the DEFINITY AUDIX system in the same countries as the DEFINITY BCS. Prompts are available in several languages. (Contact your local representative for information on available languages.) Lucent Technologies will continue to develop a wide variety of languages and bilingual capabilities. Contact your account representative for the latest options.

Summary of DEFINITY AUDIX features

DEFINITY AUDIX is a powerful voice mail system that enables you to create, store, send, and receive spoken messages electronically. Spoken prompts guide you as you enter simple one- or two-key commands at a touch-tone telephone. Subscribers can use the system 24 hours a day, sending and retrieving messages from any touch-tone telephone. The DEFINITY AUDIX system also helps to protect sensitive information by requiring users to enter a combination of subscriber login codes and passwords before granting access to the system.

Whenever you call the DEFINITY AUDIX system, you interact with it by entering commands through your telephone's touch-tone keypad. You simply specify the desired activity, and follow the voice prompts for the desired task.

Special multimedia-processing features include Voice Mail, Call Answering, Out-calling, Multilevel Automated Attendant, and Bulletin Board. The following is a summary of DEFINITY AUDIX capabilities:

- *Shared Extensions* provide personal mailboxes for persons sharing a phone.
- *Multiple Personal Greetings* allow subscribers to prepare a pool of up to nine personal greetings to save time and provide more personal customer service. Separate messages can indicate the subscriber is on the phone, away from the desk, on vacation, etc. Different messages also can apply to internal, external, or after-hours calls.
- *Message Manager* is an advanced desktop application that runs on a Windows personal computer, providing powerful and intuitive access to DEFINITY AUDIX messaging features through a customer-provided TCP/IP Local Area Network. The application's graphical user interface allows easy access to voice mail on a DEFINITY AUDIX system. The Message Manager interface is often faster and more efficient than accessing messages through the telephone. Message Manager is an optional feature that must be purchased separately. For more information, see Page 10-6.

NOTE:
The DEFINITY AUDIX system does not support fax messages.

- *Priority Messaging* places important messages ahead of others.
- *Outcalling* automatically dials a prearranged phone number or pager when messages are received in a subscriber's mailbox.
- *Priority Outcalling* provides outcalling notification of priority messages only.
- *Broadcasting* allows the same message to be sent to multiple recipients or to all subscribers on the system.
- *System Broadcast* capabilities are available in two forms: Broadcast Voice Mail and Login Announcement.
- *AUDIX Directory* allows subscribers to "look-up" the extension number of any other subscriber by simply entering that other user's name on the telephone keypad.
- *Personal Directory* shortens the time required to locate correct names by accessing a user-customized list in the Names Directory.
- *Call Answering for Nonresident Subscribers* provides DEFINITY AUDIX system mailboxes for subscribers who do not have an extension number on the system.
- *Full Mailbox Answer Mode* informs callers whenever messages cannot be left because there is no more room in a subscriber's mailbox.
- *Name Record by Subscriber* lets subscribers record their own names on the system.
- *Automatic Message Scan* plays all new messages in part or in their entirety without requiring the subscriber to press additional buttons — a feature particularly beneficial to users of car phones.
- *Sending Restrictions by Community* provides the capability to limit the communities of callers who can communicate via DEFINITY AUDIX Voice Messaging.
- *Group Lists* allows subscribers to create mailing lists of up to 250 people to use for broadcasting messages.
- *Message Forwarding* lets subscribers forward messages they have received, with or without attached comments.
- *Name Addressing* allows subscribers to enter the name of a message recipient if the recipient's extension number is not known.
- *Private Messaging* is a special coding feature that prevents recipients from forwarding sensitive messages to others.
- *Leave Word Calling* allows subscribers to simply press a button on their telephones to leave a standard *call me* message on any extension.
- *On-Line Help* provides subscribers with instant access to voiced instructions at any point in the process.

INTUITY AUDIX voice messaging

The INTUITY AUDIX system allows you to record, distribute, and receive messages in various mediums. INTUITY AUDIX is the product-of-choice with the GuestWorks offer. The system runs on a dedicated computer connected to the switch and allows the transfer of voice and fax communications to and from the switch via analog voice ports and also allows data communications to and from the switch via a data link.

The INTUITY AUDIX system offers everything you get with DEFINITY AUDIX plus Fax Messaging and enhanced Message Manager features.

Fax Messaging

INTUITY Fax Messaging works with the INTUITY Lodging application to allow subscribers to use their INTUITY Lodging mailboxes for fax messaging. With INTUITY Fax Messaging, subscribers may receive, create, send, and forward fax messages. Subscribers may also use the INTUITY Fax Messaging application to create special mailboxes for each of your fax machines. These mailboxes (known as guaranteed mailboxes) accept fax telephone calls when the fax machine is busy and then deliver the fax to the fax machine when the fax machine is available.

Message Manager

The INTUITY Message Manager provides access to INTUITY AUDIX voice processing features on a personal computer connected to a local area network (LAN). It also works with DEFINITY AUDIX. This feature requires three components:

- The AUDIX server software may be purchased with the INTUITY AUDIX System with an INTUITY Message Manager Right-to-Use. Also, this feature has INTUITY AUDIX hardware requirements.
- The Message Manager software diskettes can be purchased separately and are installed either on each user’s PC or on a LAN server.
- The local area network is owned and maintained by the customer and must meet certain requirements for the INTUITY Message Manager feature to work.

Message processing features available at a subscriber’s PC with INTUITY Message Manager include the following:

- Looking at up to 16 message headers at a time and listening to messages in the order you choose. For subscribers who get many messages, this provides an easy way to view and prioritize the messages.
- Ability to send and receive fax-only or voice-fax messages, to view faxes on your PC, and optionally to print faxes.

- Recording, addressing, and scheduling messages.
- Replying to messages and forwarding messages.
- Annotating messages with a short subject line.
- Setting up mailing lists on-line with easy text entry and editing. You can see the lists on-line and print lists on any local or network printer.
- Setting up personal greetings, multiple personal greetings, or multilingual greetings on-line makes it easier for you to manage and maintain your greetings, and annotating your greetings helps jog your memory.
- Browsing the subscriber directory.
- Administering Outcalling notification on-line with easy text entry and editing.
- Storing (archiving) voice messages on your PC for a permanent record of voice mail when needed.

NOTE:
Message Manager does not operate with guest accounts on INTUITY Lodging. It only operates with subscribers of INTUITY AUDIX.

Voice Director

Voice Director allows an INTUITY AUDIX subscriber to call, send a message, get a call transferred, or forward a message by saying the name of that person instead of keying in that person's extension number.

INTUITY Lodging

INTUITY Lodging is a separate application from INTUITY AUDIX and is used to support voice messaging for guest mailboxes. INTUITY Lodging was designed specifically for the hospitality industry. The system is described in Chapter 7, ''Hospitality Solutions.''

Voice messaging systems and call coverage

The DEFINITY AUDIX and INTUITY AUDIX systems can be set up as the last points on a coverage path. Calls are then redirected to AUDIX if they are not answered by a previous station on the path. In addition, a secretary or messaging agent who answers a call can transfer a caller to the AUDIX system "mailbox" of the original called party upon request. The caller may prefer to leave a voice mail message if the message is personal, lengthy, or highly technical.

Many other options are available for maximum flexibility:

- A caller can transfer from the system to an attendant or operator.
- A caller can transfer to another extension instead of leaving a message.
- Your company can have an automated attendant answer calls to the company and direct calls to the correct department quickly, so callers do not have to wait on hold.
- With automated attendant, callers can be instructed to enter keypad commands to direct the call to the appropriate point. This gives customers choice and control. It also allows you to make the most effective use of your personnel, while still providing your customers with the service they expect.

Mode Code interface

The system supports an analog Mode Code interface for communications with INTUITY AUDIX and Octel 100 Messaging (Messaging 2000). This interface employs DTMF tones, line signals, and feature access codes, and allows INTUITY AUDIX to exchange data with the system without using a TCP/IP or X.25 data link.

Centralized Voice Mail via Interswitch Mode Codes

Provides the capability to share a voice mail system among several switches using the Mode Code - Voice Mail System Interface. This feature provides a cost effective choice for multiple sites by eliminating the need for a voice mail system at each site.

A Centralized Voice Mail network can consist of DEFINITY BCS Issue 6 or later, DEFINITY ECS R8 or later, ProLogix™ R3 or later, Merlin Legend® R6.1 or later, and Merlin Magix™ systems. Either a DEFINITY BCS, ProLogix, or DEFINITY ECS switch must be the host for the voice mail system, and UDP and ISDN-PRI software are required at each site. See *DEFINITY ECS Administration for Network Connectivity* and *Merlin Legend Network Reference* for more information.

Octel 100 Messaging

Octel 100 Messaging (formerly Messaging 2000) is a highly integrated multimedia voice and fax messaging system. It works and interfaces with DEFINITY BCS via one-to-one analog VMI-program station ports.

Octel 100 Messaging is price-sensitive and desirable for customers familiar with the Octel interface, Octel networking, and customers that have multiple satellite locations that require less than 12 ports initially. Octel 100 Messaging has specific features not found on INTUITY AUDIX (for example, cascaded outcalling, visual architect, call screening, call routing).

Connectivity for Octel 100 Messaging is as follows:

- DEFINITY BCS switch
- Maximum of 16 ports (12 initially)
- Each port (voice or fax) requires one corresponding analog station port assigned as a VMI port on the switch.
- Visual Mailbox requires NOVELL® 3.21 or higher, or Windows NT 3.51 or higher.

Video Solutions

11

Overview

Consider the following business scenario. Your colleagues have asked to meet with you and your team as soon as possible to discuss the latest project. The project team needs to discuss how to meet the timetable and satisfy your customer's expectations. The team also needs to view the prototype that the engineering department has just finished.

Add to this that you are already traveling on business for another project, the prototype is back at the home office, and your team is spread across different sites. Standard business procedure would dictate that you cancel your meetings, get on the next airplane back to one work site or another — briefcase full of needed files, and the prototype packed carefully for shipment. Hopefully, the prototype will arrive intact, and the time away from your current assignment will not set you too far back in your already hectic schedule.

DEFINITY BCS and GuestWorks provides an alternative to business as usual — revolutionary video communication solutions. Through the use of Lucent Technologies' video products and services, you can meet with your colleagues — across the country or around the world — via video communications. So, instead of heading back to the airport, your associates at the home office take the prototype to their video conferencing room, you set a up a video conference with facilities offered by your hotel, and everyone is happy knowing this face-to-face meeting does not involve luggage or jet lag. Then once your meeting is over, you are back at your hotel room taking care of your current project assignment, and your team at the home office is ready to start implementing the decisions that were just made.

Video conferencing allows you to make quicker decisions, provides ready access to essential information, allows you to consult with specialists on an as-needed basis, and ultimately allows you to bring products to market faster.

Visual communications provides other advantages for your normal day-to-day operations. Business meetings can benefit from the nuances that a facial expression can convey sometimes more directly than the words being spoken. Employees can be trained on the latest products and procedures on a regular basis. You can meet with your suppliers without ever travelling to see them.

This section will introduce you to the visual communication products that you can connect to your system to create a premier communications solution that satisfies all your needs — voice, data, and video — just by dialing a telephone number.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

Group Video System

The Group Video System turns a telephone call into a face-to-face meeting for conducting business with people across the country or around the world. The lines of Group Video Systems, based on the PictureTel and Polycom Group Video Lines, are designed to assure that your video conferences are the most effective possible.

A Group Video System can connect directly to your system or to the network. When connected to your system via either Digital Communication Protocol ports or a DS1 interface, video calls are placed as easily as voice calls. You can also benefit by using your system's Automatic Route Selection (formerly World Class Routing) capabilities and the shared use of network facilities such as ACCUNET Switched Digital Services or Software-Defined Digital Network.

Group Video Systems are totally self-contained and include a pan/tilt/zoom camera, a monitor, the control unit, communications equipment, and the equipment electronics. You can easily add peripheral equipment such as video cassette recorders to record the conference, document cameras or scanners to review hard-copy material with distant participants, and personal computers to supply spreadsheets or other computer-based conveniences.

You can equip any office or conference room with a Group Video System. Conferees can speak and act naturally without thinking about audio and video pickup. The system adapts automatically to room acoustics, and a unique, audio-compression algorithm yields distortion-free, full-duplex, interactive video without echo. A highly-sensitive microphone is built into the control unit, and if necessary, conferees can use up to four auxiliary desktop and/or lapel microphones.

You control the conference via a desktop keypad that easily turns the system on or off, dials the call, adjusts the audio volume, selects the video source, and positions the camera. The camera in most models has an auto-focus lens, plus zoom, pan, and tilt capabilities that let users move the camera to follow conferees as they move around the room. Optional capabilities include remote control of the far-end camera and camera presets that let conferees set up to eight camera positions — four local and four remote — each accessible at the touch of a button.

An automatic feature of most Group Video System models is the use of a window (picture in a picture) for previewing, so you can see what your camera sees as well as what the far-end camera sees. The window also permits simultaneous viewing of far-end video and still-image graphics.

Group Video Systems are available in a variety of models that can accommodate an office, small meeting room, or even a spacious board room. All models have color monitors, with dual monitors available on most of them for simultaneous viewing of video and high-resolution still images. The systems available are as follows:

- PictureTel Venue
- PictureTel 4000ZX
- PictureTel Concorde 4500
- PictureTel 4200ZX
- Swiftsite 740/760/763
- Polycom 4000
- Polycom EX
- Polycom Viewstation 1287/512/MP/V.35/DCP (H.320 and H.323 video over IP)
- Polycom Showstation IP (H.320 and H.323 video over IP).

Telephone add-on is an option on all models. This enables you to add a voice conferee to the video conference. Other options include security encryption, freeze-frame graphics, and VCR recording — each providing additional benefits to your video conference.

The monitors can accommodate both the U.S. National Television System Committee (NTSC) standard and PAL, the European 625-line standard, providing global compatibility for your visual communication needs.

MultiPoint Conferencing Unit

When connecting more than two video endpoints, you can use the MultiPoint Conferencing Unit to set up and conduct multipoint video conferences. This is a stand-alone unit that provides easy-to-use multilocation video conferencing.

The MultiPoint Conferencing Unit can operate behind the system or can be directly connected to the network. The MultiPoint Conferencing Unit can support up to 96 ports. Those ports can then be used to connect multiple video end points, either Group Video System or Desktop Conferencing Systems (H.320 or H.323), in a multipoint conference. Group Video Systems can be linked at speeds from 56 kbps to full T-1.

The MultiPoint Conferencing Unit uses the ITU-T H.320 and H.323 video conferencing standard to connect the video endpoints, assuring compatibility with other video endpoints that conform to the standard. In addition to compatibility, the H.320 and H.323 standards ensure a common level of visual, graphics, and audio quality that will satisfy your visual communication requirements.

The MultiPoint Conferencing Unit is built on the DEFINITY architecture, and is available in SCC or MCC hardware configurations. The MultiPoint Conferencing Unit sits in its own carrier and takes up approximately the same space as a single-carrier cabinet. Designed for growth, the MultiPoint Conferencing Unit's architecture allows you to add additional circuit packs and carriers as needed.

With the MultiPoint Conferencing Unit, multipoint video conferences are easy to set up, operate, and manage. You can use the reservation software provided with the MultiPoint Conferencing Unit, available through the management terminal or through the optional Conference Reservation and Control System.

You can assign a number to each conference participant and set up the MultiPoint Conferencing Unit to link the video endpoints at the designated time. Calls can also be initiated through the Meet-Me function, allowing participants to dial into their call using a preassigned telephone number. The MultiPoint Conferencing Unit can also be programmed to dial-out to the video endpoints at a designated time.

The MultiPoint Conferencing Unit also supports dedicated multipoint conferencing. Your video conference users that require regular and frequent access to multipoint video conferences can be assured of system access as required.

The Universal Conference Control (UCC) feature allows the end-user to control a video conference from their desktop.

Hunt Group Solutions

12

Overview

DEFINITY hunt group applications are designed to efficiently connect each caller with an agent best suited to serve that caller. The system begins the process by capturing information about the caller even before the call is routed. That information is integrated with existing databases (see Chapter 6, "Computer-Telephone Integration Solutions"), and the combined data is used to assist the agent in call handling. Additional features politely keep callers who are waiting in queue (a holding place for incoming calls) informed about how long it will probably take to process the call. Detailed call statistics are constantly available to agents and supervisors.

Calls coming into your hunt groups are queued up and routed based on information that the system continually acquires. Each of your customers can be presented with a variety of options for leaving a voice message, leaving a fax, or monitoring the status of his or her call.

This section describes the hunt group capabilities:

- *Automatic Call Distribution*, which manages call traffic and work flow.
- *Call Vectoring*, which allows managers to create controlled routing scenarios that give each caller the best possible service at the least cost.
- *Call Prompting*, which allows you to handle incoming calls based on digits entered by the calling party.
- *Basic Call Management System*, which provides reports on the measured hunt groups, also know as “splits.”

The system provides an applications platform that consists of several elements. When these elements are integrated to meet your business requirements, you will have the advanced call distribution and management capabilities that will deliver the performance and growth necessary for your business success.

NOTE:
Some applications and products are unavailable in some countries. Please check with your local distributor for further information about which features and applications are available to you.

Automatic Call Distribution

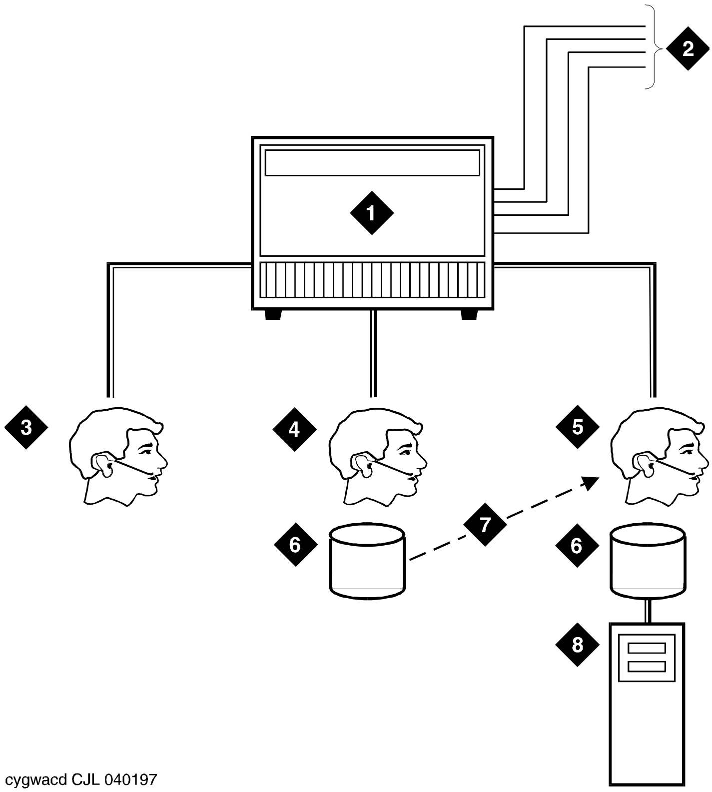
If your company has groups (such as reservations, sales, billing, or customer service) that handle incoming calls, you can benefit by using the system's automatic call distribution (ACD) capabilities. ACD is the basic building block for the hunt group applications.

ACD offers you a method for distributing incoming calls efficiently and equitably among available agents. In an ACD environment, agents are assigned to splits where all agents in that split typically handle the same types of calls. With ACD, incoming calls can be directed to the first idle, most idle, or next idle agent within the split to receive a call (circular hunting).

- With **first idle agent**, incoming calls always start with the first agent in the hunt group. If the first agent is busy, the call goes to the second agent. When the first agent becomes idle, the next call goes back to that first agent. This hunting scheme does not spread the calls evenly over all agents.
- With **most idle agent** distribution, an incoming call is routed to the agent who has been available for the longest time, resulting in balanced workloads for agents.
- With **circular hunting**, the system keeps track of the last extension in the split to which a call was connected, such that when the next incoming call arrives, the system can determine the next idle agent in the circular hunt group. Extensions in the hunt group that are busy are skipped, and the next idle extension within the hunt group is selected regardless of past call history. The caller could hear a busy tone if all extensions in the split are busy and no type of call coverage has been designated.

DEFINITY BCS and GuestWorks supports a maximum of 150 logged-in agents, with agents being part of up to four different splits. Each split has associated trunks, stations, recordings, and queues. You can link a telephone number to an ACD split by associating a published number (often an 800 or 888 toll-free number) with the split's extension number.

In Figure 12-1, Split A receives calls only when agents are available, since Split A has no queue. Calls to Split B can be queued while agents are unavailable, and redirected to Split C if not answered within an administered time. Calls to Split C are redirected to voice mail if they are not answered within an administered time.



1)	DEFINITY BCS/GuestWorks	5)	Split C: General Information
2)	Incoming Lines	6)	Queues
3)	Split A: Business Travel	7)	Call Coverage to Split C
4)	Split B: Personal Travel	8)	Voice Mail

Figure 12-1. An Example of Automatic Call Distribution

The system places all Automatic Call Distribution calls into a queue. Each call stays in the queue until an agent becomes available, until an optional timed interval expires, or until the caller hangs up. If the call has not been answered after an administrable period of time, an announcement can be played for queued callers. The call can then be connected to music to let the caller know that the call has not been dropped. The call can be sent to a coverage path, or it can be connected to another announcement.

You can set a maximum queue length in a group to anywhere from 0 to 200 calls (csi/si) or 0 to 999 (r), and you can establish a queue warning level. If the preset maximum queue length is reached, additional incoming calls are redirected to a call-coverage path (ensuring that calls are routed to an extension that will answer the call), or are given a busy signal. A priority-queuing feature allows you to designate which calls should receive priority; these calls override the standard first-in first-out queuing pattern.

Two features provide for redirection of ACD calls:

- Intraflow allows an ACD call to be redirected from one split to another through coverage paths that are assigned to determine call redirection criteria.
- Interflow allows new calls in a split's queue to overflow and be sent to another ACD split on another system using the Call Forwarding All Calls feature. Interflow can be useful during the evening, during peak operation times, or at other times when agents are unavailable.

 NOTE:
 This feature is not related to the Look-Ahead Interflow feature. The Look-Ahead Interflow feature is not supported on DEFINITY BCS nor GuestWorks.

ACD agents can use any model of telephone, but it is recommended that they use multiappearance telephones with an adequate number of feature buttons. A number of special ACD features can be assigned to their telephones to enable them to perform their jobs more effectively.

Additional features give your company even more options when using ACD:

- Queue-Status uses button lamps and telephone displays to indicate call status for calls waiting in an ACD queue. The status is available on telephones with a digital display. Queue-status can also display how long the oldest call has been waiting.
- Dialed-Number Identification Service allows agents to identify (via display telephones) the purpose of each incoming call and to greet the caller appropriately.
- Each agent can be logged in to as many as four splits at a time. However, an agent can be active on calls for only three splits at any one time.
- Malicious Call Trace allows you to designate stations that can trace emergency or threatening calls. When an agent receives a malicious call, the agent presses the Malicious Call Trace button. The system gathers trace information and connects a customer-provided voice recorder to the call (via an auxiliary trunk circuit). All equipment used to complete the call is held active (the call cannot be disconnected) until the feature is deactivated.
- Redirection on No Answer allows an unanswered, ringing call to be redirected to an ACD queue or to a vector directory number after an administered interval. The agent position will also be taken out of service.

Call Vectoring

Call Vectoring is a versatile method of routing incoming calls that can be combined with Automatic Call Distribution for maximum benefit and split efficiency. A call vector is a series of call-processing steps (such as providing ringing tones, busy tones, music, announcements, and queuing the call to an Automatic Call Distribution split) that define how calls are handled and routed. The steps, which contain vector commands, determine the type of processing that specific calls will receive.

Vector commands may direct calls to on-premises or off-premises destinations (*route-to* command), to any split (*goto* command), or to a specific call treatment such as an announcement, forced disconnect, forced busy, or music.

With combinations of different vector commands, incoming callers can be treated differently depending on the time or day of the call, the importance of the call, or other criteria. The system can route incoming callers to different vectors (10 for csi/si, 20 for r). Each vector can have up to 32 commands. The system also allows vectors to be linked via the “Go to Vector” command.

Vector Directory Numbers and Vectors

Calls access vectors using Vector Directory Numbers (VDNs). A Vector Directory Number is a “soft” extension number that is not assigned to a physical equipment location. A Vector Directory Number has several properties that are administered by the system manager and that include the extension number, Vector Directory Number name, class of restriction, display override, and the vector number associated with the Vector Directory Number.

Access to a Vector Directory Number may occur in many ways. Since a Vector Directory Number is an extension, it can be accessed in almost any way that an extension can be accessed.

Each Vector Directory Number maps to one vector. However, several Vector Directory Numbers may map to the same vector.

When answering a call, the agent will see the information (such as the name) associated with the Vector Directory Number on the telephone display and can respond to the call with knowledge of the dialed number. This operation provides Dialed-Number Identification Service, allowing the agent to identify the purpose of the incoming call.

Applications

There are many different applications for Call Vectoring. However, Call Vectoring is used primarily to handle the call activity of Automatic Call Distribution 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. Descriptions of other common applications follow.

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. Calls to different Vector Directory Numbers (and vectors) can queue to different priority 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 then disconnect the call. During business hours, the vector could queue calls to splits for connections with agents. All of this can be accomplished automatically without any intervention by the split supervisor.

Attendant Vectoring

With Attendant Vectoring, a highly flexible approach for managing incoming calls to an attendant is available. For example, with current night service operation, calls redirected from the attendant console to a night station can only ring at that station and will not follow any coverage path. With Attendant Vectoring, night service calls will follow the coverage path of the night station. The coverage path could go to another station, and then eventually to a voice mail system. The caller can then leave a message that can be retrieved and acted upon.

Off-loading of Periodic Excess Calls

A vector can check conditions in the targeted split, such as the number of calls already in queue. If the number is above a certain threshold, the vector bypasses that split and routes the call to another split, or the vector can return a busy signal. However, if the number is below the threshold, the vector queues 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. People with a common interest can be instructed to call a specific number (a Vector Directory Number) that connects to a specific announcement vector, which routes callers to a voice messaging system or to an integrated announcement circuit pack in the system.

Call Prompting

Call Prompting, an integrated subset of Call Vectoring, may be used in various applications to enhance call handling based on information collected from the calling party. Call Prompting uses Call Vectoring commands to route calls based on the information collected. It allows you to solicit and provide information to incoming callers who are in queue without causing them to lose their place in queue. The following describes four applications for Call Prompting:

- Automated attendant — Allows the calling party to enter the number of any extension on the system. The call is then routed to the extension. This allows you to reduce cost by reducing the need for live attendants.
- DIVA (data in/voice answer) — Allows the calling party to hear selected announcements based on the digits that he or she enters. This may be used for applications such as an audio bulletin board.
- Data collection — Allows the calling party to enter data that can then be used by a host computer application to assist in call handling. For example, this data may be the calling party’s account number, which could be used to support an inquiry/response application.
- Split messaging — Gives the calling party the option of leaving a message or waiting in queue for an agent. This may be used for an on-line order entry system or to further automate an incoming-split operation.

Basic Call Management System

The Basic Call Management System (BCMS), an integrated, internal capability, is a cost-effective solution that a small business can use to monitor the effectiveness of its call receipt groups. BCMS helps you fine tune your operation by providing reports with the data necessary to measure the performance of your agents. BCMS is ideal for companies that need call management features but do not require the same capacities available with the larger DEFINITY ECS Call Center applications. BCMS collects up to seven days of call data.

The switch supports a maximum of 150 logged-in ACD agents. Of those 150 agents, a maximum of 25 agents can be measured by BCMS. However, measurements are collected on a per-hunt-group basis, not a per-agent basis, and up to five hunt groups can be designated for measurement by BCMS. This means that whether you designate one hunt group or five hunt groups as being internally-measured by BCMS, no more than 25 agents can log in to those hunt groups at any one time. The other 125 agents can log in to other hunt groups as long as they are not designated for measurement by BCMS.

BCMS provides various measurements for monitoring the operations of an ACD application. The software organizes ACD calls and split measurements into functionally different reports that supply information useful for managing ACD facilities and personnel. The reports can be displayed on the system administration terminal, printed while viewing the report, or scheduled for printing at a later time via the Report Scheduler feature.

The following are the types of reports that can be generated:

- Real-time reports
  - Split Status
  - System Status
  - Vector Directory Number Status.
- Historical reports
  - Agent
  - Agent Summary
  - Split
  - Split Summary
  - Trunk Group
  - Trunk Group Summary
  - Vector Directory Number
  - Vector Directory Number Summary.

DEFINITY Extender

The DEFINITY Extender allows your agents to work from home. With DEFINITY Extender, agents can use display telephones from home and work exactly as they would in an office. See Chapter 13, "Telecommuting Solutions," for more information about the DEFINITY Extender.

Telecommuting Solutions

13

Overview

Lucent Technologies research, supported by industry studies, shows that telecommuters are generally 15 to 30 percent more productive when they work at home. They convert travel time into productive work time, are less likely to be distracted by normal office routines, and frequently end up working longer hours with greater output. During severe weather, they can continue working when others cannot.

Special system modules are available for telecommuting. In addition, many standard system and voice messaging features work well for telecommuters.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

Coverage of Calls Redirected Off-Net

Coverage of Calls Redirected Off-Net (CCRON) allows calls that have been redirected to locations outside of the switch to return to the switch for further processing. For example, an employee that telecommutes can have two coverage paths. One coverage path is used when in the office and the other coverage path is used when working from home. The coverage path used from home would have a call to the employee’s work phone cover to his or her home phone. If the employee does not answer the call or is busy on another call, the call is redirected back to the switch for further processing, such as coverage to voice mail.

DEFINITY Extender

DEFINITY Extender allows you to use a fully functional DCP telephone at a remote location. The telephone looks and performs exactly as if it were directly connected to your office system.

The system uses a module at the switch and a module at the remote location to provide full service. For residential applications, the Extender supports the 603E DCP telephone. For business applications, the Extender supports the 6408D+, 6416D+, 6416D+M, 6424D+, 6424D+M, TransTalk 9031, and 603E DCP telephones. Since these DCP phones have displays, the system works well for agents working from home or in a branch office. A dial-in number and password makes the system reasonably secure from unauthorized use.

Lucent Technologies Telecommuter Module

Lucent Technologies Telecommuter Module is a lower-end telecommuting solution that is ideal for telecommuters who are not necessarily hunt group agents. Incoming calls are redirected to the telecommuter's home number and redirected back to call coverage (voice messaging or an attendant) if the telecommuter is busy or unavailable. The seamless connections give the caller the impression that the telecommuter is actually in the office.

The module makes the power of the system available to telecommuters from any touch-tone phone. They can do the following:

- Transfer a call.
- Set up a conference call.
- Use abbreviated dialing.
- Place long-distance calls.
- Receive, leave, and retrieve voice messages.

Telecommuters need not always be at a fixed location, as the target telephone number is easily changed. The modules can be reprogrammed to accommodate different users as well. The module can be set up in two modes:

- Per Session Mode (intensive calling requirements), in which a continuous link is maintained between the telecommuter's phone and the office system. It eliminates the need to log in and log out when making calls. The telecommuter's phone is continuously off-hook, and incoming calls are indicated by a distinctive tone.
- Per Call Mode (moderate calling requirements), in which the employee must log in to make calls or use system features. The module rings the telecommuter's phone when incoming calls arrive, using a distinctive tone. This allows the employee to distinguish between business and personal calls so he or she can answer appropriately.

Each module can be shared by as many as 25 users (though only one may be logged on at any one time). Several security features make it difficult for the system to be abused by hackers.

Personal Station Access

Personal Station Access is a "hoteling" feature that allows you to apply your telephone station preferences and permissions to any compatible telephone. This includes the definition of terminal buttons, abbreviated dial lists, and Class-of-Service and Class-of-Restrictions permissions. It can be used on-site or off-site (with DEFINITY Extender). This would allow several employees to share the same office on different days of the week, with each employee making the shared telephone "theirs" for the day. Remote use requires DEFINITY Extender (described on page 13-2).

NOTE:
Personal Station Access can also be used with the system as a lock and key to prevent unauthorized access.

Station Security Codes

Station Security Codes protect access to telephone stations. Now these codes can be changed by the telephone users. This allows you to easily ensure protection of your telephone features.

All of these features are described in detail in the *DEFINITY® ECS Administrator's Guide,* under the following feature names:

- Call Coverage
- Call Forwarding
- Personal Station Access
- Station Security Codes.

AUDIX features for telecommuting

The following DEFINITY AUDIX and INTUITY AUDIX features are useful for telecommuting:

- *Multiple Personal Greetings* allow subscribers to prepare a pool of up to nine personal greetings to save time and provide more personal customer service. Separate messages can indicate that the subscriber is on the phone, away from the desk, on vacation, or otherwise unavailable to talk. Different messages also can apply to internal, external, or after-hours calls.
- *Outcalling* automatically dials a prearranged phone number or pager when messages are received in a user's mailbox. The system tells whoever answers that messages have been received.

- *Priority Outcalling* provides outcalling notification of priority messages only. This allows the telecommuter to be relatively undisturbed by notifications of messages that do not require immediate attention.
- *Call Answering for Nonresident Subscribers* provides AUDIX System mailboxes for users who do not have an extension number on the system.

For example, when working at home, you set up Priority Outcalling so the system will call you when you have important messages. Then you activate a personal greeting that says something like, “Thanks for calling. I’m working away from the office today. I’ll be checking voice mail periodically, so please leave a message. If your message is urgent, press 2 after recording it. This will give your message priority status. The system will notify me of your priority message almost immediately.”

System Management Solutions

14

Overview

DEFINITY BCS and GuestWorks are digital communications systems that can meet your most demanding voice and data requirements. But what about managing this powerful system? Managing a system was once a formidable task, requiring specially trained administrators who could operate complex programming tools. But, as the capabilities of systems become more sophisticated, so too have the demands placed on the tools used to administer them.

The system offers a variety of easy-to-use modular tools for managing your system. Whether your system is small or large, straightforward or sophisticated, or somewhere in-between, there are tools to effectively and efficiently manage that system.

Why? Because no matter how excellent a communications system is, you must be able to manage it effectively and easily for the system to really work for you. The system gives you that capability by offering easy-to-use tools for managing your system.

This section briefly describes the main areas or functions of system management. Terminal and facility administration features allow you to administer telephones, computers, facilities, and features throughout your system or network. Traffic management features allow you to measure, manage, and report on the voice and data communications traffic throughout your system or network. Maintenance features allow you to view the health of your system and to perform maintenance procedures on your own system if you choose to do so.

This broad system management philosophy extends the system's power and flexibility into the tools for managing the system. These tools are based on the user-friendly architecture which is the hallmark of DEFINITY products. The system management capabilities have been enhanced to accommodate all configurations.

We think this system management view will convince you that the system gives you not only power and flexibility in a communications system but also the power and flexibility to manage that system.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

The system supports the following system management features:

- Local access via DEFINITY Site Administration (standard)
- Local access via the System Management Terminal (optional)
- Multiple, concurrent administration and maintenance sessions
- Terminal administration, using administration without hardware and terminal translation initialization
- Performance measurements
- Call Detail Recording
- Other miscellaneous capabilities.

DEFINITY Site Administration

DEFINITY Site Administration is a single-site DEFINITY system management software application that works with DEFINITY BCS and GuestWorks. DEFINITY Site Administration supports 513 or 4410 terminal emulation and will run on Microsoft Windows 95/98 and Windows NT.

DEFINITY Site Administration uses a set of tools that makes basic system administration easier than using a DEFINITY Management Terminal. These features allow the user to navigate, display, add, modify and/or remove the switch and related data objects. DEFINITY Site Administration also contains a number of features to make common system administration tasks more convenient; these include the following:

- **Browser**

  Allows you to view, add, and change data in the switches and other systems. The DEFINITY Site Administration browser provides a tree view of the switches, voice mail systems, commands, and tasks that you can administer and the icons for accessing the task wizards.

- **Emulation**

  Used for legacy switches or other applications using either the 4410 or 513 protocols.

- **GEDI**

  Graphically Enhanced DEFINITY Interface.

- **BLP**

  Button Label Printing which is used for producing station labels for end-user terminals. The tool can print any text from a template, and can print label rectangles onto plain paper instead of button label sheets.

- **Alarm Monitor**

  Provides monitoring of alarms, and notification via the screen or email.

- **Scheduling/Task Viewer**

  Schedules tasks and monitors progress.

- **Audits**

  Provides a check for unused site data, unused and missing coverage paths, invalid coverage points, and duplicate coverage paths.

- **History Log**

  Provides a history of the changes performed to the systems through DEFINITY Site Administration.

- **Trunk Analyzer**

  Polls trunk data and provides measurements in Erlang B, Erlang C, or CCS. You can dynamically adjust the desired grade of service to recalculate results data.

- **Task-Based Wizards**

  Offers shortcuts to common switch and voice mail administration activities. These wizards present streamlined point and click administration to quickly and easily complete the task.

DEFINITY Management Terminal

The DEFINITY Management Terminal is an optional integrated management tool available with every system. The Management Terminal provides an intuitive interface with forms-based selections, help keys, and a language-based interface (several languages are available).

The system administrator uses the Management Terminal to access the system to perform “task-oriented” administration and maintenance procedures. Several types of asynchronous terminals can be used as the Management Terminal. One such terminal is the Model 715 Multitasking Terminal.

Using the DEFINITY Management Terminal, the system manager can do the following:

- Manage system, voice-terminal, and data-terminal features on a day-to-day basis.
- Perform system backups.
- Monitor system performance.
- Perform selected maintenance procedures.
- Maintain system security.

Concurrent user sessions

To increase the efficiency of administration and maintenance functions, the system accommodates multiple concurrent administration and maintenance user sessions. Up to three users can be connected to the system to perform administration and/or maintenance tasks simultaneously (this limit drops to two concurrent users if the DEFINITY Management Terminal is one of those users). The concurrent sessions can be in any combination of local and remote connections.

This feature increases the volume of administrative activity that can be performed in a given time period, allowing administrators to handle peak demand more effectively.

Telephone Administration

The system includes two features that ease, simplify, and accelerate the administration process.

Administration without hardware

Administration without hardware gives you the ability to administer station forms without specifying a port location. Administered stations 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. Phantom extensions can be used for Automatic Call Distribution Dialed Number Identification Service. This allows a phantom extension to be administered on the system for each call type that needs to be identified to hunt group agents. The phantom Automatic Call Distribution extension either is “call forwarded” (via an attendant console) to an Automatic Call Distribution hunt group or has its coverage path defined to include the Automatic Call Distribution hunt group. The name field administered for the phantom extension will identify to the Automatic Call Distribution agent which service the caller is attempting to reach, thereby allowing the agent to properly address the caller.

Administration Without Hardware also supports the ability to store station templates (models). These templates can later be used with the “duplicate station” command to implement many station forms of the same type in the system.

Administration Without Hardware can be used to streamline system initializations, major additions, and rearrangement/changes by allowing telephone translations to be entered before the actual ports are assigned.

Administration Without Hardware can be used on the following equipment:

- Analog telephones
- Digital Communications Protocol telephones
- Hybrid telephones
- Attendant consoles
- Voice/computers (such as Digital Communications Protocol terminals with voice and data capabilities)
- Data modules
- ISDN-BRI telephones and computers
- Analog queue warning ports
- Recorded announcement ports.

Terminal Translation Initialization

Terminal translation initialization (TTI) is a feature that works with Administration Without Hardware. Terminal translation initialization associates the terminal translation data with a specific port location through the entry of a special feature access code, a terminal translation initialization security code, and an extension number from a terminal that is connected to a wired, but untranslated jack.

Once a terminal is connected to an appropriate jack, the terminal user can dial the appropriate codes followed by a pretranslated extension number of an Administration Without Hardware terminal. The system will complete the administration of the terminal by associating the translation data with the port location and performing appropriate checks.

Terminal translation initialization reduces the labor associated with system initializations, major additions, rearrangement and changes, and building wiring. Translation data entry can be performed without knowledge of the physical layout of circuit packs. End-users can move their own station equipment if a building is wired to support it, thereby reducing costs for station moves. Individual lines need only be wired to the correct type of port, rather than to a specific port.

Administrators maintain control over the use of terminal translation initialization through the use of security codes. By activating and deactivating security codes, administrators can control who uses terminal translation initialization — and when.

Traffic reports

A number of performance measurements are available on the system. These measurements are available in the form of system-based reports for local or remote access, and can be collected for subsequent analysis and reporting by adjuncts and operation support systems using the operation support system interface protocol. These reports include the following:

- The Call Coverage reports display measurements of the distribution of traffic offered to call-coverage groups. Separate reports for all calls and external calls are supplied. Each report has sections that: define group attributes, provide a summary of coverage-group call dispositions, and show the disposition of traffic at each coverage point. You can select which coverage groups are monitored via administration. The report fields are as follows:

  — Group Attributes shows the group number, number of principals, number and type of station at each coverage point, and the number of ring cycles before the call is advanced to the next coverage point.

  — Call Summary shows the number of calls offered, advanced to coverage, answered, and abandoned before being answered for all calls offered to the group and for external calls offered to the group.

  — Coverage Points differs based on whether “All Calls” or “External Calls” is selected. The “All Calls” report shows detail data for all calls to the group; the “External Calls” report shows detail data for only the external calls offered to the group. For each coverage point in the group, the quantity of calls offered, abandoned while at that coverage point, and overflowed to the next coverage point are listed.

  These measurements can be used to engineer group sizes at coverage points and to detect station user abuse of the call-coverage feature.

- The Processor Occupancy report provides summary information on how heavily the processor is loaded. It includes peg counts of the number of various call types and total calling rates for the measurement period. The data fields of this report are as follows:

  — Processor occupancy for call processing (including the link subsystem) plus system management processes

  — Call processing (including the link subsystem), system management, and packet interface processor occupancy

  — Total calls, number of station-to-station calls, number of incoming trunk calls, number of outgoing trunk calls, and number of tandem calls

  These measurements are listed for the last hour, today’s peak hour, and yesterday’s peak hour.

Large systems offer additional measurements that help configure the system, determine the system's capacity for growth, and report unauthorized access attempts. These measurements include the following:

- The Traffic Summary report provides a performance summary of the system. These can be used to verify that your system and its users are not experiencing performance degradation due to overloaded system resources. The following are included in the report:
  - Processor occupancy for call processing and system management
  - Attendant speed of service
  - Total system-network blocking probability, as well as blocking probability of the highest port network and highest center-stage link
  - Total number of security violations as defined in the security violations report
  - A list of the trunk groups that experienced blocking higher than an administered design grade of service
  - Total trunks that are out of service
  - Total number of Call Detail Recording record buffer high-water-mark violations and buffer overflows
  - Time stamps for when the following events last occurred:
    - Major alarm
    - The list of trunk groups to be studied and when they were last changed
    - The list of coverage groups to be studied and when they were last changed
    - The list of Automatic Alternate Routing/Automatic Route Selection routing patterns to be studied and when they were last changed.
- The Attendant Position report lists the following:
  - Attendant usage
  - Number of calls answered
  - Total time the attendant was available to answer a new call
  - Average holding time on calls answered
- Security Violations report collects the following measurements:
  - System Management includes the number of successful and unsuccessful logins, the number of valid and invalid passwords, and the number of times a login name was valid but three successive invalid passwords were entered.

— Call Processing lists the number of valid and invalid authorization codes entered for the system, the stations on the system, all tie trunks, and the attendant consoles. In addition, the time and dial access code/extension from which the last ten violations occurred are recorded.

— Maintenance Board lists the number of valid and invalid attempts to access the maintenance circuit pack.

- The Tandem Traffic report provides information on facilities that serve tandem traffic.

The following measurements help you evaluate the network engineering design for possible reconfiguration. They can help you decide how to reconfigure networks for lower-cost operation.

- Split Measurements lists various information including the number of calls that overflowed the group queue.
- Automatic Route Selection Pattern Measurements collects information on Automatic Route Selection patterns from when the report was administered into the measured pattern list until it is removed from the measured pattern list.
- Trunk Group Detailed Measurements reports on the traffic on a selected subset of trunk groups for a sequence of 24 measurement intervals whose length is customer-selectable among the options of 15 minutes, 30 minutes, or 1 hour. The report is divided into two sections:

  — Group Identification includes the trunk group number, type, direction, and size.

  — Measurements lists total usage, maintenance usage, total calls, incoming calls, tandem calls, group overflow, calls queued, queue overflow, percentage of all trunks busy, and percentage of outgoing blocking.

- The Blockage Study report shows the blockages that occur for Time Division Multiplexing attempts.

All of these measurements are accessible to an external host via the operation support system interface.

Call-Charge Information

The system provides two ways to know the approximate charge for outgoing calls:

- Advice of Charge — For ISDN trunks

  Advice of Charge collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it is recorded as either a charging or currency unit.

- Periodic Pulse Metering — For non-ISDN trunks

  Periodic Pulse Metering accumulates pulses transmitted from the public network at periodic intervals during an outgoing call. At the end of the call, the number of pulses collected is the basis for determining charges.

Call-charge information helps you to account for the cost of outgoing calls without waiting for the next bill from your network provider. This is especially important in countries where telephone bills are not itemized. You can also use this information to let employees know the cost of their phone calls, encouraging them to save money on toll calls.

Call Detail Recording

Also included in the timely and efficient management of your communications system is the management and control of call costs. Call Detail Recording allows you to monitor and analyze call patterns and usage in your system.

The Call Detail Recording feature has the following capabilities:

- Distinguish voice from data on trunk calls.
- Choose whether to record the Call Vectoring number in the “Dialed Number” field of the Call Detail Recording record, or to record the agent’s extension in the same field.
- Allow Call Detail Recording records to be generated for internal calls (calls to and from a set of extensions, including data end points) so administered (a maximum of 100 extensions in large configurations).
- With Call Privacy, allow up to seven digits of the dialed number to be blanked from the Call Detail Recording record.
- Use a second Call Detail Recording port for sending Call Detail Recording data to a second source.

The system includes the Variable Format Records feature, which provides a flexible means of incorporating new fields in the call detail record as new switch features and new Call Detail Recording devices become available. The variable format allows you to define a record in terms of its content (from a set of available data elements), the position of its fields, and the spacing between the fields. This method can be used to construct the 15-, 18-, and 24-word standard formats and custom formats.

If calls come in while the Call Detail Recording link is down and the buffer is completely filled, the system gives you the following administrable call-record handling options:

- Block the calls with reorder.
- Allow the calls to overwrite records.

- Route the calls to an attendant with the option to proceed as a non-Call Detail Recording call.

As you can see, the system call-record handling capabilities are designed to be flexible, adapting to meet your present and future business needs.

Call Detail Recording devices

The following output devices are supported by the system:

- Local storage devices (such as the Call Detail Recording Unit/SE) and any customer-provided storage device with an RS-232C interface
- Processing devices (such as the Lucent Technologies Call Accounting System Plus for Windows, Cost Allocator, or host processors) that are supported over an RS-232C interface with XON/XOFF flow control
- Asynchronous ASCII printers with an RS-232C interface.

The enhanced variable format records feature supports any customer-defined data presentation, and therefore can support any devices over an RS-232C interface.

Call Accounting System for Windows

The Call Accounting System for Windows allows you to generate comprehensive and accurate accounting reports using the familiar Microsoft Windows environment, which allows you to run several tasks at once. Detailed or summary reports can be expressed in two- or three-dimensional, color charts and graphs, or in text files suitable for downloading to other applications.The optional toll fraud detection module allows you to detect fraudulent use of your long-distance services.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

You can generate reports that identify the following:

- Most frequently dialed numbers
- Most expensive calls
- Longest duration calls.

You can search the accounting data for a variety of information, including dialed numbers, partial numbers, dates, times, call types, departments, and calling extensions.

This enables you to reduce telephone expenses, optimize resources, assign costs, and identify abuse. The Call Accounting System for Windows helps you to understand your telephone expenses and convey that understanding to others.

You can define up to five levels of reporting hierarchy to which you can assign costs. The system archives your data for one accounting period. A flexible markup capability allows service businesses to adjust call pricing for each client.

Call Accounting System for Windows can generate 20 standard historical or real-time reports from as many as 100 locations. An individual system is capable of polling different types of call detail storage units or other Call Accounting System for Windows systems. The remote systems forward call records and alarms as the call records are generated.

A traffic engineering option allows you to monitor trunk usage, calling patterns, incoming traffic, and outgoing calls by area code. This allows you to analyze trends that summarize how your equipment is being used.

Call Accounting for Windows is widely compatible and requires little maintenance, even while collecting data, generating reports, and managing remote data collection sites.

Call Accounting System Terminal

Lucent Technologies Call Accounting System Terminal is an easy-to-install hardware and software package that allows you to assign expenses to as many as three organizational levels. For example, you can assign costs at the department, cost center, or extension level.

The system makes it easy for you to generate a wide variety of accounting and system reports. For example, the Facility Grade of Service Report helps identify the number of trunk lines needed to respond efficiently to incoming calls. You can also generate toll fraud reports and alarms that identify excessive personal calls, unauthorized calls, and calls to expensive dial-up recordings.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

INTUITY Lodging Call Accounting System

If you are using the INTUITY Lodging voice messaging product in a hospitality environment, the INTUITY Lodging Call Accounting System is available as a call accounting solution to you. The system, developed by Homisco, works exclusively with INTUITY products, which reside on a dedicated computer co-resident with the INTUITY Lodging system. (For more information on INTUITY products, see

Chapter 10, ''Voice Messaging Solutions.'') While offering many of the same features as the Call Accounting System for Windows (described in the previous section), the system also serves to help integrate your INTUITY Lodging applications.

Call Detail Recording Unit/SE

The Call Detail Recording Unit/SE can be used when call detail record storage is more than 7000 records.

NOTE:
Some features and solutions are unavailable in some countries. Please contact your local account manager or authorized Lucent Technologies representative for further information about which features and solutions are available to you.

The Call Detail Recording Unit/SE is a special-purpose processor containing a built-in modem and permanently-installed Call Detail Recording Unit software. It is a local Call Detail Recording storage device that collects, optionally filters, and stores Call Detail Recordings from a variety of systems, including the switch.

Upon request from a Lucent Technologies polling system, the Call Detail Recording Unit/SE transmits all Call Detail Recordings received since the last poll. The same Call Detail Recordings can be retrieved by up to two polling systems. The polled Call Detail Recordings are then available for processing via the Lucent Technologies Cost Allocator Call Processing Accounting Management Solution. The following are some of the Call Detail Recording Unit/SE highlights:

- Stores 77000 Call Detail Recordings for a 24-word record
- Stores 127000 Call Detail Recordings for an 18-word record
- Supports 15-, 18-, and 24-word Call Detail Recordings including ISDN
- Supports fixed and variable length Call Detail Recordings
- Supports non-Lucent Technologies record formats up to 132 ASCII characters plus end-of-record characters
- Collects up to 3600 Call Detail Recordings per hour
- Can be remotely-administered
- Provides 72 hours of call record retention via 9-volt battery backup
- Uses three filtering options based on input from the system to reduce the number of useless Call Detail Recordings
- Monitors/displays Call Detail Recordings as they are collected
- Offers automatic initialization and power failure recovery
- Supports polling or remote maintenance via a built-in, 1200 bps or 2400 bps modem

- Uses a password for security protection for polling and administration
- Has alarm relay contacts for wiring to an alarm reporting device
- Performs on-line diagnostic tests (ROM, RAM, internal clock)
- Provides an on-line, real-time system status report.

Other management capabilities

The following are other management features that enhance your investment in the system:

- Access Security Gateway
- Security Violation Notification
- Call Restrictions
- Reporting Capabilities
- System-based Reports.

Access Security Gateway

Access Security Gateway is an authentication interface used to secure the system administration and maintenance ports and/or logins on the system. Access Security Gateway employs a challenge/response protocol to confirm the validity of a user and reduce the opportunity for unauthorized access. Successful authentication is accomplished when the feature communicates with a compatible key. The challenge/response negotiation is initiated once an RS-232 session is established and a valid system login ID has been supplied by a user. The authentication transaction consists of a challenge, issued by the system and based on the login ID supplied by the user, followed by receipt of the expected response, which is supplied by the user.

Implemented using a symmetric key form of cryptography, the core of this scheme is a secret key, which is information possessed by both the lock and the key. Interception of either the challenge or response during the course of transmission will not compromise the security of the system, as the relevance of the authentication token used to perform the challenge/response is limited to the current challenge/response exchange. The challenge/response tool is called the Access Security Gateway Key. This is either a calculator-like device or a program you can install on your PC.

Security Violation Notification

Security violation notification identifies potential hackers' attempts to access the system. It notifies you when the number of invalid login attempts is greater than the administered threshold. A monitor report displays the last 16 invalid login attempts. This report is automatically updated every 30 seconds.

Call Restrictions

By dialing an access code, administrators and attendants have the ability to restrict users from making or receiving certain types of calls. There are five restrictions:

- Outward — User cannot place external calls.
- Station-to-station — User cannot place or receive internal calls.
- Termination — User cannot receive any calls (except priority calls).
- Toll — User can place local calls, but cannot place toll calls.
- Total — User can neither place nor receive calls.

The risks of unauthorized access can be minimized by combining the use of Remote Access with the following:

- Have an unpublished remote access number.
- Deactivate unassigned barrier codes immediately.
- Change barrier codes frequently.
- Inform remote access users of their responsibility.
- Monitor call detail reports for unauthorized or abnormal calling patterns.

Reporting Capabilities

Ongoing management of your system can be enhanced by data made available through reports. The system gives you several options for obtaining reports.

System-Based Reports

The system has built-in capabilities for generating reports required for all systems. These reports are available without special hardware or software.

System Measurements reports supply information on the status of all communication facilities. These reports help determine the efficiency of resources including, but not limited to, trunk groups, hunt groups, and the attendant group.

System Status reports supply information associated with the attendant group, major and minor alarms, and traffic measurements.

The Recent Change History feature reports on the most recent administration and maintenance commands that have been entered. The system also supplies the following:

- New site data on the station form. New fields include the set color, building, floor, and headset. In addition, user-defined validation checks are provided for a subset of the site data items.

- Scaling enhancements, as well as a ranging and filtering capability, for large systems. These allow your administrator to restrict data reporting to only the desired number of parameters.

The system also includes the following reports:

- Class-of-Restriction report lists the extensions that have a particular Class-of-Restriction value or that fall within a range of Class-of-Restriction values.
- Class-of-Service report lists the extensions that have a particular Class-of-Service value or that fall within a range of Class-of-Service values.
- Site Data report lists, by extension, the site data associated with stations in the system. Ranging and filtering capabilities are provided for selected site fields.

Features

A

Overview

This appendix provides a description of each feature supported with the DEFINITY BCS or GuestWorks offers. The features are grouped in the following categories:

- "Automatic Routing features" on Page A.-2
- "Basic features" on Page A.-5
- "Hospitality features" on Page A.-39
- "Hunt Group features" on Page A.-44
- "Private Networking features" on Page A.-47
- "Trunk Group features" on Page A.-49.

Each feature is described briefly, though most features have many complex capabilities and options. The *DEFINITY® ECS Administrator's Guide* describes each feature in detail and provides complete implementation and administration information. Some features, such as Call Detail Recording and AUDIX, are systems of their own and have their own documentation. See your local Lucent Technologies representative or distributor for more information on each of these features.

NOTE:
Not all features are available with each model of the system. Please see the *DEFINITY® ECS System Description* for information on feature availability by model. In addition, not all system applications or adjunct applications may be available in all countries. Please check with your local Lucent Technologies representative for further information about what is available in your country. Information about these country differences can be found in *DEFINITY® ECS Application Notes for Type Approval*. This document is currently available from your Lucent Technologies Center of Excellence (COE).

This appendix also contains information about the Dial by Name feature on Page A-53. This is a special feature of DEFINITY BCS and GuestWorks that is not described in any other DEFINITY documents. Please share this information with any personnel that need to use or administer this feature.

Automatic Routing features

Provides a variety of automatic-routing features for public and private networks. Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) are the foundation for these automatic-routing features. They route calls based on the preferred (normally the least expensive) route available at the time the call is placed. Generally, AAR routes calls over a private network, and ARS routes calls using the public network numbering plan. However, both AAR and ARS support public and private networks. You can use the other features listed in this section when you use AAR and ARS.

Automatic Alternate Routing

Automatic Alternate Routing (AAR) allows private network calls to originate and terminate at one or many locations without accessing the public network. When a user dials an access code and phone number, AAR selects the most desirable route for the call and performs digit conversion as necessary. If the first choice route is unavailable, another route is chosen automatically.

The numbers called using AAR are normally private-network numbers. However, users can call a public-network number, a service code, an international number, operator access code, or an operator-assisted dialing number. With AAR and Subnet Trunking, users have a convenient way to place international calls to frequently-called foreign cities. Such calls route as far as possible over the private network, and then access the public network. This saves toll charges and allows users to use your private network as much as possible.

Automatic Route Selection

Automatic Route Selection (ARS) routes public network calls on the most desirable (usually the most economical) trunking facilities available on your system when the call destinations are accessible through your public network.

The system supports up to 40 routing patterns. Each routing pattern consists of up to 16 routing preferences (types of facilities) set up in the order in which you want them checked when a call is placed. Typically, the least expensive facility will be first on the list; the most expensive will be last.

If Generalized Route Selection is not being used when a call is made, the system selects a routing pattern based on the digits dialed. The routing preferences in that pattern are checked in the order in which they were listed, and the first available facility is used to place the call. If no facility is available, the call can be queued until a facility becomes available.

AAR/ARS Overlap Sending

Overlap sending can be used on AAR and ARS calls that are routed over ISDN trunk groups. Overlap sending sends ISDN call-address information one digit at a time instead of all the address information going out in one block. This significantly decreases call setup time in countries with complex public-network numbering plans, and is most useful for tandemed calls.

AAR/ARS Partitioning

Allows AAR and ARS to be partitioned into eight user groups within a single system and provides individual routing treatment for each of these user groups.

User groups share the same Partition Group Number, which indicates the choice of routing tables that are used on a particular call. Each Class of Restriction is assigned a specific Partition Group Number or Time-of-Day specification. Different classes of restriction may be assigned the same Partition Group Number.

Alternate Facility Restriction Levels

Allows the system to adjust facility restriction levels or authorization codes for lines or trunks. Each line or trunk is normally assigned a facility restriction level. With this feature, alternate facility restriction levels are also assigned. Attendants can change to the alternates, thus changing access to lines and trunks. Users might want to use this feature to disable most long-distance calling at night, for example, to prevent unauthorized staff from making long-distance calls.

CAUTION:

This feature may change the AAR and ARS routing preferences. Using it on tandem and tie-trunk applications affects entire networks. Calls that are part of a cross-country private network may be blocked.

Facility Restriction Levels and Traveling Class Marks

Allows certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed to use central office trunks to other corporate locations, while other users may be restricted to less expensive private-network lines. Up to eight levels of restriction may be administered for users of AAR and ARS.

Generalized Route Selection

Provides voice and data call-routing capabilities. It is used to select not only the least-cost routing, but also optimal routing over the appropriate facilities. It enhances AAR and ARS by providing additional parameters in the routing decision and maximizing the chance of using the right facility to route the call. Also, if endpoint incompatibility exists, Generalized Route Selection provides a conversion resource (such as a modem from a modem pool) to attempt to match the right facility with the right endpoint.

Look-Ahead Routing

Provides an efficient way to use trunking facilities. It allows the system to continue to try to reroute an outgoing ISDN-PRI call that is not completing. When the system receives a cause value that indicates congestion, Look-Ahead Routing tells the system what to do next. For each routing preference, you can indicate whether the next routing preference should be attempted or if the current routing preference should be attempted again.

NOTE:
This feature is not related to the Look-Ahead Interflow feature. The Look-Ahead Interflow feature is not supported on DEFINITY BCS nor GuestWorks.

Subnet Trunking

Subnet Trunking modifies the number dialed so an AAR or ARS call can route over different trunk groups that may terminate in switches with different dial plans. Subnet Trunking inserts digits, deletes digits, pauses, and/or waits for dial tone in digit outpulsing, as required, so calls route as follows:

- To or through a remote switch
- Over Tie trunks to a private network switch
- Over CO trunks to the serving CO.

Subnet Trunking is required on calls routing to or through a remote switch, regardless of the call's destination.

Time-of-Day Routing

Time-of-Day Routing allows you to select the most economical routing of ARS and AAR calls based on the time of day and the day of the week a call is made. Up to eight Time-of-Day routing plans may be administered, each scheduled to change up to six times a day for each day in the week.

With Time-of-Day Routing, your company can take advantage of lower calling rates during specific times. If your company has locations in different time zones, you can maximize the use of your public or private network facilities by utilizing those facilities in the location that has the lowest calling rates at the particular time a call is made. You can also use this feature to change the routing patterns when an office is closed and to eliminate unauthorized calls.

Basic features

The following features come standard with the system.

Abbreviated Dialing

Provides lists of stored numbers users can use to do the following:

- Place local, long-distance, and international calls.
- Activate features.
- Access remote computer equipment.

Users dial the list number and the one-, two-, or three-digit number associated with the phone number the user wants. The number is then automatically dialed by the system. A frequently called number can be stored on an abbreviated dialing button that a user needs to press once to make the call.

Access Security Gateway

Access Security Gateway is an authentication interface used to secure the system administration and maintenance ports and/or logins on the system. Access Security Gateway employs a challenge/response protocol to confirm the validity of a user and reduce the opportunity for unauthorized access. Successful authentication is accomplished when the feature communicates with a compatible key. The challenge/response negotiation is initiated once an RS-232 session is established and a valid system login ID has been supplied by a user. The authentication transaction consists of a challenge, issued by the system and based on the login ID supplied by the user, followed by receipt of the expected response, which is supplied by the user.

Active Dialing

The 6400-series telephones have a dialing option that allows the set to send S-channel button codes when the user presses a number on the dial pad while on-hook. This allows the user to begin dialing without first going off-hook.

Administered Connections

Automatically establishes an end-to-end connection between two access or data endpoints based on administered attributes. This feature provides capabilities such as:

- Alarm notification, including an administrable alarm type and threshold
- Automatic restoration of connections established over a Software-Defined Data Network
- ISDN-PRI trunk group [service may be referred to as ISDN-PRI (AC/AE) Service]
- Scheduled as well as continuous connections
- Administrable-retry interval for failed connection attempts.

Administrable Language Displays

Allows the messages that appear on telephone display units to be shown in the language spoken by the user. These messages are available in English (the default), French, Italian, Spanish, or one other user-defined language. The language for display messages is selected by each user. The feature requires 40-character display telephones.

Administrable Loss Plan

The Administrable Loss Plan provides the ability to administer signal loss and gain for telephone calls. To do this, switch endpoints are classified into 17 endpoint types, and the loss plan can be administered for trunks, stations, and personal CO lines. Loss values are in the range of 15 dB loss to 3 dB gain. Preset defaults are available and are based on country type.

Administration Without Hardware

Allows you to administer telephones that are not yet physically present on the system. This feature works the same as administration with hardware: when stations are moved, user-activated features such as Call Forwarding and Send All Calls are preserved and functional. This greatly facilitates the speed of setting up and making changes to the telephones on the system.

Alphanumeric Dialing

Allows users to place data calls by entering an alphanumeric name rather than a long string of numbers.

Alternate Operations Support System Alarm Number

Allows you to establish a second number for the system to call when an alarmable event occurs. This feature is useful for alerting a second support organization, such as INADS or OneVision.

Answer Detection

For purposes of call-detail recording, it is important to know when the called party answers a call. The system provides three ways to determine whether the far end has answered an outgoing call.

- Network Answer Supervision — The central office (CO) sends back a signal to indicate that the far end has answered the call. If a call has traveled over a private network before reaching the CO, the signal is transmitted back over the private network to the originating system. This method is extremely accurate, but is not available in the United States over CO, FX, or WATS trunks.
- Answer Detection — A call-classifier circuit pack detects tones and voice-frequency signals on the line and determines whether a call has been answered. This method is fairly accurate.
- Answer Supervision by Timeout — A timer is set for each trunk group. If the caller is off-hook when the timer expires, the system assumes that the call has been answered. This is the least accurate method. Calls that are shorter than the timer duration do not generate call records, and calls that ring for a long time produce call records whether they are answered or not.

Attendant Auto-Manual Splitting

Allows an attendant to announce a call or consult privately with the called party without being heard by the calling party on the call. It splits the calling party away so the attendant can confidentially determine if the called party can accept the call.

Attendant Backup

Notifies backup attendants that the primary attendant cannot immediately pick up a call. It provides both audible ringing and visual alerting to backup stations when the attendant queue reaches its queue warning level. When the queue drops below the queue warning level, alerting stops. Audible alerting also occurs when the attendant console is in night mode, regardless of the attendant queue size.

Attendant Call Waiting

Allows an attendant to let a single-line telephone user who is on the phone know that a call is waiting. The attendant is then free to answer other calls. The attendant hears a call waiting ringback tone and the busy telephone user hears a call waiting tone. This tone is heard only by the called telephone user.

Attendant Calling of Inward Restricted Stations

A telephone with a Class of Restriction that is inward-restricted cannot receive public network, attendant-originated, or attendant-extended calls. This feature allows attendants to override this restriction.

Attendant Console

A digital call-handling station with push-button control used not only to answer and place calls, but also to manage and monitor some system operations.

Attendant Control of Trunk Group Access

Allows an attendant to control trunk groups and prevents telephone users from directly accessing a controlled trunk group. 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.

Attendant Direct Extension Selection with Busy Lamp Field

Allows the attendant to keep track of extension status — whether the extension is idle, busy, or has Send All Calls active — and to place or extend calls to extension numbers without having to dial the extension number. The attendant can use this feature in two ways: using standard Direct Extension Selection access, or using enhanced Direct Extension Selection access.

If the user's extension is idle, the lamp is dark. If the user is busy on a call, the lamp lights steadily. If the user has Send All Calls activated, the lamp flashes.

Attendant Direct Trunk Group Selection

Allows the attendant direct access to an idle outgoing trunk by pressing the button assigned to the trunk group. This feature eliminates the need for the attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. Pressing a labelled button selects an idle trunk in the desired group.

Attendant Display

Shows call-related information that helps the attendant to operate the console. Also shows personal service and message information. Information is shown on the alphanumeric display on the attendant console. Attendants may select one of several available display message languages: English, French, Italian, or Spanish. In addition, your company may define one additional language for use by users and attendants on their displays.

Attendant Intrusion (Call Offer)

Allows an attendant to enter an existing call to inform the person being called about a message or another call. If administered, an intrusion tone warns the callers that the attendant is breaking in on the call.

Attendant Override of Diversion Features

Allows an attendant to bypass diversion features such as Send All Calls and Call Coverage by putting a call through to an extension even when these diversion features are on. This feature, together with Attendant Intrusion, can be used to get an emergency or urgent call through to a telephone user.

Attendant Priority Queue

Places incoming calls to the attendant in an orderly queue when these calls cannot go immediately to the attendant. This feature allows you to define 12 different categories of incoming attendant calls, including emergency calls, which are given the highest priority.

Attendant Recall

Allows users to recall the attendant when they are on a two-party call or on an Attendant Conference call held on the console. 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 remain on the connection when either button is used.

Attendant Release Loop Operation

Allows the attendant to hold a call off the console if the call cannot immediately go through to the person being called. A timed reminder begins once the call is on hold. If the call is not answered within the allotted time, the call returns to the queue for the attendant. Timed reminders attempt to return the call to the attendant who previously handled it. Only when the original attendant is unavailable are calls returned to the queue.

Attendant Serial Calling

Enables an attendant to transfer trunk calls that return to the same attendant after the called party hangs up. The caller, after reconnecting with the attendant, can then be transferred to another station within the switch. This feature is useful if trunks are scarce and Direct Inward Dialing services are unavailable. An outside caller may have to redial often to get through because trunks are busy. Once callers get through to an attendant, they can use the same line into the switch for multiple calls. The attendant's display shows if an incoming call is a serial call.

Attendant Split Swap

Allows the attendant to alternate between active and split calls. This operation may be useful if the attendant needs to transfer a call, but first must talk independently with each party before completing the transfer.

Attendant Trunk Group Busy/Warning Indicators

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. This feature is particularly helpful in showing the attendant that the Attendant Control of Trunk Group Access feature needs to be invoked.

Audible Message Waiting

Places a stutter at the beginning of the dial tone when a telephone user picks up the phone. The stutter dial tone indicates that the user has a message waiting. This feature is particularly useful for visually impaired people who may not be able to see a message light. It is often used with telephones that have no message waiting lights, but may not be available in countries that restrict the characteristics of dial tones provided to users.

Audio Information Exchange Interface

AUDIX is a message-handling system for recording and distributing spoken messages or voice mail. Stored voice prompts guide users in creating, sending, retrieving, answering, saving, and forwarding spoken messages.

Several versions of AUDIX are available. DEFINITY AUDIX is comprised of a circuit pack resident in the switch. INTUITY AUDIX is external to the switch and connects to the switch by station lines and data links. AUDIX systems can also be networked through switches or through other AUDIX systems. The systems rely on a data link between the AUDIX adjunct on the switch and the other AUDIX systems.

Authorization Codes

Authorization codes are used on particular calls to temporarily raise a telephone's Facility Restriction Level. This is useful for those who make calls from telephones other than their own or from outside the network. If a call you dial is blocked because the telephone's Facility Restriction Level is too low, you can enter your authorization code. If the Facility Restriction Level associated with the authorization code is equal to or higher than the Facility Restriction Level of the trunk facilities required to place the call, the call is then completed. Up to 5000 (csi/si) or 90000 (r) different authorization codes can be provided for your system at any one time. Authorization codes can be from 4 to 13 digits long. Using the system's management tools, you can assign authorization codes and change their associated Facility Restriction Level and network access permissions.

Auto Start and Don't Split

Allows the attendant to make a telephone call without pushing the start button first. If the attendant is on an active call and presses digits on the keypad, the system automatically splits the call and begins dialing the second call. The Don't Split feature deactivates the Auto Start feature and allows the sending of touch tones over the line for the purposes of such things as picking up messages.

Automated Attendant

Automated Attendant (formerly known as Direct Access Calling) uses call vectoring commands to allow a caller to enter the extension of the party or department the caller wishes to reach. The call is routed to that extension by the vector.

Automatic Callback

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

When a user activates Automatic Callback, the system monitors the called telephone. When the called telephone becomes available to receive a call, the system originates the Automatic Callback call. The originating party receives priority ringing. The calling party then lifts the handset and the called party receives the same ringing provided on the original call.

Automatic Circuit Assurance

Assists in identifying possible trunk problems. The system maintains a record of the performance of individual trunks and automatically calls a designated user when a possible failure is detected. This feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time.

Automatic Incoming Call Display

Displays information about an incoming call even if the telephone is already in use.

Automatic Transmission Measurement System

Measures voice and data trunk facilities for satisfactory transmission performance. The measurement report contains data on trunk signal loss, noise, signaling return loss, and echo return loss. Acceptable performance, the scheduling of tests, and report contents are administrable.

Barrier Codes

A security code used with Remote Access to prevent unauthorized access to your system. To increase your system's security, use a 7-digit barrier code with Remote Access Barrier Code Aging. A barrier code automatically expires if an expiration date or number of accesses has exceeded the limits you set. If both a time interval and access limits are administered for a barrier code, the barrier code expires when one of the conditions is satisfied.

NOTE:
Barrier codes are *not* tracked by Call Detail Recording (CDR). Barrier codes are incoming access codes, whereas, authorization codes are primarily outgoing access codes.

Bellcore Calling Name ID

Allows the system to accept calling name information from a local exchange carrier (LEC) network that supports the Bellcore calling name specification. The system can send calling name information in the format if Bellcore Calling Name ID is administered. The following Caller ID protocols are supported.

- Bellcore (default) - US protocol (Bellcore transmission protocol with 212 modem protocol)
- V23-Bell - Bahrain protocol (Bellcore transmission protocol with V.23 modem protocol).

Block Collect Call

Blocks collect calls on class-of-restriction basis. This feature is available for any switch that uses the Brazil country code. If enabled for a station, all trunk calls that terminate to the station will send back a double answer to the CO. This double answer tells the CO that this particular station cannot accept collect calls. The CO then tears down the call if it is a collect call.

Bridged Call Appearance — Multi-Appearance Telephones

Allows calls to be handled from more than one telephone. A bridged call appearance is set up by administering a primary extension and the button number associated with it on a two-lamp button on another telephone. One way this feature is most often used is by secretaries or assistants who answer or handle calls to the primary extension (an executive, for example). When the primary extension receives a call, the bridged call appearance flashes or rings, and the call can be handled as if the primary extension user was answering it. You can have up to 64 bridged call appearances.

Bridged Call Appearance — Single-Line Telephones

Allows single-line telephones to have a bridged appearance on a multi-appearance phone. You can have up to 64 bridge call appearances.

Bulletin Board

The bulletin board is a place on the switch where people can post information and receive messages from other switch users, including Lucent Technologies personnel. Anyone with appropriate permissions can use the bulletin board for everyday messages. In addition, Lucent Technologies personnel can leave high-priority messages, which are displayed on the first ten lines of the bulletin board.

Busy Verification of Terminals and Trunks

Allows attendants and users of multi-appearance telephones to make test calls to trunks, telephones, and hunt groups to check the status of an apparently busy resource. With this feature, an attendant or multifunction telephone user can distinguish between a telephone that is truly busy and one that only appears busy because of some problem. Users can quickly identify faulty trunks.

Call Charge Information

The system provides two ways to know the approximate charge for calls made on outgoing trunks:

- Advice of Charge — For ISDN trunks

  Advice of Charge (AOC) collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it is recorded as either a charging or currency unit.

- Periodic Pulse Metering — For non-ISDN trunks

  Periodic Pulse Metering (PPM) accumulates pulses transmitted from the public network at periodic intervals during an outgoing trunk call. At the end of the call, the number of pulses collected is the basis for determining charges.

Call-charge information helps you to account for the cost of outgoing calls without waiting for the next bill from your network provider. This is especially important in countries where telephone bills are not itemized. You can also use this information to let employees know the cost of their phone calls, and to encourage them to help manage the company's telecommunications expenses.

NOTE:
This feature is not offered by the public network in some countries, including the United States.

Call Coverage

Provides automatic redirection of calls that meet specified criteria to alternate answering positions in a Call Coverage path. A coverage path can include any of the following: a telephone, an attendant group, a uniform call distribution hunt group, a direct department calling hunt group, an automatic call distribution hunt group, a voice messaging system, or a coverage answer group established to answer redirected calls.

In addition to redirecting a call to a local answering position, Call Coverage can be administered to do the following:

- Redirect calls based on time-of-day.
- Redirect calls to a remote location.

Call Detail Recording

Records detailed call information on incoming and outgoing calls for the purpose of call accounting and sends this call information to a call detail recording output device. You can specify the trunk groups and extensions for which you want records to be kept as well as the type of information to be recorded. You can keep track of both internal and external calls. This application contains a wide variety of administrable options and capabilities.

Call Forwarding

Call Forwarding provides four functions:

- Call Forwarding All Calls — Allows calls to be forwarded to an internal extension, external (off-net) number, an attendant, or an attendant group.

- Call Forwarding Override — Allows the user at the forwarded-to extension to override Call Forwarding and either initiate a call or transfer a call back to the forwarded-from extension.
- Call Forward Busy/Don't Answer — Allows calls to be forwarded when the called extension is busy or when the call is not answered after an administrable interval. If the extension is busy, the call forwards immediately. If the extension is not busy, the incoming call rings the called extension, then forwards only if the call remains unanswered longer than the administered interval.
- Call Forwarding Off-Net — Allows calls forwarded off-net to be tracked for busy or no-answer conditions. The system brings the call back for further call-coverage processing if the Coverage of Calls Redirected Off-Net (CCRON) feature is active. This feature is particularly useful for telecommuters who can have their on-site office calls forwarded to their home offices.

Call Park

Allows users to put a call on hold and then retrieve that same call from any other telephone on the system. This is helpful when a user is on a call and needs to go to another location for information. It also allows a user to answer a call from any telephone after being paged.

Call Pickup

Along with Directed Call Pickup, allows a user to answer calls for other telephones within a specified call pickup group. Directed Call Pickup allows a user to pick up any call on the system. With this feature, users do not have to leave their location to answer a call for a nearby telephone. The user simply dials an access code or presses a Call Pickup button.

Call Pickup — Group

Allows you to answer a call that is ringing in a different call pickup group than your own. This feature is ideal if you work in an open office environment and have a need to answer phones ringing in a different group. You dial a facilities access code (FAC) and then a pickup group number to answer a call from a different call pickup group.

Call Timer

Automatically starts the local timer of a 6400-series telephone when a call is received. The timer stops automatically when the call ends, but is displayed for a few seconds. When a call is placed on hold, the timer continues to run but is not displayed. When the call comes off hold, the total elapsed call time displays.

Call Waiting Termination

Allows users of single-line telephones who are on a call to be notified of a second call. This feature enables the second call to wait and sends a distinctive call waiting tone to the user who is being called.

Calling/Connected Party Number Restriction

Per-Line CPN Restriction

Users may block the calling party number when originating calls. For ISDN calls, the Calling/Connected Party Number (CPN) Presentation Indicator is encoded accordingly. For non-ISDN calls to a public network that supports the CPN Restriction feature, the network specific Feature Activation Code is passed to the network for interpretation and activation.

If Per-Line CPN Restriction is administered for a station, it will override any ISDN trunk group administration for sending Calling Party Number.

Per-Call CPN Restriction

Users may indicate Calling Number privacy information. For ISDN calls, the CPN Presentation Indicator is encoded accordingly. For non-ISDN calls to a public network that supports the CPN Restriction feature, the network-specific Feature Activation Code is passed to the network for interpretation and activation.

If Per-Call CPN Restriction is activated for an outgoing call, it will override any Per-Line CPN Restriction administered for the calling station, and will override any ISDN trunk group administration for sending Calling Party Number.

Class of Restriction

Defines many different classes of call origination and termination privileges. Systems may have no restrictions, only a single class of restriction, or may have as many classes of restrictions as necessary to effect the desired restrictions. Many different types of classes of restriction can be assigned to many types of facilities on the switch. For example, you can use a calling-party COR to prevent callers from accessing the public network.

Class of Service

Defines whether telephone users can access the following features and functions: Automatic Callback, Call Forwarding, Data Privacy, Priority Calling, Restrict Call Forwarding Off-Net, Call Forward Busy/Don’t Answer, Personal Station Access, Extended Forwarding and Busy/Don’t Answer, Trunk-to-Trunk Transfer Restriction Override, Off-Hook Alert, Console Permission, or Client Room.

Code Calling Access

Allows attendants, users, and tie trunk users to page someone using coded chime signals. This feature is helpful for users who are often away from their telephones or at a location where a ringing telephone might be disturbing.

Conference — Attendant

Allows an attendant to set up a conference call for as many as six conferees, including the attendant. Conferences from inside and outside the system can be added to the conference call.

Conference — Telephone

Allows multi-appearance telephone users to set up six-party conference calls without attendant assistance. Single-line telephone users can set up three-party conference calls without attendant assistance.

Consult

Allows a covering user, after answering a call received through Call Coverage, to call and consult with the originally-called party. Consult can be used to let a covering user ask the called party if they want to speak with the calling party.

Controlled Restrictions

Allows an attendant or telephone user with console permission to activate and deactivate for an individual telephone or a group of telephones the following restrictions: outward, total, station-to-station, toll, and termination restrictions.

Coverage Callback

Allows a covering user to leave a message for the called party to call back the person who called.

Coverage Incoming Call Identification

Allows multi-appearance telephone users without a display in a Coverage Answer Group to identify an incoming call to that group.

Coverage of Calls Redirected Off-Net

Coverage of Calls Redirected Off-Net (CCRON) allows calls that have been redirected to locations outside of the switch to return to the switch for further processing. For example, an employee that telecommutes can have two coverage paths. One coverage path is used when in the office and the other coverage path is used when working from home. The coverage path used from home would have a call to the employee's work phone cover to his or her home phone. If the employee does not answer the call or is busy on another call, the call is redirected back to the switch for further processing, such as coverage to voice mail.

Crisis Alert

Visibly and audibly alerts attendant consoles, administered digital display stations or TransTalk telephones, or digital numeric pagers when an emergency call is placed. The feature indicates from where an emergency call is made (room or extension), which allows the attendant or other personnel to direct emergency-service personnel to the caller. Though often used in the hospitality industry, it can be set up to work with any standard attendant console, digital display station, or digital numeric pager.

Audible alerting sounds like an ambulance siren. Visual alerting consists of flashing of the crisis-alert button lamp and display of the caller name and extension. When crisis alerting is active, the console or station is placed in position-busy mode so that other incoming calls can not interfere with the emergency call notification. The console or station can still originate calls to allow notification of other personnel. Once a crisis alert call has arrived at a console, the console user must press the position-busy button to unbusy the console, and press the crisis-alert button to deactivate audible and visual alerting.

Customer-Provided Equipment Alarm

Provides you with an indication that a system alarm has occurred and that the system has attempted to contact a service organization. A device that you provide, such as a lamp or a bell, is used to indicate the alarm situation. You can administer the level of alarm about which you want to be notified.

Data Call Setup

Enables the setting up of data calls using a variety of methods such as keyboard dialing, telephone dialing, Hayes command dialing, permanent switched connections, administered connections, automatic calling unit interface, and hotline dialing. Data Call Setup is provided for both DCP and ISDN-BRI telephones.

Data Hot Line

Provides for automatic placement of a data call when the originator hangs up. Data Hot Line may be used for security purposes. This feature offers fast and accurate call placement to commonly-called data endpoints. Data terminal users who constantly call the same number can use Data Hot Line to automatically place the call when they hang up the telephone.

Data Privacy

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data Privacy is activated when a user dials an activation code at the beginning of the call.

Data Restriction

Like Data Privacy, this feature protects analog data calls from being disturbed by any of the system's overriding or ringing features. It is administered at the system level to selected analog and multi-appearance telephones and trunk groups.

Default Dialing

Provides data terminal users who dial a specific number the majority of the time with a very simple method of dialing that number. This feature enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a preadministered destination in several different ways, depending on the type of data module. Data Terminal Dialing and Alphanumeric Dialing are unaffected.

Demand Print

Allows a user to print undelivered messages without calling the Message Center.

Dial Access to Attendant

Allows a user to reach an attendant by dialing an access code. The attendant can then extend the call to a trunk or to another telephone.

Dial by Name

Allows a caller to "dial" someone by entering their name from their touch-tone keypad. This feature is accessible by using the Call Vectoring feature and the integrated announcement circuit pack (TN750C) to create an "auto-attendant" procedure in which one of the options allows callers to enter a person's name instead of that person's extension number. The system processes the name characters received, and, if a single match is found, the number is dialed automatically.

The Dial-by-Name feature is described in greater detail in "Dial by Name" on Page A.-53.

Dial Plan

The dial plan is the system’s guide to digit translation. When the system receives dialed digits, the system must know what to expect next based on the digits received so far. For example, if a user dials 4, the dial plan tells the system how many more digits to expect before the call is processed.

Dialed Number Identification Service

Displays, for a called party or answering position, the service or product associated with an incoming call.

Directory

Allows users with display-equipped telephones to access the system database, use the touch-tone buttons to enter a name, and retrieve an extension number from the system directory. The directory contains the names and extensions assigned to all telephones on the system.

Distinctive Ringing

Helps users and attendants distinguish among various types of incoming calls by distinctive ringing patterns. Users can set up ringing patterns to indicate many different types of calls: for example, internal, external, and priority calls.

By default, internal calls ring with a one-burst pattern, external calls with a two-burst pattern, and attendant and priority calls with a three-burst pattern.

Dual DCP I-Channels

Support the use of dual DCP I-channels for AUDIX networking. In this case, networking refers to the ability to send data files between AUDIX systems, not to communications with the switch.

Emergency Access to the Attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by the system when a user leaves a telephone off-hook, or can be dialed by system users. Emergency access calls can receive priority handling by the attendant.

Enhanced Abbreviated Dialing

Supplements Abbreviated Dialing by providing one enhanced number per system. Enhanced number lists can contain any number or dial access code. System Administrators designate privileges for group number lists, system number lists, and enhanced number lists. With privileged lists, users can access otherwise restricted numbers (for example, stations without long-distance access can be programmed to access specified long-distance numbers.)

For users with 6400-series telephones, Abbreviated Dialing has been enhanced further. Users with display telephones now receive display messages to help them program Abbreviated Dialing, and users can personalize labels for their Abbreviated Dialing softkeys. Users with active speakerphones can also begin programming without having to lift the handset or having to press the speaker button. See *Using the New Abbreviated Dialing Program Feature* document for more information.

Enhanced Night Service

The switch informs the Voice Mail System (VMS) that it is in Night Service, allowing the VMS to perform different actions and call handling for out-of-hours operation. For example, the VMS can be administered to provide recorded announcements after hours. This enhancement was done for the Mode Code Voice Mail Interface.

Enhanced Voice Terminal Display

The Enhanced Voice Terminal Display feature allows users to choose the character set that they want to see on the soft keys and displays of their voice terminals. In addition to the standard Roman character set, users can choose either the Katakana character set or characters used for most European languages.

Extended User Administration of Redirected Calls

Extended User administration of Redirected CAlls allows you to change the lead Call Coverage path or forwarding extension from any on-site or off-site location. For example, you can change the coverage path or forwarding extension from your home office.

External Device Alarming

Allows you to assign analog ports to alarm interfaces for external devices. You can specify a port location, information to identify the external device, and the alarm level to report when a contact closure occurs.

Facility Busy Indication

Allows users of multi-appearance telephones to see which lines, trunk groups, terminating extension groups, hunt groups, or paging zones (called resources or facilities) are busy. When the lamp associated with the resource is lit, the resource is busy.

You can store extension numbers, trunk group access codes, and Loudspeaker Paging access codes in a Facility Busy Indication button. The Facility Busy Indication button provides direct access to any of the facilities.

Facility Test Calls

Allows telephone users to make test calls to access specific trunks, dual tone multifrequency receivers, time slots, and system tones. The user dials an access code and makes the test call to make sure the facility is operating properly. Security measures are included to prevent unauthorized use.

Fiber Link Administration

Port cabinets are connected via direct fiber links or through fiber links to a center-stage switch to provide the connections required for voice and data information transfer. The center-stage switch is composed of switch node carriers that are interconnected by fiber links. The center-stage switch provides both circuit-switched and packet-switched connections. Fiber Link Administration creates the translation data defining these links by identifying the endpoint pairs for each link. Endpoints can be an expansion interface or a switch-node-interface circuit pack.

Go to Cover

Allows users who call another internal extension to send the call directly to coverage.

Group Listen

Simultaneously activates the speakerphone in listen-only mode and the handset or headset in listen-and-speak mode on 6400-series telephones. This allows a user to serve as spokesperson for a group. The user can participate in a conversation while everyone else in the room is listening to what is said.

Group Paging

Allows a user to make an announcement to a group of people via their speakerphones. The speakerphones are automatically turned on when the user begins the announcement. The recipients can listen to the message via the handset if they wish, but they cannot speak to the user in return. Group page members will not receive the page if the member is active on a call appearance, has a call ringing, is off-hook, has “send-all calls” active, or has “do not disturb” active.

Hold

Allows a user to disconnect from a call temporarily, use the telephone for other call purposes, and then return to the original call.

Hold — Automatic

Allows attendants and multi-function telephone users to alternate easily between two or more calls. For example, with automatic hold, selection of a second call appearance automatically puts the active call (if any) on hold and makes the second call appearance active. This feature can be activated on a system-wide basis only. When automatic hold is not activated, pressing a second call appearance would drop the first call.

Hunt Groups

A group of extensions that can handle multiple calls simultaneously to a single phone number. For each call to the phone number, the system hunts for an available extension in the group and connects the call to that extension.

A hunt group is especially useful when you expect a high number of calls to a particular phone number. A hunt group might consist of people trained to handle calls on specific topics. For example, the group might be as follows:

- A benefits department within your company
- A service department for products you sell
- A travel reservations service
- A pool of attendants.

In addition, a hunt group might consist of a group of shared telecommunications facilities. For example, the group might be as follows:

- A group of data-line circuit ports
- A group of data modules.

Individual Attendant Access

Allows a user to call a specific attendant console. Each attendant console can be assigned an individual extension number.

Integrated Services Digital Network — Basic Rate Interface

Enables connection of the system to equipment or end points that support an Integrated Services Digital Network (ISDN) by using a standard format called the Basic Rate Interface (BRI). This feature is a 192 kbps interface that carries two 64 kbps B-channels and one 16 kbps D-channel.

ISDN is a global access standard that uses a layered protocol. It eliminates the need for multiple, separate access arrangements for voice, data, facsimile, and video services and networks. Using the same pair of wires that now carry simple telephone calls, ISDN can deliver voice, data, and video services in a digital format.

The ISDN-BRI Trunk circuit pack allows the system to support the T interface and the S/T interface as defined by ISDN standards (ITU-T recommendation I.411). The circuit pack provides eight ports to the network and supports two B channels and one D channel. An ISDN-BRI Trunk provides the following advantages:

- Provides an inexpensive way to connect to ISDN services provided by the network provider
- Meets almost all ETSI Country protocol requirements
- Supports essential (not supplementary) ISDN services.

Intercept Treatment

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.

Intercom — Automatic

Allows two users to talk together easily. Calling users press the Automatic Intercom button and lift the handset. The called user receives a unique intercom ring and an intercom lamp, if provided, flashes. With this feature, users who frequently call each other can do so by pressing one button instead of dialing an extension number.

Intercom — Dial

Allows multi-appearance telephone users to easily call others within an administered group. The calling user lifts the handset, presses the Dial Intercom button, and dials the one- or two-digit code assigned to the desired party. The called user's phone rings, and an intercom lamp, if provided, flashes. With this feature, a group of users who frequently call each other can do so by pressing one button and dialing a one- or two- digit code instead of dialing an extension number.

Internal Automatic Answer

Allows specific telephones to answer incoming internal calls automatically. This feature is intended for use with telephones that have speakerphones or headsets. A user simply presses an Internal Automatic Answer feature button, and calls are automatically answered when the telephone is idle. Internal calls can be answered using automatic answer, but only attendants can use automatic answer to answer external calls directed to the attendant.

Last Number Dialed

Allows a user to automatically redial the last number dialed. The system saves the first 24 digits of the last number dialed, whether the call attempt was manually dialed or dialed using Abbreviated Dialing. When the user presses the Last Number Dialed button or dials the Last Number dialed feature access code, the system places the call again.

Leave Word Calling

Allows internal system users to leave a short preprogammed message (usually "Call" with the calling user's name, extension number, and the time of the call) for other internal users. When the message is stored on the system, the message lamp on the called telephone automatically lights. Leave Word Calling messages can be retrieved using a telephone display, Voice Message Retrieval, or AUDIX. Messages may be retrieved in English, French, Italian, Spanish, or a user-defined language.

Line Lockout

Removes single-line telephone extension numbers from service when users fail to hang up after receiving dial tone for 10 seconds (default) and then an intercept tone for 30 seconds (default). These intervals are administrable. The out-of-service condition lasts until the telephone user hangs up the phone.

Listed Directory Number

Allows outside callers to access your attendant group in two ways, depending on the type of trunk used for the incoming call. You can allow attendant group access via incoming direct inward dial trunks, or you can allow attendant group access via incoming central office and foreign exchange trunks.

Long Hold Recall Warning

Gives a visual or audible warning on the telephone set that the administered hold time limit has been exceeded for the call on hold. This serves as a reminder if a caller has been put on hold for a long time. The characters “hr” will appear on display telephones.

Loudspeaker Paging Access

Provides attendants and telephone users dial access to voice-paging equipment. As many as nine paging zones can be provided by the system, and one zone can be provided that activates all zones at the same time. A zone is the location of the loudspeakers — for example, conference rooms, warehouses, or storerooms. A user can activate this feature by dialing the trunk access code of the desired paging zone, or the access codes can be entered into Abbreviated Dialing Lists. Once the user has activated this feature, the user simply speaks into the handset to make the announcement.

Deluxe Loudspeaker Paging Access (called Deluxe Paging) provides attendants and telephone users with integrated access to voice-paging equipment and Call Park capabilities. When Deluxe Paging is activated, the call is automatically parked. The parked call returns to the user that parked the call with distinctive alerting when the time-out interval expires.

Malicious Call Trace

Allows a user to trace malicious calls. You define a group of terminal users who can notify others in the group when they receive a malicious call. These users can then retrieve information related to the call. Using this information, the user can identify the malicious call source or provide information to personnel at an adjacent system to complete the trace. It also allows the user to record the malicious call.

Manual Message Waiting

Allows multi-appearance telephone users to light the status lamp associated with the manual Message Waiting button at another multi-appearance telephone. They do this by simply pressing a button on their own telephone. This feature can be administered only to pairs of telephones such as those of a secretary and an executive. The secretary might press the button to signal to the executive that a call needs answering or someone has arrived for an appointment. The executive might use the button to indicate that he or she should not be disturbed.

Manual Originating Line Service

Connects single-line telephone users to the attendant automatically when the user lifts the handset. The attendant number is stored in an Abbreviated Dialing list. When the telephone user lifts the handset, the system automatically routes the call to the attendant using the Hot Line Service feature.

Manual Signaling

Allows one user to signal another user. The receiving user hears a 2-second ring. The signal is sent each time the button is pressed by the signaling user. The meaning of the signal is prearranged between the sender and the receiver. Manual Signaling is denied if the receiving telephone is already ringing from an incoming call.

Message Retrieval

Users can retrieve messages in two ways:

- Display Retrieval — Users having digital telephones with displays or a personal computer integrated with a telephone can display messages.
- Speak-to-Me — Using any touch-tone telephone, users can dial the Speak-to-Me service and hear a synthesized voice read their messages over the telephone.

Misoperation Handling

Defines how calls are handled when a misoperation occurs. A misoperation is when calls are left on hold when the controlling station goes on hook.

For example, a misoperation can occur under either of the following conditions:

- If a user hangs up prior to completing a feature operation (in some cases, hanging up completes the operation, as in call transfer). If, for example, a user places a call on hold, begins to transfer the call, dials an invalid extension number, and then hangs up, that is a misoperation.
- When the system enters night service while attendant consoles have calls on hold.

The system administrator can alter the standard Misoperation Handling to ensure that an external caller is not left on hold indefinitely, or dropped by the system after a misoperation with no way to reach someone for help.

This feature is currently used only in France and Italy.

Multi-Appearance Preselection and Preference

Provides options for placing or answering calls on selected call appearances. Ringing Appearance *Preference* automatically connects a user to the incoming ringing call when the user picks up the handset. *Idle Appearance Preference* automatically connects a user to an idle appearance. *Preselection* allows the user to manually select an appearance. Preselection is used, for example, when a user wants to reconnect with a held call or when a user wants to activate a feature. Preselection can be used with a feature button. For example, if a user presses an Abbreviated Dialing button, the call appearance is automatically selected and, if the user picks up the handset within 5 seconds, the call is automatically placed. The Preselection option overrides both of the other preference options.

Music-on-Hold Access

Automatically provides music, silence, or tone to a caller. Music lets the caller know that the connection is still valid.

Night Service

There are five Night Service features:

- Hunt Group Night Service allows an attendant or a split supervisor to assign a hunt group or a split to Night Service mode. All calls for the hunt group are then redirected to the hunt group's designated Night Service extension. When a user activates Hunt Group Night Service, the associated button lamp lights.
- Night Console Service directs all calls for primary and daytime attendant consoles to a night console. When a user activates Night Console Service, the Night Service button for each attendant lights and all attendant-seeking calls (and calls waiting) in the queue are directed to the night console. To activate and deactivate this feature, the attendant presses the Night button on the principal attendant console or designated console.
- Night Station Service directs incoming calls for the attendant to designated extensions. Attendants can activate Night Station Service by pressing the Night button on the principle console if there is not an active night console. If the night station is busy, calls (including emergency attendant calls) receive a busy tone. The calls do not go into the attendant queue.

- Trunk Answer from Any Station allows telephone users to answer all incoming calls to the attendant when the attendant 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 and a voice-terminal user dials an access code to answer the call.
- Trunk Group Night Service allows an attendant or a designated telephone user to individually assign a trunk group or all trunk groups to the night service mode. Specific trunk groups individually assigned to the service are in Individual Trunk Night Service Mode. Calls coming into these trunk groups are redirected to designated night service extensions. Incoming calls on other trunk groups are processed normally.

Outgoing Call No-Answer (by Call Type)

Disconnects unanswered outgoing calls after a predetermined amount of time. When local, toll, or international calls go unanswered after a certain amount of time, the switch disconnects the call. The caller hears busy tone followed by howler tone. This feature requires that the country code is set for China.

Pass Advice of Charge Information to World Class BRI Endpoints

Provides Advice of Charge (AOC) information to World Class BRI (WCBRI) endpoints. On a call using a WCBRI endpoint, AOC information will be displayed on the endpoint after the call has completed and the far end has hung up.

Personal Station Access

Allows users to transfer their telephone station preferences and permissions to any other compatible telephone. This includes the definition of terminal buttons, abbreviated dial lists, and Class-of-Service and Class-of-Restrictions permissions. It can be used on-site or off-site (with DEFINITY Extender). This has several telecommuting applications. For example, several telecommuting employees can share the same office on different days of the week. The employees can easily and remotely make the shared telephone “theirs” for the day. Remote use requires DEFINITY Extender.

With the hoteling application, this feature is used as a lock and key to prevent unauthorized access.

Personalized Ringing

Allows users of certain telephones to uniquely identify their own calls. Each user can choose one of up to eight possible ringing patterns. The ringing patterns are tone sequences consisting of different combinations of three tones. With this feature, users working closely in the same area can each specify a different ringing pattern to better identify their own calls.

Power Failure Transfer

Provides service to and from the local telephone company central office, including Wide Area Telecommunications System, during a power failure. This allows users to make or answer important or emergency calls during a power failure. This feature is also called Emergency Transfer.

Priority Calling

Allows a user to ring another telephone with a distinctive signal that tells the called party that the incoming call requires immediate attention. The called party can then handle the call accordingly. A user activates priority calling by dialing a Priority Calling access code or by pressing a feature button, followed by the extension number. A user can use Priority Calling only if the telephone has been administered with the required class of service.

Privacy — Attendant Lockout

Prevents an attendant from reentering a multiple-party call that is being held on the console unless the attendant is recalled by one of the parties on the call. This feature is administered on a system-wide basis and is either activated or not activated.

Privacy — Auto Exclusion

With Auto Exclusion active, the Manual Exclusion feature can be automatically activated on a class-of-service (COS) basis. If the COS for Automatic Exclusion is set to yes, then exclusion is automatically activated when you go off hook on a telephone set which has an Exclusion button assigned.

With Automatic Exclusion active, a held exclusion call can be taken off hold by any telephone user with an appearance of the extension that put the call on hold.

Privacy — Manual Exclusion

Allows multi-appearance telephone 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.

Public Network Call Priority

Provides call retention, forced disconnect, intrusion, mode-of-release control, and re-ring to switches on public networks. Different countries frequently refer to these capabilities by different names.

Pull Transfer

Allows either the party who was originally called, or the party to whom the held call will be transferred, to complete the transfer. This is 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. It interfaces with satellite workstations via incoming and outgoing tie trunks and is always available for calls that use those types of trunks.

Recall Signaling

Allows the user of an analog station to place a call on hold, use the voice terminal for other call purposes, and then return to the original call.

Recent Change History

Allows the system manager to view or print a history report of the most recent administration and maintenance changes on the switch. This report may be used for diagnostic or information purposes.

Recorded Announcements

Provides an announcement to callers under a variety of circumstances. For example, announcements let a caller know that his or her call cannot be completed as dialed, that the call is in queue, or that all lines are busy.

A log of changes to recorded announcements allows changes to be tracked by administrative personnel. Users can also list the announcements that are being used for Vector Directory Numbers.

Recorded Telephone Dictation Access

Allows telephone 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. The start/stop function can be voice or dial controlled. Other functions such as initial activation and playback are controlled by additional dial codes.

Remote Access

Permits authorized callers from remote locations to access the system via the public network and then to use the system’s features and services. There are a variety of ways to access the feature. After gaining access, a user hears a system dial tone. For system security, users should be required to dial a barrier code.

Remote Call Coverage

Allows calls to be redirected to a remote location. This allows users to have calls placed to their on-site offices redirected to their home offices. You can administer the system to either monitor calls and bring them back for additional processing (if not answered), or to leave calls at the remote (off-net) location.

Reset Shift Call

Allows calls made to a busy station to hear a special dial tone. When this special dial tone is heard, a single digit may be entered which replaces the last digit of the originally dialed extension. At this point, the call is sent to the new extension. If call coverage is supplied for the dialed extension, the call routes to coverage as normal. If the coverage extension is busy, the special dial tone is heard and a single digit may be entered again. This feature is active for station-to-station calls, not for incoming or outgoing calls. The procedure is basically the same for transferred or conferenced calls.

Ringback Queuing

Places calls in an ordered queue (first-in, first-out) when all trunks are busy. The telephone user who is trying to make a call is automatically called back and hears a distinctive three-burst signal when a trunk becomes available.

Ringer Cutoff

Allows the user of a multi-appearance telephone to turn audible ringing signals on and off. Visual alerting is not affected by this feature. When this feature is enabled, only Priority (three-burst) ring, Redirect Notification, Intercom ring, and manual signaling ring at the telephone. Internal and external calls do not ring.

Ringing — Abbreviated and Delayed

Allows you to manually or automatically assign one of four ring types to each call appearance on a telephone. Whatever treatment you assign to a call appearance is automatically assigned to each of its bridged call appearances.

Security Violation Notification

Allows you to set security-related parameters and to receive notification when the limits that you have established are violated. You can run reports related to both valid and invalid access attempts. You can also automatically disable a login ID or Remote Access authorization that is associated with a security violation. This keeps the system secure from access by that login ID until the problem has been corrected.

Send All Calls

Allows users to temporarily direct all incoming calls to coverage regardless of the assigned call-coverage redirection criteria. Covering users can temporarily remove their voice terminals from the coverage path. The feature is activated and deactivated via a button or access code.

Special Dial Tone

Provides a special dial tone when the following features have been applied to a telephone: Call Forwarding — All Calls, Call Forwarding — Busy/Don't Answer, Send All Calls, and Do Not Disturb. The special dial tone is an indication to the user that they are not going to receive any calls because at least one of these features is active on their telephone.

Station Hunt Before Coverage

This option works with the Suite Check-In feature. With Station Hunt Before Coverage, the call routes to the other phones in the "suite" of rooms before going to coverage, if the primary number called is busy.

Station Hunting

Routes calls made to a busy extension to another extension. To use Station Hunting, you create a station hunting chain that governs the order in which a call routes from one extension to the next when the called extension is busy. Each extension in the chain links *to* only one subsequent extension. However, an extension may be linked *from* any number of extensions.

Station Hunting - Circular

Allows the system to route calls based on how the extensions are administered. When administering a circular station hunt group, the order in which the extensions for those stations assigned to the hunt group are administered is the order in which calls are directed. The system keeps track of the last extension in the hunt group to which a call was connected, such that when the next incoming call arrives, the system can determine the next idle extension in the circular hunt group. Extensions in the hunt group that are busy are skipped and the next idle extension within the hunt group is selected regardless of past call history. The caller hears a busy tone if all extensions in the hunt group are busy and no type of call coverage has been designated.

Station Security Codes

To provide additional security around the customer options, the "init" login has been provided with additional security for the purpose of establishing an authentication procedure for attempts to remotely log in to the system.

Station Self Display

Displays the assigned extension number on a telephone set. A user can dial an FAC or press a previously administered “inspect” button to view the extension number. This is a handy feature if your work environment requires people to sit at different desks from day-to-day. It is also helpful for maintenance personnel that want to verify that the correct extension number was administered.

Telephone Self Administration

Allows users to program feature buttons on 6400-series telephones.

Temporary Bridged Appearance

Allows multi-appearance telephone users in a terminating extension group or personal central office line group to bridge on to an existing group call. If a call has been answered using the Call Pickup feature, the originally-called party can bridge on to the call. This feature also allows a called party to bridge on to a call that redirects to coverage before the called party can answer it.

Terminal Translation Initialization

Allows you to merge an Administration Without Hardware station to a valid port from a terminal connected to that port. You simply dial a system-wide security code and the extension. This feature also allows you to separate a station from its port by dialing a similar separate digit sequence. This action causes the station to be administered without hardware.

Terminating Extension Group

Allows an incoming call to ring (either audible or silent alerting) as many as four telephones at one time. Any user in the group can answer the call. Any telephone can be administered as a group member. However, only a multi-appearance telephone can be assigned a feature button with an associated status lamp. The feature button allows the user to select a Terminating Extension Group call appearance for answering or bridging onto an existing call but not for call origination. For example, a department in a large store might have three telephones. Anyone in the department can answer the call. The salesperson most qualified to answer the call can then bridge onto the call.

Time Supervision and Forced Release

Provides a forced disconnect on certain call types. When any of the following timers expire during an outgoing call attempt, the switch applies busy tone followed by howler tone:

- Pre-dialing and interdigit timer
- Outgoing seizure acknowledge timer
- Answer supervision timer
- 60-, 90-, and 120-second no-answer disconnect timers, based on ARS call type
- 120-second timer used for calls without a call type, such as calls to trunk access codes.

Timed Reminder and Attendant Timers

Automatically alerts the attendant after an administered time interval for the following types of calls: extended calls to be answered or waiting to be connected to a busy single-line telephone, one-party calls placed on hold on the console, and transferred calls that have not been answered after transfer. 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. The system supports a variety of administrable attendant timers for use in a variety of situations.

Transfer

Allows telephone users to transfer active or held trunk or internal calls to other telephones within the system without attendant assistance. This feature provides a convenient way to connect a party to someone better qualified to handle the call. Single-line telephone users momentarily flash the switchhook or press the Recall button, dial the desired extension, and hang up. Multi-appearance telephone users press the Transfer button, dial the desired extension number, and either press the Transfer button again or hang up. If the telephone has a display, a message confirming the transfer is displayed to the user.

Transfer Abort

Transfer Abort allows you to abort a transfer when you select another call appearance in the middle of the transfer operation or hang up. If you decide to cancel the transfer for any reason (that is, you get an important call, you dialed the wrong number), all you need do is select another call appearance or hang up. The original call you were transferring is put on hold, and you can then go back to that call at your convenience and re-initiate the transfer.

NOTE:
You cannot have the Transfer Upon Hangup feature enabled if you want to abort the transfer when hanging up.

This feature applies to DCP, Hybrid, ISDN-BRI and wireless telephones, but not to analog telephones.

Transfer — Outgoing Trunk to Outgoing Trunk

Allows a user or attendant to initiate two or more outgoing trunk calls and then transfer the trunks together. The transfer operation removes the original user from the connection and conferences the outgoing trunks. Alternatively, the controlling party can establish a conference call with the outgoing trunks and then drop out of the conference, leaving only the outgoing trunks on the conference. This is an optional enhancement to Trunk-to-Trunk Transfer and requires careful administration and use.

Transfer Recall

Redirects a transferred call (with priority ringback) back to the person that transferred the call only if the call is not answered by the person to which the call was transferred to or if the call is not answered within the administered timer level.

Trunk Flash

Allows a feature or function button on a multifunction telephone or attendant console to be assigned as a Flash button. Pressing this button while connected to a trunk (which must have been administered to allow trunk flash) causes the system to send a flash signal over the connected trunk.

Trunk Flash enables multifunction voice terminals to access central office customized services that are provided by the central office to which the system is connected. These services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the system station on an active trunk call. The Trunk Flash feature can help to reduce the number of trunk lines connected to the system.

Trunk Identification by Attendant

Allows an attendant or display-equipped telephone 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 telephone. This feature is particularly helpful for identifying a faulty trunk. That trunk can then be removed from service and the problem quickly corrected.

Trunk-to-Trunk Transfer

Allows the attendant or telephone user to connect an incoming trunk call to an outgoing trunk call. This feature 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. 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.

Visually Impaired Attendant Service

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. This feature defines six new attendant buttons to aid visually impaired attendants:

- Visually Impaired Service Activation/Deactivation button: activates or deactivates the feature. All ringers previously disabled (for example, recall and incoming calls) become reenabled.
- Console Status button: voices 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: voices what is shown on the console display. VIAS support is not available for all display features (for example, class-of-restriction information, personal names, and some call purposes).
- Last Operation button: voices the last operation performed.
- Last Voiced Message button: repeats the last voiced message.
- Direct Trunk Group Selection Status button: voices the status of an attendant-monitored trunk group.

The visually impaired attendant may use the Inspect mode to locate each button and determine the feature assigned to each without actually executing the feature.

Voice Message Retrieval

Allows telephone users, Remote Access users, and attendants to retrieve Leave Word Calling and Call Coverage voice messages. It can be used to retrieve a user's own messages or messages for another user. However, a different user's messages can be retrieved only by a user at a telephone or attendant console in the coverage path, by an administered system-wide message retriever, or by a Remote Access user when the extension and associated security code are known. The system restricts unauthorized users from retrieving messages.

Voice Messaging and Call Coverage

Often an AUDIX system is set up as the last point in a Call Coverage path. An assistant or colleague who answers a redirected call intended for a user can also transfer the caller to the user's AUDIX mailbox. The caller may prefer to leave a voice message as opposed to a written message.

Many other options are available. For example, a caller can redirect a call from the AUDIX system to an attendant. Or, the caller can transfer to another extension instead of leaving a message. The AUDIX automated attendant can answer all calls to a company, and can then send the calls to various extension numbers; with this feature, callers are instructed to enter keypad commands to redirect the call.

Voice Terminal Ringing Options

Provides multi-appearance telephone users with different ringing patterns. This feature primarily affects audible ringing for calls directed to telephones that are off-hook, or calls directed to idle and active CALLMASTER telephones.

Voice Terminal Display

Provides multi-appearance telephone users with updated call and message information. This information is displayed on a display-equipped telephone. The information displayed depends on the display mode selected by the user. Information that allows personalized call answering is available on many calls.

Users may select any of the following as the display message language: English (default), French, Italian, or Spanish. In addition, messages can be administered on the system in a fifth language. The language for display messages is selected by each user.

Whisper Page

Allows an assistant or colleague to bridge onto a telephone conversation and give a user a message without being heard by the other party or parties on the call. This feature operates on DCP and BRI telephones.

World Class Tone Detection

Enables the system to identify and handle different types of call progress tones, depending on the system administration. You can use the tone detector and identification to display on Data Terminal Dialing and to decide when to send digits on trunk calls through Abbreviated Dialing, Automatic Route Selection, Automatic Alternate Routing, and Data Terminal Dialing. Tone detect modes are as follows:

- “Tone detect mode 1” designates countries that use the same tone plan as Italy.
- “Tone detect mode 2” designates countries that use the same tone plan as Australia.
- “Tone detect mode 3” designates countries that use the same tone plan as the United Kingdom.
- “Tone detect mode 4” designates countries that use dial tones between 345 Hz and 625 Hz.
- “Tone detect mode 5” designates countries that use dial tones between 345 Hz and 1190 Hz.
- The “level of tone detection precise” is used in countries that, except for the continuous dial tone and discontinuous other tone, have tones with characteristics that do not match those expected by the tone detector circuit pack’s detect mode.
- The “level of tone detection broadband” is used in countries that have a discontinuous dial tone.

World Class Tone Generation

Allows you to define call-progress tones. You can select values for frequency and cadence. If you do not define a call-progress tone, the system sends silence.

Hospitality features

The following features are designed for use in the hospitality industry. However, other features listed elsewhere may be of use in this industry. The Attendant Crisis Alert feature, for example, described in the Basic Features section of this appendix, is primarily used in lodging establishments. That feature is listed as a basic feature because it is available on any system that has the appropriate attendant console.

Attendant Backup

The Attendant Backup feature allows you to access most attendant console features from one or more specially-administered backup telephones. This allows you to answer calls more promptly, thus providing better service to your guests and prospective clients.

When the attendant console is busy, you can answer overflow calls from the backup telephones by pressing a button or dialing a feature access code. You can then process the calls as if you are at the attendant console. The recommended backup telephones are the Lucent Technologies Models 6408, 6416, or 6424.

Attendant Room Status

Allows an attendant to see whether a room is vacant or occupied and what the housekeeping status of each room is. This feature is available only when you have Enhanced Hospitality enabled for your system. This feature combines the property management capabilities of Check-In/Check-Out and Housekeeping Status, but does not require that you have a Property Management System.

Automatic Selection of Direct Inward Dialing Numbers for Guest Rooms

This feature allows the system to automatically choose a number from a list of available Direct Inward Dialing (DID) numbers that will be assigned to a guest's room extension when checking in.

With this feature, hotels can give a guest a phone number that is different from their room number, thereby protecting the guest's privacy. When a particular DID number is called, the call routes to the guest's room extension, and covers as if the room was called directly. Besides improving guest security, this eliminates the need for an attendant or front desk staff to extend a call to a guest room.

Automatic Wakeup

Allows attendants, front desk users, and guests to request that one or two wake-up calls be placed automatically to a certain extension number at a later time. When a wake-up call is placed and answered, the system can provide a recorded announcement (which can be a speech synthesis announcement), music, or simply silence. With the Integrated Announcement feature, multiple announcements enable international guests to use wake-up announcements in a variety of languages. See also Daily Wakeup, Dual Wakeup, and VIP Wakeup.

Check-In/Check-Out

Allows front desk personnel to check guests into a hotel and, when the guests leave, check them out. There are two ways this is done: through the PMS terminal or through the attendant console (or backup voice terminal). Check-in and check-out from the attendant console should be used only if there is no PMS, or if the link to the PMS is down. If the PMS is installed and working, check guests using the PMS.

For guest check-in or check-out from the console, there are two buttons on the attendant console (or backup voice terminal): one labeled Check In and the other labeled Check Out. The check-in procedure performs two functions: it deactivates the restriction on the telephone in the room allowing outward calls, and it changes the status of the room to occupied.

Controlled Restrictions

Allows an attendant or telephone user with console permission to activate and deactivate the following restrictions for an individual telephone or a group of telephones: outward/toll, total, station-to-station/toll, and termination restrictions. This feature is available in a non-hospitality environment, but is used extensively in hospitality offers.

Daily Wakeup

Allows a guest or front desk personnel to schedule a single wakeup request for a daily wakeup call. For example, if a guest needs to receive a wakeup call at 5:30 a.m. for the duration of his or her stay, one request can be placed on the system instead of placing a separate request for each day.

Dial by Name

The Dial-by-Name feature allows callers to the system to access guest rooms simply by dialing the name of the guest they are trying to contact. This feature uses recorded announcements and the Call Vectoring feature to set up an automatic attendant procedure. This automatic attendant procedure gives callers the ability to enter a guest's name. When a single or unique match is found, the call is redirected to the guest's telephone.

Do Not Disturb

Allows guests, attendants, and authorized front desk users to request that no calls, other than priority calls, be connected to a particular extension until a specified time.

Dual Wakeup

Allows guests to have two separate wakeup calls. The Dual Wakeup feature is an enhancement to the standard Automatic Wakeup feature used in hospitality environments. With the standard wakeup feature, guests or front desk personnel can create one wakeup call per extension. The Dual Wakeup feature allows guests and front desk personnel to create either one or two wakeup calls. The Dual Wakeup feature for guests is valid only when the system is not equipped with a speech synthesizer circuit pack (TN725B).

Housekeeping Status

Records the status for up to six housekeeping codes and reports them to the property management system. These status codes are usually entered by the housekeeping staff from the guest room or from a designated telephone, but they can also be updated by the front office personnel using the attendant console or a backup voice terminal. Six status codes can be used from guest rooms, and four status codes can be used from telephones that do not have the client room Class of Service.

Names Registration

Automatically sends a guest's name, room extension, and coverage path from the Property Management System to the switch at check-in, and automatically removes this information at check-out. The information may be displayed on any attendant console or display-equipped telephone at various hotel locations (for example, Room Service, or Security).

Property Management System Digit to Insert/Delete

Many customer configurations base the room telephone extension on the room number by adding an extra leading digit. The PMS Insert/Delete Digit feature allows users to delete the leading digit of the extension in messages. The feature is useful for a hotel that has multiple extensions sharing an extra leading digit in front of the room number. The leading digit is automatically inserted when the message goes to the PBX.

NOTE:
The PMS interface supports three-, four-, or five-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, do not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Property Management System Interface

The Property Management System allows a customer to control features used in both a hospital-type and a hotel/motel-type (GuestWorks) environment. The communications link allows the Property Management System to interrogate the system and allows information to be passed between the system and the Property Management System. GuestWorks exchanges guest status information (room number, call coverage path, and other data) with the property management system.

There are two ways that the guest data can be encoded:

- Using a combination of Binary Coded Decimal (BCD) encoding and the ASCII character set
- Using only the ASCII character set.

Single-Digit Dialing and Mixed Station Numbering

Allows hotel staff and guests easy access to internal hotel/motel services and provides the capability to associate room numbers with guest room telephones. The feature provides the following dial plan types: single-digit dialing, prefixed extensions, and mixed numbering.

Suite Check-In

This feature provides the capability to have the system automatically check-in several related extensions with one check-in command. This feature allows hotels that offer “suite” rooms with several phones the ability to check in all the phones associated with that “suite” at one time.

VIP Wakeup

Allows front desk personnel to provide personalized wakeup calls to important guests. When a wakeup call has been scheduled for an important guest, a wakeup reminder call is placed to the front desk personnel, who in turn call the guest personally to provide the wakeup call.

Wake-Up Activation via Confirmation Tones

If a speech synthesizer circuit pack is not installed, guests can still enter their own wake-up calls (two if the Dual Wakeup feature is active). The guests do not receive voice prompts as they would using the speech synthesizer circuit pack; guests will receive call progress tones (recall dial tone and confirmation tone) to set up their wake-up calls.

Hunt Group features

The system offers the following features designed to help you set up and maintain hunt groups (splits) and agents.

Abandoned Call Search

Allows a central office that does not provide timely disconnect supervision to identify abandoned calls. An abandoned call is one in which the calling party hangs up before the call is answered. Abandoned Call Search is suitable only for older central offices that do not provide timely disconnect supervision.

Agent Call Handling

Allows you to administer functions that Automatic Call Distribution agents use when handling incoming calls. You define specific agent capabilities and can plan capacities based on those capabilities.

Attendant Vectoring

With Attendant Vectoring, a highly flexible approach for managing incoming calls to an attendant is available. For example, with current night service operation, calls redirected from the attendant console to a night station can only ring at that station and will not follow any coverage path. With Attendant Vectoring, night service calls will follow the coverage path of the night station. The coverage path could go to another station, and then eventually to a voice mail system. The caller can then leave a message that can be retrieved and acted upon.

Auto-Available Split

Allows agents of an ACD split to be in Auto-In work mode continuously. An agent in Auto-In work mode becomes available for another ACD call immediately after disconnecting from an ACD call. You can use this feature to bring ACD agents back into Auto-In work mode after a system restart. Although not restricted to such, this feature is intended to be used for splits containing only recorders or voice-response units.

Automatic Call Distribution

Allows incoming calls to connect automatically to specific splits. An Automatic Call Distribution (ACD) split is designed to receive a high volume of similar calls. Calls to specific splits are automatically distributed among the agents assigned to that split. Calls queue to the split until an agent is available. You can assign a supervisor to each split. The split supervisor can listen in on agent calls, monitor the split queue status, and assist agents. If you have Basic Call Management System, you can measure and create reports on the status of ACD agents, splits, and trunks.

Basic Call Management System

Provides real-time and historical reports to assist you in managing agents, ACD splits, VDNs, and trunk groups. You can display reports on the Management Terminal or print them. In addition, you can schedule historical reports to print automatically on the system printer.

The switch supports a maximum of 150 logged-in ACD agents. Of those 150 agents, a maximum of 25 agents can be measured by BCMS. However, measurements are collected on a per-hunt-group basis, not a per-agent basis, and up to five hunt groups can be designated for measurement by BCMS. This means that whether you designate one hunt group or five hunt groups as being internally-measured by BCMS, no more than 25 agents can log in to those hunt groups at any one time. The other 125 agents can log in to other hunt groups as long as they are not designated for measurement by BCMS.

Call Prompting

Allows the system to collect information from the calling party and to direct the calls via Call Vectoring. The caller is verbally prompted by the system and enters information in response to the prompts. This information is then used to redirect the call or to handle the call in some other way (taking a message, for example). This feature is mostly used to enhance the efficient handling of calls in the Automatic Call Distribution application.

Call Vectoring

Processes incoming and internal calls according to a programmed set of commands. Vector commands may direct calls to on-premise or off-premise destinations, to any split, or to a specific call treatment such as: an announcement, forced disconnect, forced busy, or delay treatment. For example, the system can collect digits from the user via Call Prompting and can then route calls to a destination specified by those digits. There are many different applications of the Call Vectoring feature; however, Call Vectoring is primarily used to handle the call activity of Automatic Call Distribution splits.

Dialed Number Identification Service

Displays, for a called party or answering position, the service or product associated with an incoming call.

Intraflow and Interflow

Allow you to redirect ACD calls from one split to another split. Intraflow redirects calls to other splits within the system using Call Coverage or Call Forwarding All Calls. Interflow redirects calls to an external split or location using Call Forwarding All Calls. You can have calls redirected from one split to another conditionally, according to the coverage path's redirection criteria. For example, you can define a split's coverage path to automatically redirect incoming ACD calls to another split when a terminal is busy or unanswered.

NOTE:
This feature is not related to the Look-Ahead Interflow feature. The Look-Ahead Interflow feature is not supported on DEFINITY BCS nor GuestWorks.

Multiple Call Handling on Request

Allows agents to receive an ACD call while other types of calls are alerting, active, or on hold.

Queue Status Indications

Allows you to assign queue-status indicators for Automatic Call Distribution calls based on the number of calls queued and the length of time the calls have been in queue. You can assign these indications to lamps on agent, supervisor, or attendant consoles, or telephones to help monitor queue activity. In addition, you can define auxiliary queue warning lamps to track queue status. On display telephones, you can display the number of calls queued and the time in queue of a split's oldest call.

Redirection on No Answer

Redirects a ringing ACD call or Direct Agent Call after an administered number of rings. This prevents an unanswered call from ringing indefinitely. The call can redirect either to the split to be answered by another agent or to a vector directory number (VDN) for alternative call handling. Direct Agent Calls route to the agent's coverage path, or to a VDN if no coverage path is administered. You must have ACD enabled to use this feature.

Service Observing

Allows a specified user, such as a supervisor, to observe or monitor another user's calls. Observers can observe in listen-only or listen-and-talk mode using a feature button on their telephone. You set up Service Observing to observe a particular extension, not to observe all calls to all extensions at a telephone.

NOTE:
Service Observing may be subject to federal, state, or local laws, rules, or regulations or may require the consent of one or both of the call parties. Familiarize yourself and comply with all applicable laws, rules, and regulations before using this feature.

VDN in a Coverage Path

Enhances Call Coverage and Call Vectoring to allow you to assign vector directory numbers (VDNs) as the last point in coverage paths. Calls that go to coverage can be processed by vectoring/prompting to extend Call Coverage treatments.

Private Networking features

The great expandability of the system makes it a logical choice for setting up private networks. Consequently, the system includes many private networking features.

Centralized Voice Mail via Interswitch Mode Codes

Provides the capability to share a voice mail system among several switches using the Mode Code - Voice Mail System Interface. This feature provides a cost effective choice for multiple sites by eliminating the need for a voice mail system at each site.

A Centralized Voice Mail network can consist of DEFINITY BCS Issue 6 or later, DEFINITY ECS R8 or later, ProLogix™ R3 or later, Merlin Legend® R6.1 or later, and Merlin Magix™ systems. Either a DEFINITY BCS, ProLogix, or DEFINITY ECS switch must be the host for the voice mail system, and UDP and ISDN-PRI software are required at each site. See *DEFINITY ECS Administration for Network Connectivity* and *Merlin Legend Network Reference* for more information.

Extended Trunk Access

Used with Uniform Dial Plan, allows the system to send any unrecognized number (such as an extension not administered locally) to another system for analysis and routing. Such unrecognized numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the Uniform Dial Plan table. Non-Uniform Dial

Plan numbers are administered on either the First Digit Table (on the Dial Plan Record form) or the Second Digit Table. They also are not administered on the Extended Trunk Access Call Screening Table. Extended Trunk Access helps you make full use of automatic routing and Uniform Dial Plan.

Inter-PBX Attendant Service

Allows attendants for multiple locations to be concentrated at one location. Incoming trunk calls to the unattended location, as well as attendant-seeking calls, route over tie trunks to the main location.

Japanese National Private Networking Support

Provides support for Japanese private ISDN networks. The Japanese private network ISDN protocol is different from the standard ISDN protocol. The switch will now support extensions to the ISDN protocol for switches using the Japanese country code.

Private Network Access

Allows calls to other systems in a private network. These calls do not use the public network. They are routed over your dedicated facilities.

QSIG Basic

Provides compliance to the International Organization for Standardization (ISO) ISDN-PRI private-networking specifications. QSIG is defined by ISO as the worldwide standard for private networks.

QSIG is the generic name for a family of signaling protocols. The Q-reference point or interface is the logical point where signaling is passed between two peer entities in a private network. QSIG signaling can provide feature transparency in a single-vendor or multi-vendor environment.

The system provides QSIG Basic Call Setup with Number Identification and Transit Counter.

Number Identification

Allows a switch to send and receive the calling number. Additional parameters that control the display of the connected number are administered on the Feature-Related System Parameters form. QSIG Number Identification displays up to 15 digits for the calling and connected numbers across ISDN-PRI interfaces.

Transit Counter

The system provides QSIG Transit Counter as defined in ISO/IEC 6B032 and 6B033. It prevents indefinite looping, connections giving poor transmission performance, and inefficient use of network resources. This feature is invoked automatically for ISDN-PRI basic calls.

Uniform Dial Plan

Provides a common four- or five-digit dial plan that can be shared among a group of switches. Both interswitch and intraswitch dialing require four- or five-digit dialing. This feature is used with an electronic tandem network (ETN).

Trunk Group features

The system supports a variety of interfaces to voice and data networks. Trunks supply links between the system, the public network, and other systems. Digital Signal Level 1 interfaces offer high-speed digital connectivity between systems. For a complete listing of the trunk and line interfaces available in different countries, see the *DEFINITY® ECS System Description*.

Automatic TEI

The user side will support automatic TEI assignment by the network. Both fixed and automatic TEI assignment are supported on the network side.

BRI Trunk Service

Supports public-network access outside the U.S. on point-to-midpoint connections, with the restriction that the system must not be configured in a passive bus arrangement with other BRI endpoints. It will also support the use of ISDN-BRI trunks as inter-PBX tie lines using the QSIG peer protocol. The system supports a two-wire U interface, the four-wire T interface, and the U.S. National ISDN protocol.

Call-by-Call Service Selection

Enables a single ISDN-PRI trunk group to carry calls to a variety of services, rather than requiring each trunk group to be dedicated to a specific service. It allows you to set up various voice and data services and features for a particular call.

CAMA - E911 Trunk Group

Sends Caller's Emergency Service Identification (CESID) information over Centralized Automatic Message Accounting (CAMA) trunks to the local community's Enhanced 911 (E911) system through the local Central Office. The information sent can be one of the following:

- Each extension on the switch can send precise location information.
- Groups of extensions can send the same location information; for example, one number for Building A, one number for Building B, and so on.
- If there are no CAMA trunks, the system sends the same location information for all extensions at the site.

DS1 Trunk Service

Bit-oriented signaling that multiplexes 24 channels into a single 1.544 Mbps stream. DS1 can be used for voice or voice-grade data and for data-transmission protocols. E1 trunk service is bit-oriented signaling that multiplexes 32 channels into a single 2.048 Mbps stream. Both DS1 and E1 provide a digital interface for trunk groups.

E&M Signaling — Continuous and Pulsed

Provides continuous and pulsed E&M signaling. Continuous and pulsed E&M signaling is a modification to the E&M signaling used in the United States. Continuous E&M signaling is intended for use in Brazil, but can also be used in Hungary. Pulsed E&M signaling is intended for use in Brazil.

ETSI Functionality

The full set of ETSI public-network and private-network ISDN features is officially supported. This includes Look-ahead Routing and Usage Allocation. It does not include Non-Facility Associated Signaling or D-Channel Backup.

Facility and Non-Facility Associated Signaling

Allows an ISDN-PRI DS1/E1 interface D-channel to carry signaling information for B-channels (voice or data). D-Channel Backup can also be administered to increase system reliability.

ICLID on Analog CO Trunk

This feature displays the calling party name and number when that information is provided by the Central Office over the TN429D CO Trunk (analog, loop start) circuit pack. Display of name and number will work with all digital voice terminals (DCP and BRI) equipped with 32-character or 40-character alphanumeric displays.

Countries supported by this feature are the United States and Japan. Name and calling number are available from United States central offices; only calling number is available from central offices in Japan. This feature may also be used in other countries that comply with either United States or Japan requirements. For more information on feature compatibility, see the *DEFINITY® ECS System Description*.

IP Trunks

IP trunks allow you to route voice and fax calls over Internet Protocol (IP) networks such as the Internet and private intranets, reducing long-distance charges and giving you added flexibility in routing traffic between sites. Both the originating and destination switches must have the DEFINITY Internet Protocol Trunk (DEFINITY IP Trunk) application or Lucent's Internet Telephony Server-Enterprise (ITS-E) Release 1.2. The DEFINITY IP Trunk feature consists of the following components:

- An IP Trunk circuit pack, which contains a Windows NT server
- The DEFINITY IP Trunk software, which routes telephone calls and faxes over the Internet or your company's intranet
- Configuration Manager software, which lets you administer the operation and performance of DEFINITY IP Trunk service.

Both the IP trunk software and Configuration Manager reside on the Windows NT server on the IP Trunk circuit pack. For information about Internet Telephony Server-Enterprise, contact your Lucent Technologies representative.

NOTE:
DEFINITY BCS and GuestWorks do not support the full IP Solutions feature of the DEFINITY ECS, only IP trunks.

ISDN — General

Gives you access to a variety of public and private network services and facilities. The ISDN standard consists of layers 1, 2, and 3 of the Open System Interconnect (OSI) model. The system can be connected to ISDN using standard frame formats: Basic Rate Interface (BRI) and the Primary Rate Interface (PRI).

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, 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.

ISDN Restriction Presentation

Restricts the display of calling/connected numbers over ISDN trunks. ISDN trunk groups can be administered to control the display of calling/connected numbers. Each trunk group can be administered to display “Presentation restricted,” “Number no available due to networking,” or an administered text string instead of the calling/connected number.

Layer 1 Deactivation

Tells call processing and maintenance software whether to expect the network to drop Layer 1 when the BRI port is idle. When acting as the TE side, the switch supports the case where the network deactivates Layer 1 when both B-channels of a BRI port are idle. When acting as the NT side, the switch deactivates Layer 1 only when the BRI port is busied out.

Multiple Public Network Calling/Connected Numbers/System

Allows multiple calling/connected numbers to be administered for trunks associated with different network providers. This ensures that the proper calling/connected numbers are sent out based on the trunks used.

Multiple Subscriber Number - Limited

Lets customers assign multiple extensions to a single BRI endpoint. The Multiple Subscriber Number (MSN) feature works with BRI end points that allow the Channel ID information element to be encoded as “preferred.”

NT Interface on TN556C

Support for the NT (network) side of the T interface has been added using the TN556C circuit pack. This gives full tie trunk capability using BRI trunks. The system supports leased BRI connections through the public network, with a TN2185 circuit pack on each end of the leased connection. The system will not, however, allow customers to administer both the endpoints and the trunks on the same TN556C circuit pack.

NT QSIG Peer Protocol

The NT side of the QSIG Peer Protocol is available on the switch.

Dial by Name

The Dial-by-Name feature allows a caller to “dial” someone by entering that person’s name from the caller’s touch-tone keypad. This feature is accessible by using the Call Vectoring feature and the integrated announcement circuit pack (TN750C) to create an “auto-attendant” procedure in which one of the options allows callers to enter a person’s name instead of the person’s extension number. The system processes the name characters received, and, when if a single match is found, the number is dialed automatically.

NOTE:
Not all features are available with each model of the system. Please see the *DEFINITY® ECS System Description* for information on feature availability by model. In addition, not all system applications or adjunct applications may be available in all countries.

A typical scenario might be as follows:

- When a call comes in to the system (usually to a Listed Directory Number), a vector routes the call to an announcement that says, “Hello. You have reached A1 Hotel. Please press 0 for the operator, press 1 for the front desk, press 2 if you know the guest’s extension, press 3 if you know the guest’s name, press 4 if you want to choose from a list of extensions, or press 5 if you wish to hear these options again.”
- When the caller selects 3, the caller is then instructed to enter the person’s name.
- As soon as a single match is found, the call is placed to that person.

The database for the names used in this feature comes from names entered into a management terminal or from names entered into a property management system (PMS) terminal, which are then communicated to the system during a database update.

User operation

The user operations are as follows:

NOTE:
This feature is not accessible from rotary telephones or telephones that do not have a labeled dial keypad. This feature operates using the Roman alphabet only.

1. Dial the published directory number.
   - The call is routed to the auto-attendant procedure.
2. Listen to the recorded announcements, and select the option that allows you to enter a name.
   - You are prompted to enter the person's name.
3. Enter the first four characters of the person's last name.
   - If only one name matches the four characters entered, the call is placed to that person.
   - If there is more than one match, continue with Step 4.
   - If there are not matches, continue with Step 6.
4. If there is more than one match for the first four characters, you are prompted to enter the rest of the characters in the person's last name. After you enter the rest of the characters, press the [#] key.
   - If only one name matches the characters entered, the call is placed to that person.
   - If there is more than one match, continue with Step 5.
   - If there are not matches, continue with Step 6.
5. If there are still multiple matches, you are prompted to enter the first two characters of the person's first name.
   - If only one name matches the characters entered, the call is placed to that person.
   - Otherwise, the call cannot be completed using Dial by Name. Continue with Step 6.
6. You can dial [1] and try entering the name again, or you can dial [0] and the call is routed to a designated extension (usually the attendant or a voice mailbox).
   - If routed to an attendant, the attendant can then attempt to connect the call.
   - If routed to a voice mailbox, the caller can leave a message.

Considerations

Consider the following when implementing the Dial-by-Name feature:

- The names used for this feature cannot have any accent marks or be characters other than the Roman alphabet. If non-Roman characters must be entered, the logical equivalent should be used in the names database.
- Special characters, such as the dash (-) and the apostrophe (‘), are ignored if entered into the names’ database when it comes to using the Dial-by-Name feature. For example, when searching on the name O’Neill, a user should enter “onei” for the initial search. The * key can be entered to represent a dash or apostrophe, but the users must be aware that special characters are an option.
- Special characters, such as the pound sign (#), the asterisk (*), and numbers 0-9, cause names in the database to be unsearchable. That is, if a name in the database has any of these characters, a user cannot search on that name.
- If a person’s last name is less than four characters long, the caller must press the # key to signify end-of-dialing. This instruction should be part of the recorded announcement.
- The system supports a maximum length of 15-character names (last name, first name). If the last name is longer than 15 characters, the first 15 characters should be entered. If two or more people have the same last name and that name is 15 characters long (or longer), the Dial-by-Name feature cannot be used to dial those persons.
- There are no “canned” announcements recorded on the announcement circuit pack. All of the announcements for the Dial-by-Name feature must be customized on-site.
- This feature may provide a security issue for some industries (such as a hotel or a hospital). If there are people that should not be accessible using Dial-by-Name, then those people’s names can be entered into the database using a numerical digit at the beginning of the last name (such as 9Carrier). This can be done only if the property management system will allow non-alphabetic characters at the beginning of names.

Administration

change system-parameters special-applications

- On Page 2 of this form, enter **y** in the `Dial by Name` field.

add vdn XXXX (XXXX is the extension number)

- Use this form to specify which vector directory number (VDN) callers will access when the VDN is dialed. The number used to support the Dial-by-Name feature is usually the published telephone number for the company. You can make this number accessible for outside callers, guests within the hotel, and employees.

change vector X

- You can assign several vectors that define how calls will be handled as users select the different prompts. The following example shows an “auto-attendant” procedure that can be used to access the Dial-by-Name feature. Step numbers 1-20 contain the basic auto-attendant steps, and Steps 21-32 contain the Dial-by-Name steps. Contact Lucent Technologies or your authorized dealer for support in setting up your procedures.

```
change vector 2                                    Page   1 of   3
                              CALL VECTOR

Procedure: 2                 Name Dial by Name


01 wait-time     2   secs hearing ringback
02 collect       1  digits after announcement 381
03
04 route-to      number 0              with cov n if digit             =  0
05 route-to      number 105            with cov n if digit             =  1
06 goto          step   12  if digits          =      2
07 goto          step   21  if digits          =      3
08 goto          step   19  if digits          =      4
09 goto          step   16  if digits          =      5
10 route-to      number 0              with cov n if unconditionally
11
```

```
change vector 2                                    Page   2 of   3
                              CALL VECTOR


12 collect       3  digits after announcement 382
13 route-to      digits with coverage y
14 route-to      number 0              with cov n if unconditionally
15
16 goto          step   2   if unconditionally
17
18
19 collect       3  digits after announcement 383
20 goto          step   13  if unconditionally
21 collect       4 digits after announcement 661
22 route-to      name1 with coverage y
```

```
change vector 2                                     Page   3 of   3
                              CALL VECTOR


23 goto            step 30 if nomatch
24 collect         11 digits after announcement 662
25 route-to        name2 with coverage y
26 goto            step 30 if nomatch
27 collect         2 digits after announcement 663
28 route-to        name3 with coverage y
29 goto            step 30 if nomatch
30 collect         1 digits after announcement 660
31 goto            step 21 if digits = 1
32 route-to        number 0                 with cov n if unconditionally
```

This example does the following:

1. When someone calls the system, the caller receives ringback for 2 seconds.
2. Announcement 381 plays. This announcement asks the caller to do one of the following:
   - Press [0] or wait if the caller wants the operator; if the caller presses [0] or waits for the timeout, the call is routed to the operator.
   - Press [1] if the caller wants the front desk; if the caller presses [1], the call is routed to extension 105, which is the front desk.
   - Press [2] if the caller knows the person's extension; if the caller presses [2], the call is routed to announcement 382, which instructs the caller to dial the person's extension.
   - Press [3] if the caller knows the person's name; if the caller presses [3], the following sub-procedure occurs:
     a. Announcement 661 plays requesting that the caller enter the first four characters of the person's last name.
        — If there is a single match, the call is redirected.
        — If there are multiple matches, continue with Step b..
        — If there is no match, go to Step d..
     b. Announcement 662 plays requesting that the caller enter the rest of the person's last name, followed by the [#] key.
        — If there is a single match, the call is redirected.
        — If there are multiple matches, continue with Step c..
        — If there is no match, go to Step d..

c. Announcement 663 plays requesting that the caller enter the first two characters of the person's first name.

— If there is a single match, the call is redirected.

— If there is no match, continue with Step d..

d. Since there are still no matches, announcement 660 plays telling the caller that he or she can press 1 to try again, or press 0 to get an operator.

- Press 4 if the caller knows the department he or she wish to access (such as housekeeping); if the caller presses 4, the call is routed to announcement 383, which gives the caller a list of several departments that the caller can dial directly.
- Press 5 to start over again; if the caller presses 5, the caller hears announcement 381, which repeats all of the options.
- If the caller dials anything else, the call is routed to the operator.

Required hardware

The integrated announcement circuit pack (TN750C) is required for this feature.

Features Not Supported

B

The following DEFINITY ECS features and adjuncts are not supported on the DEFINITY BCS and GuestWorks offers:

- Adjunct Switch Application Interface (ASAI)
- Asynchronous Transfer Mode
- Basic Call Management System Enhancements
  - VuStats Login IDs
  - VuStats Service Level
- CallMaster VI
- CallVisor ASAI
- Call Vectoring Enhancements
  - Adjunct Routing
  - Advanced Vector Routing
  - ASAI Routing (not the same as Adjunct Routing)
  - Automatic Number Identification/Information Interchange Digits Routing
  - Best Service Routing
  - Call Information Forwarding (CINFO)
  - Expert Agent Selection
  - G3V4 Enhanced Features
  - Look-Ahead Interflow
  - Vector Directory Number of Origin Announcements
  - Vector Directory Number Return Destination
- Call Work Codes
- Centralized Attendant Service (Main and Branch)
- CentreVu® Advocate

- CentreVu Call Management System
- CONVERSANT®
- DEFINITY IP Solutions (only IP trunking is supported on DEFINITY BCS and GuestWorks)
- DEFINITY Network Administration
- DEFINITY Network Management
- Digital Multiplexed Interface
- Direct Agent Calling
- Distributed Communications System
- Distributed Communications System Plus
- Dual-Tone Multifrequency Feedback Signals for Voice Recognition Unit
- Expert Agent Selection
- Extension Number Portability
- Flexible Billing
- Forced Automatic Call Distribution Calls
- Inbound Call Management
- Main/Satellite
- Modem Pooling
- Multimedia Application Server Interface
- Multimedia Call Handling (MMCH)
- Multiple Call Handling - Forced
- Multiple Music Sources (requires Tenant Partitioning)
- Outbound Call Management
- PC Application Software Translation Exchange (PASTE)
- Port Node Controller Duplication
- QSIG Enhancements
  - Basic Supplementary Services
  - Call Independent Signaling Connections
  - CAS Attendant Display of COR
  - CAS Attendant Return Call
  - CAS Display Enhancements
  - CAS Priority Queue
  - CAS RLT Emulation via PRI
  - Call Independent Signaling Connections (CISC) Enhancements

- DEFINITY / OMD QSIG Integration
- Distributed Communications System Interworking
- Manufacturers Specific Information
- Message Waiting Indication
- Lucent “VALU” features
- Path Replacement
- Supplementary Services with Rerouting
- Temporary Signaling Connections (Call-Independent Signaling Connections)
- Uniform Dial Plan

- Reason Codes
- Service Observing
  - Remote by Feature Access Code
  - Vector Directory Numbers
- Survivable Remote Expansion Port Network
- Tenant Partitioning
- Timed After Call Work Code
- Voice Recognition Integration
- Voice Recognition Unit
- VuStats
- Wideband Switching.

Related Documents

C

This appendix includes a brief description of documents that are used with the DEFINITY BCS and GuestWorks offers.

NOTE:
Some of these documents contain information about features not supported with the DEFINITY BCS or GuestWorks offers. See Page B-1 for a listing of the features not supported with these systems.

To order copies of these documents, refer to the ordering information on the back of the title page.

Reference documents

DEFINITY® ECS Documentation Library 555-233-813

A CD-ROM that contains all of the documents referenced in this section. The documents are presented in “pdf” format using the Adobe® Acrobat® reader.

DEFINITY® ECS Change Description 555-233-411

Gives a high-level overview of what is new in DEFINITY ECS R8, the base load for DEFINITY BCS and GuestWorks Issue 6. Describes the hardware and software enhancements and lists the problem corrections for this release. It also includes any last-minute changes that come in after the remaining documents have gone to production.

DEFINITY® ECS System Description **555-233-200**

Provides hardware descriptions, site requirements, technical specifications, and capacity limits.

DEFINITY® ECS and GuestWorks Property Management System Interface Specifications **555-231-601**

Describes the property management system (PMS) interface for several Lucent Technologies systems. Provides detailed interface specifications. Written for property management system vendors to design products that interface to these systems. Includes a description of each protocol mode and feature code. Covers the following releases: System 75 R1V3, DEFINITY G1, DEFINITY G3 Versions 1-4, DEFINITY ECS, and GuestWorks.

DEFINITY® Terminals and Adjuncts Reference **555-015-201**

Describes peripheral equipment that can be used with the DEFINITY systems. Written for both customers and Lucent Technologies account teams who select the correct peripherals to accompany a system.

BCS Products Security Handbook **555-025-600**

Written for console operators, telecommunications managers, and other telecommunications management personnel with responsibilities for implementing a security policy. Discusses security risks and measures that you can take to help prevent external telecommunication fraud. Includes specific information on the DEFINITY and INTUITY systems.

MERLIN LEGEND® Network Reference **555-661-150**

Describes networking used with the MERLIN LEGEND product. This document includes information about the Centralized Voice Mail via Interswitch Mode Codes feature.

Service documents

GuestWorks® Technician Handbook **555-231-109**

Describes how to connect, administer, and test the adjuncts of the GuestWorks system. Written for technician's and software consultants. Includes connectivity diagrams, administration screens, and hardware testing procedures.

DEFINITY® ECS Installation, Upgrades and Additions for Compact Modular Cabinets **555-233-118**

DEFINITY® ECS Installation for Single-Carrier Cabinets **555-233-120**

DEFINITY® ECS Installation and Test for Multi-Carrier Cabinets **555-233-114**

Provides procedures for installing the different system cabinets.

DEFINITY® ECS Installation for Adjuncts and Peripherals **555-233-116**

Provides procedures and information for hardware installation and initial testing of typical adjuncts and peripherals used with the DEFINITY communications system.

DEFINITY® ECS Administrator's Guide **555-233-506**

Provides step-by-step procedures for administering features and services on the switch.

DEFINITY® ECS Administration for Network Connectivity **555-233-504**

Provides step-by-step procedures for administering networking features and services on the switch. This document includes information about the Centralized Voice Mail via Interswitch Mode Codes feature.

DEFINITY® System's Little Instruction Book for Basic Administration	**555-233-756**
DEFINITY® System's Little Instruction Book for Advanced Administration	**555-233-757**
DEFINITY® System's Little Instruction Book for Basic Diagnostics	**555-233-758**
DEFINITY® System's Little Instruction Box	**555-233-908**

Provides step-by-step procedures for doing basic and advanced administration, and basic diagnostics on DEFINITY systems. The Little Instruction Box set includes the first three Little Instruction Books.

Maintenance for DEFINITY® ECS R8csi	**555-230-119**
Maintenance for DEFINITY® ECS R8si	**555-233-123**
Maintenance for DEFINITY® ECS R8r	**555-233-117**

Provides detailed descriptions of the procedures for monitoring, testing, and maintaining each type of system. Included are maintenance commands, step-by-step trouble-clearing procedures, the procedures for using all tests, and explanations of the system's error codes.

DEFINITY® ECS Upgrades and Additions for R8si	**555-233-122**
DEFINITY® ECS Upgrades and Additions for R8r	**555-233-115**

Provides procedures for upgrading and adding to existing *si* and *r* systems.

DEFINITY® ECS Reports	**555-233-505**

Provides detailed descriptions of the measurement, status, security, and recent change history reports available in the system and is intended for administrators who validate traffic reports and evaluate system performance. Includes corrective actions for potential problems.

User documents

DEFINITY® ECS Console Operations **555-230-700**

DEFINITY® ECS Console Operations Quick Reference **555-230-890**

Provides operating instructions for the attendant console. Included are descriptions of the console control keys and functions, call-handling procedures, basic system troubleshooting information, and routine maintenance procedures.

GuestWorks and DEFINITY® ECS Hospitality Operations **555-233-755**

Describes how to use the hospitality features of the system. Included are guest procedures, front desk procedures, housekeeping staff procedures, administration, and reports.

Using the New Abbreviated Dialing Program Feature **555-233-705**

Provides instructions on how to use the new enhancements to the Abbreviated Dialing feature.

DEFINITY® ECS Basic Call Management System Operations **555-230-706**

A complete description of the Basic Call Management System. Includes instructions for generating reports; sample reports; explanations of all data in reports; information for interpreting reports for call center management; and procedures for using the report scheduler.

GuideBuilder™ Software for DEFINITY® ECS Telephones **555-230-755**

Telephone Guide Builder is a PC software product that allows you to produce laser-printed documentation for specific voice terminals. The software is supported by a comprehensive user’s guide and online help.

GuestWorks® server INTUITY® Lodging Call Accounting User's Guide 555-231-205

Describes the INTUITY Lodging call accounting system offered with GuestWorks. Provides procedures for setting up reports and capturing call record data. Written for customers. Includes accessing the system, managing the database, accessing and printing reports, sample reports, backing up the system, and troubleshooting.

DEFINITY® BCS and GuestWorks® Issue 6 Call Vectoring Guide 555-231-785

Describes how to use the Call Vectoring feature as supported by the DEFINITY BCS and GuestWorks offers.

Glossary and Abbreviations

A

AAR
See Automatic Alternate Routing (AAR).

Abbreviated Dialing (AD)
A feature that allows callers to place calls by dialing one or two digits.

AC
1. Alternating current.
2. See Administered Connection (AC).

ACA
See Automatic Circuit Assurance (ACA).

ACD
See Automatic Call Distribution (ACD).

ACU
See Automatic Calling Unit (ACU)

access code
A one- to four-digit dial code used to activate or cancel a feature, or to access an outgoing trunk.

access endpoint
Either a nonsignaling channel on a DS1 interface or a nonsignaling port on an analog tie-trunk circuit pack that is assigned a unique extension.

access tie trunk
A trunk that connects a main communications system with a tandem communications system in an electronic tandem network (ETN). An access tie trunk can also be used to connect a system or tandem to a serving office or service node. Also called access trunk.

ACCUNET
A trademarked name for a family of digital services offered by AT&T in the United States.

AD
See Abbreviated Dialing (AD).

ADAP
AUDIX Data Acquisition Package

ADC
See analog-to-digital converter (ADC).

ADM
Asynchronous data module

Administered Connection (AC)
A feature that allows the switch to automatically establish and maintain end-to-end connections between access endpoints (trunks) and/or data endpoints (data modules).

administration group
See capability group.

Administration Without Hardware (AWOH)
A feature that allows administration of ports without associated telephones or other hardware being present.

ADU
See asynchronous data unit (ADU).

AE
See access endpoint.

AIOD
Automatic Identification of Outward Dialing

ALM-ACK
Alarm acknowledge

AMW
Automatic Message Waiting

analog
The representation of information by continuously variable physical quantities such as amplitude, frequency, and phase. See also digital.

analog data
Data that is transmitted over a digital facility in analog (PCM) form. The data must pass through a modem either at both ends or at a modem pool at the distant end.

analog telephone
A telephone that receives acoustic voice signals and sends analog electrical signals along the telephone line. Analog telephones are usually served by a single wire pair (tip and ring). The model-2500 telephone set is a typical example of an analog telephone.

analog-to-digital converter (ADC)
A device that converts an analog signal to digital form. See also digital-to-analog converter (DAC).

ANI
See Automatic Number Identification (ANI).

ANSI
American National Standards Institute**.** A United States professional/technical association supporting a variety of standards.

answer tone
A high-pitched continuous tone that indicates a data endpoint has answered.

answerback code
A number used to respond to a page from a code-calling or loudspeaker-paging system, or to retrieve a parked call.

appearance
A software process that is associated with an extension and whose purpose is to supervise a call. An extension can have multiple appearances. Also called call appearance, line appearance, and occurrence. See also call appearance.

ARS
See Automatic Route Selection (ARS).

ASCII (American Standard Code for Information Interchange)
The standard code for representing characters in digital form. Each character is represented by an 8-bit code (including parity bit).

asynchronous data transmission
A method of transmitting data in which each character is preceded by a start bit and followed by a stop bit, thus permitting data characters to be transmitted at irregular intervals. This type of transmission is advantageous when transmission is not regular (characters typed at a keyboard). Also called asynchronous transmission. See also synchronous data transmission.

asynchronous data unit (ADU)
A device that allows direct connection between RS-232C equipment and a digital switch.

attendant
A person who uses a console to provide personalized service for incoming callers and voice-services users by performing switching and signaling operations. See also attendant console.

attendant console
The workstation used by an attendant. The attendant console allows the attendant to originate a call, answer an incoming call, transfer a call to another extension or trunk, put a call on hold, and remove a call from hold. Attendants using the console can also manage and monitor some system operations. Also called console. See also attendant.

Audio Information Exchange (AUDIX)
A fully integrated voice-mail system. Can be used with a variety of communications systems to provide call-history data, such as subscriber identification and reason for redirection.

auto-in trunk group
Trunk group for which the CO processes all of the digits for an incoming call. When a CO seizes a trunk from an auto-in trunk group, the switch automatically connects the trunk to the destination — typically an ACD split where, if no agents are available, the call goes into a queue in which callers are answered in the order in which their calls arrive.

Auto-In Work mode
One of four work modes: the mode in which an agent is ready to process another call as soon as the current call is completed.

Automatic Alternate Routing (AAR)
A feature that routes calls to other than the first-choice route when facilities are unavailable.

Automatic Callback (ACB)
A feature that enables internal callers, upon reaching a busy extension, to have the system automatically connect and ring both parties when the called party becomes available.

Automatic Call Distribution (ACD)
A feature that answers calls, and then, depending on administered instructions, delivers messages appropriate for the caller and routes the call to an agent when an agent becomes available.

Automatic Call Distribution (ACD) Split
A method of routing calls of a similar type among agents in a split. Also, a group of extensions that are staffed by agents trained to handle a certain type of incoming call.

Automatic Calling Unit (ACU)
A device that places a telephone call.

Automatic Circuit Assurance (ACA)
A feature that tracks calls of unusual duration to facilitate troubleshooting. A high number of very short calls or a low number of very long calls may signify a faulty trunk.

Automatic Number Identification (ANI)
Representation of the calling number, for display or for further use to access information about the caller. Available with Signaling System 7.

automatic restoration
A service that restores disrupted connections between access endpoints (nonsignaling trunks) and data endpoints (devices that connect the switch to data terminal and/or communications equipment). Restoration is done within seconds of a service disruption so that critical data applications can remain operational.

Automatic Route Selection (ARS)
A feature that allows the system to automatically choose the least-cost way to send a toll call.

automatic trunk
A trunk that does not require addressing information because the destination is predetermined. A request for service on the trunk, called a seizure, is sufficient to route the call. The normal destination of an automatic trunk is the communications-system attendant group. Also called automatic incoming trunk and automatic tie trunk.

AUX
Auxiliary

auxiliary equipment
Equipment used for optional system features, such as Loudspeaker Paging and Music-on-Hold.

auxiliary trunk
A trunk used to connect auxiliary equipment, such as radio-paging equipment, to a communications system.

Aux-Work mode
A work mode in which agents are unavailable to receive ACD calls. Agents enter Aux-Work mode when involved in non-ACD activities such as taking a break, going to lunch, or placing an outgoing call.

AVD
Alternate voice/data

AWOH
See Administration Without Hardware (AWOH).

AWG
American Wire Gauge

AWT
Average work time

B

bandwidth
The difference, expressed in hertz, between the defined highest and lowest frequencies in a range.

baud
A unit of transmission rate equal to the number of signal events per second. See also bit rate and bits per second (bps).

BCC
See Bearer capability class (BCC).

BCMS
Basic Call Management System

Bearer capability class (BCC)
A code that identifies the type of a call (for example, voice and different types of data). Determination of BCC is based on the caller's characteristics for non-ISDN endpoints and on the Bearer Capability and Low-Layer Compatibility Information Elements of an ISDN endpoint. Current BCCs are 0 (voice-grade data and voice), 1 (56 kbps data transmission), 2 (synchronous/asynchronous data transmission up to 19.2 kbps) 3 (64 kbps circuit/packet data transmission), 4 (64 kbps synchronous data), 5 (temporary signaling connection, and 6 (wideband call, 128–1984 kbps synchronous data; not supported by this system).

bit (binary digit)
One unit of information in binary notation, having two possible values: 0 or 1.

bits per second (bps)
The number of binary units of information that are transmitted or received per second. See also baud and bit rate.

bit rate
The speed at which bits are transmitted, usually expressed in bits per second. Also called data rate. See also baud and bits per second (bps).

BLF
Busy Lamp Field

BPN
Billed-party number

bps
See bits per second (bps).

bridge (bridging)
The appearance of a telephone's extension at one or more other telephone.

BRI
The ISDN Basic Rate Interface specification.

bridged appearance
A call appearance on a telephone that matches a call appearance on another telephone for the duration of a call.

buffer
1. In hardware, a circuit or component that isolates one electrical circuit from another. Typically, a buffer holds data from one circuit or process until another circuit or process is ready to accept the data.
2. In software, an area of memory that is used for temporary storage.

bus
A multiconductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

busy tone
A low-pitched repeating tone that indicates the dialed number is in use.

BX.25
A version of the CCITT X.25 protocol for data communications. BX.25 adds a fourth level to the standard X.25 interface. This uppermost level combines levels 4, 5, and 6 of the ISO reference model.

bypass tie trunks
A one-way, outgoing tie trunk from a tandem switch to a main switch in an ETN. Bypass tie trunks, provided in limited quantities, are used as a last-choice route when all trunks to another tandem switch are busy. Bypass tie trunks are used only if all applicable intertandem trunks are busy.

byte
A sequence of (usually eight) bits processed together.

C

cabinet
Housing for racks, shelves, or carriers that hold electronic equipment.

call appearance
1. For the attendant console, six buttons, labeled a–f, used to originate, receive, and hold calls. Two lights next to the button show the status of the call appearance.
2. For a telephone, a button labeled with an extension and used to place outgoing calls, receive incoming calls, or hold calls. Two lights next to the button show the status of the call appearance.

Call Detail Recording (CDR)
A feature that uses software and hardware to record call data (same as CDRU).

Call Detail Recording utility (CDRU)
Software that collects, stores, optionally filters, and outputs call-detail records.

callback call
A call that automatically returns to a user's telephone who activated the Automatic Callback or Ringback Queuing feature.

Call Vectoring directory number
An extension that provides access to the Call Vectoring feature on the switch. Call Vectoring allows a customer to specify the treatment of incoming calls based on the dialed number.

call-waiting ringback tone
A low-pitched tone identical to ringback tone except that the tone decreases in the last 0.2 seconds (in the United States). Call-waiting ringback tone notifies the attendant that Attendant Call Waiting is active and that the called party is aware of the waiting call. Tones in international countries may sound different.

call-waiting tone
One, two, or three beeps (short bursts of high-pitched tone) that indicate to a busy single-line telephone that an incoming call is waiting. The type of incoming call determines the number of beeps the busy telephone receives: one beep indicates that the call is from another telephone in the system, two beeps indicate that the call is from the attendant or an outside caller, and three beeps indicate that the waiting call is a priority call.

CAMA
Centralized Automatic Message Accounting

carrier
An enclosed shelf containing vertical slots that hold circuit packs.

carried load
The amount of traffic served by traffic-sensitive facilities during a given interval.

CAS
Call Accounting System

CCS or hundred call seconds
A unit of call traffic. Call traffic for a facility is scanned every 100 seconds. If the facility is busy, it is assumed to have been busy for the entire scan interval. There are 3600 seconds per hour. The Roman numeral for 100 is the capital letter C. The abbreviation for call seconds is CS. Therefore, 100 call seconds is abbreviated CCS. If a facility is busy for an entire hour, then it is said to have been busy for 36 CCS. See also Erlang.

capability
A request or indication of an operation. For example, *Third Party Make Call* is a request for setting up a call; *event report* is an indication that an event has occurred.

capability group
A set of capabilities, determined by switch administration, that can be requested by an application. Capability groups denote association types. For example, *Call Control* is a type of association that allows certain functions (the ones in the capability group) to be performed over this type of association. Also referred to as administration groups or application service elements (ASEs).

CCIS
Common-Channel Interoffice Signaling

CCITT
CCITT (Comitte Consultatif International Telephonique et Telegraphique), now called *International Telecommunications Union* (ITU). See International Telecommunications Union (ITU).

CCS
Centum Call Seconds (CCS or hundred call seconds).

CCSA
Common-Control Switching Arrangement

CDM
Channel-division multiplexing

CDR
See Call Detail Recording (CDR).

CDRP
Call Detail Record Poller

CDRR
Call Detail Recording and Reporting

CDRU
See Call Detail Recording utility (CDRU).

CEM
Channel-expansion multiplexing

Center-Stage Switch
The Center-Stage Switch is a connection hub that provides port network communication. It is an essential component of a system configuration if the system is composed of more than three port networks.

central office (CO)
The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

central office (CO) codes
The first three digits of a seven-digit public-network telephone number in the United States.

central office (CO) trunk
A telecommunications channel that provides access from the system to the public network through the local CO.

CEPT1
European Conference of Postal and Telecommunications Rate 1

channel
1. A circuit-switched call.
2. A communications path for transmitting voice and data.
3. A DS0 on a T1 or E1 facility not specifically associated with a logical circuit-switched call; analogous to a single trunk.

channel negotiation
The process by which the channel offered in the Channel Identification Information Element (CIIE) in the SETUP message is negotiated to be another channel acceptable to the switch that receives the SETUP message and ultimately to the switch that sent the SETUP. Negotiation is attempted only if the CIIE is encoded as *Preferred*.

circuit pack
A board on which electrical circuits are printed, and IC chips and electrical components are installed. A circuit pack is installed in a switch carrier.

CISPR
International Special Committee on Radio Interference

Class of Restriction (COR)
A feature that allows up to 95 classes of call-origination and call-termination restrictions for telephones, telephone groups, data modules, and trunk groups. See also Class of Service (COS).

Class of Service (COS)
A feature that uses a number to specify if telephone users can activate the Automatic Callback, Call Forwarding All Calls, Data Privacy, or Priority Calling features. See also Class of Restriction (COR).

CO
See central office (CO).

common-control switching arrangement (CCSA)
A private telecommunications network using dedicated trunks and a shared switching center for interconnecting company locations.

communications system
The software-controlled processor complex that interprets dialing pulses, tones, and keyboard characters and makes the proper connections both within the system and external to the system. The communications system itself consists of a digital computer, software, storage device, and carriers with special hardware to perform the connections. A communications system provides

voice and data communications services, including access to public and private networks, for telephones and data terminals on a customer's premises. See also switch.

companding
Compress + Expand. Compress the digital code prior to transmission, and expand the received code prior to reconstructing the analog signal. This is a nonlinear encoding technique to minimize data rate requirements yet preserve signal quality.

confirmation tone
Three short bursts of tone that confirms a feature activation, deactivation, or cancellation has been accepted. This tone also can indicate that an outgoing call from a single-line telephone was placed in a ringback queue.

connectivity
The connection of disparate devices within a single system.

console
See attendant console.

contiguous
Adjacent DS0s within one T1 or E1 facility or adjacent TDM or fiber time slots. The first and last TDM bus, DS0, or fiber time slots are not considered contiguous (no wraparound). For an E1 facility with a D-channel, DS0s 15 and 17 are considered contiguous.

control cabinet
See control carrier.

control carrier
A carrier in a multicarrier cabinet that contains the SPE circuit packs and, unlike an R5r control carrier, port circuit packs. Also called control cabinet in a single-carrier cabinet. See also switch-processing element (SPE).

controlled station
A station that is monitored and controlled via a domain-control association.

COR
See Class of Restriction (COR).

COS
See Class of Service (COS).

coverage answer group
A group of up to eight telephones that ring simultaneously when a call is redirected to it by Call Coverage. Any one of the group can answer the call.

coverage call
A call that is automatically redirected from the called party's extension to an alternate answering position when certain coverage criteria are met.

coverage path
The order in which calls are redirected to alternate answering positions.

coverage point
An extension or attendant group, Call Vectoring directory number, or ACD split designated as an alternate answering position in a coverage path.

coverage tone
A long-burst of tone indicating to the calling party that a call to an extension is being answered at another extension by a covering user.

covering user
A person at a coverage point who answers a redirected call.

CP
Circuit pack

CPE
Customer-premises equipment

CPN
Called-party number

CSA
Canadian Safety Association

CSD
Customer-service document

CSSO
Customer Services Support Organization

D

DAC
1. Dial access code
2. See digital-to-analog converter (DAC).

data channel
A communications path between two points used to transmit digital signals.

data-communications equipment (DCE)
The equipment (usually a modem, data module, or packet assembler/disassembler) on the network side of a communications link that makes the binary serial data from the source or transmitter compatible with the communications channel.

data link
The configuration of physical facilities enabling end terminals to communicate directly with each other.

data module
An interconnection device between a BRI or DCP interface of the switch and data terminal equipment or data communications equipment.

data port
A point of access to a computer that uses trunks or lines for transmitting or receiving data.

data rate
See bit rate.

data service unit (DSU)
A device that transmits digital data on transmission facilities.

data terminal
An input/output (I/O) device that has either switched or direct access to a host computer or to a processor interface.

data terminal equipment (DTE)
Equipment consisting of the endpoints in a connection over a data circuit. In a connection between a data terminal and host, the terminal, the host, and their associated modems or data modules make up the DTE.

DC
Direct current

DCE
Data-communications equipment

D-channel backup
A type of backup used with Non-Facility Associated Signaling (NFAS). A primary D-channel provides signaling for an NFAS D-channel group (two or more PRI facilities). A second D-channel, on a separate PRI facility of the NFAS D-channel group, is designated as backup for the D-channel. Failure of the primary D-channel causes automatic transfer of call-control signaling to the backup D-channel. The backup becomes the primary D-channel. When the failed channel returns to service, it becomes the backup D-channel.

DCP
Digital Communications Protocol

DDC
Direct Department Calling

DDD
Direct Distance Dialing

delay-dial trunk
A trunk that allows dialing directly into a communications system (digits are received as they are dialed).

designated telephone
The specific telephone to which calls, originally directed to a certain extension, are redirected. Commonly used to mean the forwarded-to telephone when Call Forwarding All Calls is active.

dial tone
A continuous tone indicating that a user can dial a number or activate features.

dial-repeating trunks
A PBX tie trunk that is capable of handling PBX station-signaling information without attendant assistance.

dial-repeating tie trunk
A tie trunk that transmits called-party addressing information between two communications systems.

DID
Direct Inward Dialing

digit conversion
A process used to convert specific dialed numbers into other dialed numbers.

digital
The representation of information by discrete steps. See also analog.

digital communications protocol (DCP)
A proprietary protocol used to transmit both digitized voice and digitized data over the same communications link. A DCP link is made up of two 64 kbps information (I-) channels and one 8 kbps signaling (S-) channel. Digital Communications Protocol. The DCP protocol supports two

information-bearing channels, and thus two telephones/data modules. The I1 channel is the DCP channel assigned on the first page of the 8411 station form. The I2 channel is the DCP channel assigned on the analog adjunct page of the 8411 station form or on the data module page.

The DCP protocol supports two information-bearing channels, and thus two telephones/data modules. The I1 channel is the DCP channel assigned on the first page of the 8411 station form. The I2 channel is the DCP channel assigned on the analog adjunct page of the 8411 station form or on the data module page.

digital data endpoints
In the system, devices such as the 510D terminal or the 515-type business communications terminal (BCT).

digital signal level 0 (DS0)
A single 64 kbps voice channel. A DS0 is a single 64 kbps channel in a T1 or E1 facility and consists of eight bits in a T1 or E1 frame every 125 microseconds.

digital signal level 1 (DS1)
A single 1.544 Mbps (United States) or 2.048 Mbps (outside the United States) digital signal carried on a T1 transmission facility. A DS1 converter complex consists of a pair, one at each end, of DS1 converter circuit packs and the associated T1/E1 facilities.

digital terminal data module (DTDM)
An integrated or adjunct data module that shares with a digital telephone the same physical port for connection to a communications system. The function of a DTDM is similar to that of a PDM and MPDM in that it converts RS-232C signals to DCP signals.

digital-to-analog converter (DAC)
A device that converts data in digital form to the corresponding analog signals. See also analog-to-digital converter (ADC).

digital transmission
A mode of transmission in which information to be transmitted is first converted to digital form and then transmitted as a serial stream of pulses.

digital trunk
A circuit that carries digital voice and/or digital data in a telecommunications channel.

DIOD
Direct Inward and Outward Dialing

Direct Extension Selection (DXS)
A feature on an attendant console that allows an attendant direct access to telephones by pressing a group-select button and a DXS button.

Direct Inward Dialing (DID)
A feature that allows an incoming call from the public network (not FX or WATS) to reach a specific telephone without attendant assistance.

Direct Inward Dialing (DID) trunk
An incoming trunk used for dialing directly from the public network into a communications system without help from the attendant.

DIVA
Data In/Voice Answer

DLC
Data line circuit

DND
Do not disturb

DNIS
Dialed-Number Identification Service

DOD
Direct Outward Dialing

DOSS
Delivery Operations Support System

DS1
Digital Signal Level 1

DS1C
Digital Signal Level-1 protocol C

DS1 CONV
Digital Signal Level-1 converter

DSI
Digital signal interface

DSU
Data service unit

DTDM
Digital-terminal data module

DTE
Data-terminal equipment

DTGS
Direct Trunk Group Select

DTMF
Dual-tone multifrequency

DWBS
DEFINITY Wireless Business System

DXS
Direct extension selection

E

E1
A digital transmission standard that carries traffic at 2.048 Mbps. The E1 facility is divided into 32 channels (DS0s) of 64 kbps information. Channel 0 is reserved for framing and synchronization information. A D-channel occupies channel 16.

E & M
Ear and mouth (receive and transmit)

ear and mouth (E & M) signaling
Trunk supervisory signaling, used between two communications systems, whereby signaling information is transferred through two-state voltage conditions (on the E and M leads) for analog applications and through a single bit for digital applications.

EBCDIC
Extended Binary-Coded Decimal Interexchange Code

ECMA
European Computer Manufacturers Association

EIA
Electronic Industries Association

EIA-232
A physical interface specified by the EIA. EIA-232 transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet. EIA-232 replaces RS-232 protocol in some applications.

electronic tandem network (ETN)
A tandem tie-trunk network that has automatic call-routing capabilities based on the number dialed and the most preferred route available. Each switch in the network is assigned a unique private network office code (RNX), and each telephone is assigned a unique extension.

Electronics Industries Association (EIA)
A trade association of the electronics industry that establishes electrical and functional standards.

emergency transfer
If a major system failure occurs, automatic transfer is initiated to a group of telephones capable of making outgoing calls. The system operates in this mode until the failure is repaired and the system automatically returns to normal operation. Also called power-failure transfer.

EMI
Electromagnetic interference

end-to-end signaling
The transmission of touch-tone signals generated by dialing from a telephone to remote computer equipment. These digits are sent over the trunk as DTMF digits whether the trunk signaling type is marked as tone or rotary and whether the originating station is tone or rotary. Example: a call to a voice-mail system or automated-attendant service. A connection is first established over an outgoing trunk. Then additional digits are dialed to transmit information to be processed by the computer equipment.

enhanced private-switched communications service (EPSCS)
An analog private telecommunications network based on the No. 5 crossbar and 1A ESS that provides advanced voice and data telecommunications services to companies with many locations.

EPSCS
Enhanced Private Switched Communications Services

ERL
Echo return loss

Erlang
A unit of traffic intensity, or load, used to express the amount of traffic needed to keep one facility busy for one hour. One Erlang is equal to 36 CCS. See also CCS or hundred call seconds.

ESF
Extended superframe format

ESPA
European Standard Paging Access

ETA
1. Extended Trunk Access
2. Enhanced Terminal Administration

ETN
Electronic tandem network

ETSI
European Telecommunications Standards Institute

extension-in
Extension-In (ExtIn) is the work state agents go into when they answer (receive) a non-ACD call.

extension-out
The work state that agents go into when they place (originate) a non-ACD call.

extension
A 1- to 5-digit number by which calls are routed through a communications system or, with a Uniform Dial Plan (UDP), through a private network.

external call
A connection between a communications system user and a party on the public network or on another communications system in a private network.

F

FAC
Feature Access Code

facility
A telecommunications transmission pathway and associated equipment.

facility-associated signaling (FAS)
Signaling for which a D-channel carries signaling only for those channels on the same physical interface.

FAS
Facility-associated signaling

FCC
Federal Communications Commission

FEAC
Forced Entry of Account Codes

feature
A specifically defined function or service provided by the system.

feature button
A labeled button on a telephone or attendant console used to access a specific feature.

fiber optics
A technology using materials that transmit ultrawideband electromagnetic light-frequency ranges for high-capacity carrier systems.

FNPA
Foreign Numbering-Plan Area

foreign-exchange (FX)
A CO other than the one providing local access to the public telephone network.

foreign-exchange trunk
A telecommunications channel that directly connects the system to a CO other than its local CO.

foreign numbering-plan area code (FNPAC)
An area code, other than the local area code, that must be dialed to call outside the local geographical area.

FRL
Facilities Restriction Level

FX
Foreign exchange

G

generalized route selection (GRS)
An enhancement to Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) that performs routing based on call attributes, such as Bearer Capability Classes (BCCs), in addition to the address and facilities restriction level (FRL), thus facilitating a Uniform Dial Plan (UDP) that is independent of the type of call being placed.

glare
The simultaneous seizure of a 2-way trunk by two communications systems, resulting in a standoff.

grade of service
The number of call attempts that fail to receive service immediately. Grade of service is also expressed as the quantity of all calls that are blocked or delayed.

ground-start trunk
A trunk on which, for outgoing calls, the system transmits a request for services to a distant switching system by grounding the trunk ring lead. To receive the digits of the called number, that system grounds the trunk tip lead. When the system detects this ground, the digits are sent.

GRS
Generalized Route Selection

H

H0
An ISDN information transfer rate for 384 kbps data defined by CCITT and ANSI standards.

H11
An ISDN information transfer rate for 1536 kbps data defined by CCITT and ANSI standards.

H12
An ISDN information transfer rate for 1920 kbps data defined by CCITT and ANSI standards.

handshaking logic
A format used to initiate a data connection between two data module devices.

hertz (Hz)
A unit of frequency equal to one cycle per second.

HNPA
See home numbering-plan area code (HNPA).

holding time
The total length of time in minutes and seconds that a facility is used during a call.

home numbering-plan area code (HNPA)
The local area code. The area code does not have to be dialed to call numbers within the local geographical area.

hop
Nondirect communication between two switch communications interfaces (SCI) where the SCI message passes automatically without intermediate processing through one or more intermediate SCIs.

hunt group
A group of extensions that are assigned so that a call to a busy extension reroutes to an idle extension in the group. See also ACD split.

hunt group condition
A condition whereby a caller is temporarily separated from a connection with an attendant. A hunt group condition automatically occurs when the attendant, active on a call, presses the start button.

hunt group number
The hunt group's identity to the switch and BCMS.

hunt group report
A report that provides historical traffic information for internally-measured hunt groups.

I

I1
The first information channel of DCP.

I2
The second information channel of DCP.

ICC
Intercabinet cable or intercarrier cable

ICDOS
International Customer-Dialed Operator Service

ICI
Incoming call identifier

ICM
Inbound Call Management

IDDD
International Direct Distance Dialing

IDF
Intermediate distribution frame

immediate-start tie trunk
A trunk on which, after making a connection with a distant switching system for an outgoing call, the system waits a nominal 65 ms before sending the digits of the called number. This allows time for the distant system to prepare to receive digits. On an incoming call, the system has less than 65 ms to prepare to receive the digits.

INADS
Initialization and Administration System

incoming gateway
A PBX that routes an incoming call on a trunk *not* administered for Supplementary Services Protocol B to a trunk *not* administered for Supplementary Services Protocol B.

information exchange
The exchange of data between users of two different systems, such as the switch and a host computer, over a LAN.

INS
ISDN Network Service

inside call
A call placed from one telephone to another within the local communications system.

Integrated Services Digital Network (ISDN)
A public or private network that provides end-to-end digital communications for all services to which users have access by a limited set of standard multipurpose user-network interfaces defined by the CCITT. Through internationally accepted standard interfaces, ISDN provides digital circuit-switched or packet-switched communications within the network and links to other ISDNs to provide national and international digital communications. See also Integrated Services Digital Network Basic Rate Interface (ISDN-BRI) and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI).

Integrated Services Digital Network Basic Rate Interface (ISDN-BRI)
The interface between a communications system and terminal that includes two 64 kbps B-channels for transmitting voice or data and one 16 kbps D-channel for transmitting associated B-channel call control and out-of-band signaling information. ISDN-BRI also includes 48 kbps for transmitting framing and D-channel contention information, for a total interface speed of 192 kbps. ISDN-BRI serves ISDN telephones and digital terminals fitted with ISDN terminal adapters. See also Integrated Services Digital Network (ISDN) and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI).

Integrated Services Digital Network Primary Rate Interface (ISDN-PRI)
The interface between multiple communications systems that in North America includes 24 64 kbps channels, corresponding to the North American digital signal level-1 (DS1) standard rate of 1.544 Mbps. The most common arrangement of channels in ISDN-PRI is 23 64 kbps B-channels for transmitting voice and data and one 64 kbps D-channel for transmitting associated B-channel call control and out-of-band signaling information. With nonfacility-associated signaling (NFAS), ISDN-PRI can include 24 B-channels and no D-channel. See also Integrated Services Digital Network (ISDN) and Integrated Services Digital Network Basic Rate Interface (ISDN-BRI).

intercept tone
An alternating high and low tone that indicates a dialing error or denial of the service requested.

internal call
A connection between two users within a system.

International Telecommunications Union (ITU)
Formerly known as International Telegraph and Telephone Consultative Committee (CCITT), ITU is an international organization that sets universal standards for data communications, including ISDN. ITU members are from telecommunications companies and organizations around the world.

International Telegraph and Telephone Consultative Committee
See International Telecommunications Union (ITU).

interflow
The ability for calls to forward to other splits on the same PBX or a different PBX using the Call Forward All Calls feature.

intraflow
The ability for calls to redirect to other splits on the same PBX on a conditional or unconditional basis using call coverage busy, don't answer, or all criteria.

internal measurements
BCMS measurements that are made by the system.

in-use lamp
A red light on a multi-appearance telephone that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

INWATS
Inward Wide Area Telephone Service

ISDN
See Integrated Services Digital Network (ISDN).

ISDN trunk
A trunk administered for use with ISDN-PRI. Also called ISDN facility.

ISDN-PRI terminal adapter
An interface between endpoint applications and an ISDN PRI facility. ISDN-PRI terminal adapters are currently available from other vendors and are primarily designed for video conferencing applications. Accordingly, currently available terminal adapters adapt the two pairs of video codec data (V.35) and dialing (RS-366) ports to an ISDN PRI facility.

ISO
International Standards Organization

ITU
International Telecommunications Union

IXC
Interexchange carrier code

L

LAN
Local area network

LAP-D
Link Access Procedure on the D-channel

LAPD
Link Access Procedure data

LATA
Local access and transport area

LDN
Listed directory number

LEC
Local exchange carrier

LED
See light-emitting diode (LED).

light-emitting diode (LED)
A semiconductor device that produces light when voltage is applied. LEDs provide a visual indication of the operational status of hardware components, the results of maintenance tests, the alarm status of circuit packs, and the activation of telephone features.

lightwave transceiver
Hardware that provides an interface to fiber-optic cable from port circuit packs and DS1 converter circuit packs. Lightwave transceivers convert electrical signals to light signals and vice versa.

line
A transmission path between a communications system or CO switching system and a telephone or other terminal.

line appearance
See appearance.

line buildout
A selectable output attenuation is generally required of DTE equipment because T1 circuits require the last span to lose 15–22.5 dB.

line port
Hardware that provides the access point to a communications system for each circuit associated with a telephone or data terminal.

link
A transmitter-receiver channel that connects two systems.

link-access procedure on the D-channel (LAPD)
A link-layer protocol on the ISDN-BRI and ISDN-PRI data-link layer (level 2). LAPD provides data transfer between two devices, and error and flow control on multiple logical links. LAPD is used for signaling and low-speed packet data (X.25 and mode 3) on the signaling (D-) channel and for mode-3 data communications on a bearer (B-) channel.

local area network (LAN)
A networking arrangement designed for a limited geographical area. Generally, a LAN is limited in range to a maximum of 6.2 miles and provides high-speed carrier service with low error rates. Common configurations include daisy chain, star (including circuit-switched), ring, and bus.

logical link
The communications path between a processor and a BRI terminal.

loop-start trunk
A trunk on which, after establishing a connection with a distant switching system for an outgoing call, the system waits for a signal on the loop formed by the trunk leads before sending the digits of the called number.

LSU
Local storage unit

LWC
Leave Word Calling

M

MADU
Modular asynchronous data unit

main distribution frame (MDF)
A device that mounts to the wall inside the system equipment room. The MDF provides a connection point from outside telephone lines to the PBX switch and to the inside telephone stations.

maintenance
Activities involved in keeping a telecommunications system in proper working condition: the detection and isolation of software and hardware faults, and automatic and manual recovery from these faults.

management terminal
The terminal that is used by the system administrator to administer the switch. The terminal may also be used to access the BCMS feature.

major alarm
An indication of a failure that has caused critical degradation of service and requires immediate attention. Major alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, logged to the alarm log, and reported to a remote maintenance facility, if applicable.

Manual-In work mode
One of four work modes: the mode in which an agent is ready to process another call manually. *See* Auto-In Work mode for a contrast.

MAPD
Multiapplication platform for DEFINITY

MCC
Multicarrier cabinet

MCT
Malicious Call Trace

MCU
Multipoint conferencing unit

MDF
Main distribution frame

MDM
Modular data module

MDR
Message detail record

MET
Multibutton electronic telephone

MF
Multifrequency

MFB
Multifunction board

MFC
Multifrequency code

minor alarm
An indication of a failure that could affect customer service. Minor alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, sent to the alarm log, and reported to a remote maintenance facility, if applicable.

MIS
Management information system

modular processor data module (MPDM)
A processor data module (PDM) that can be configured to provide several kinds of interfaces (RS-232C, RS-449, and V.35) to customer-provided data terminal equipment (DTE). See also processor data module (PDM).

modular trunk data module (MTDM)
A trunk data module that can be configured to provide several kinds of interfaces (RS-232, RS-449, and V.35) to customer-provided data terminal equipment.

MOS
Message-oriented signaling

MPDM
Modular processor data module

MS
Message server

MSA
Message servicing adjunct

MSG
Message service

MT
Management terminal

MTDM
Modular trunk data module

MTP
Maintenance tape processor

MTT
Multitasking terminal

multiappearance telephone
A telephone equipped with several call-appearance buttons for the same extension, allowing the user to handle more than one call on that same extension at the same time.

Multicarrier cabinet
A structure that holds one to five carriers. See also single-carrier cabinet.

Multifrequency Compelled (MFC) Release 2 (R2) signaling
A signal consisting of two frequency components, such that when a signal is transmitted from a switch, another signal acknowledging the transmitted signal is received by the switch. R2 designates signaling used in the United States and in countries outside the United States.

multiplexer
A device used to combine a number of individual channels into a single common bit stream for transmission.

multiplexing
A process whereby a transmission facility is divided into two or more channels, either by splitting the frequency band into a number of narrower bands or by dividing the transmission channel into successive time slots. See also time-division multiplexing (TDM).

multirate
The new N x DS0 service (see N x DS0).

MWL
Message-waiting lamp

N

N+1
Method of determining redundant backup requirements. Example: if four rectifier modules are required for a DC-powered single-carrier cabinet, a fifth rectifier module is installed for backup.

NANP
North American Numbering Plan

narrowband
A circuit-switched call at a data rate up to and including 64 kbps.

native telephone support
A predefined telephone type exists in switch software, eliminating the need to alias the telephone (that is, manually map call appearances and feature buttons onto some other natively supported telephone type).

NAU
Network access unit

NCA/TSC
Noncall-associated/temporary-signaling connection

NCOSS
Network Control Operations Support Center

NEMA
National Electrical Manufacturer's Association

NETCON
Network-control circuit pack

network-specific facility (NSF)
An information element in an ISDN-PRI message that specifies which public-network service is used. NSF applies only when Call-by-Call Service Selection is used to access a public-network service.

network interface
A common boundary between two systems in an interconnected group of systems.

NFAS
See Nonfacility-associated signaling (NFAS).

NID
Network Inward Dialing

NM
Network management

node
A switching or control point for a network. Nodes are either tandem (they receive signals and pass them on) or terminal (they originate or terminate a transmission path).

Nonfacility-associated signaling (NFAS)
A method that allows multiple T1 and/or E1 facilities to share a single D-channel to form an ISDN-PRI. If D-channel backup is not used, one facility is configured with a D-channel, and the other facilities that share the D-channel are configured without D-channels. If D-channel backup is used, two facilities are configured to have D-channels (one D-channel on each facility), and the other facilities that share the D-channels are configured without D-channels.

NPA
Numbering-plan area

NPE
Network processing element

NQC
Number of queued calls

O

OA
Operator assisted

occurrence
See appearance.

OCM
Outbound Call Management

ONS
On-premises station

OPS
Off-premises station

OPX
Off-premises extension

OQT
Oldest queued time

OSHA
Occupational Safety and Health Act

OSI
Open Systems Interconnect

OSS
Operations Support System

OSSI
Operational Support System Interface

OTQ
Outgoing trunk queuing

outgoing gateway
A PBX that routes an incoming call on a trunk administered for Supplementary Services Protocol B to a trunk *not* administered for Supplementary Services Protocol B.

P

PACCON
Packet control

packet
A group of bits (including a message element, which is the data, and a control information element (IE), which is the header) used in packet switching and transmitted as a discrete unit. In each packet, the message element and control IE are arranged in a specified format. See also packet bus and packet switching.

packet bus
A wide-bandwidth bus that transmits packets.

packet switching
A data-transmission technique whereby user information is segmented and routed in discrete data envelopes called packets, each with its own appended control information, for routing, sequencing, and error checking. Packet switching allows a channel to be occupied only during the transmission of a packet. On completion of the transmission, the channel is made available for the transfer of other packets. See also BX.25 and packet.

paging trunk
A telecommunications channel used to access an amplifier for loudspeaker paging.

party/extension active on call
A party is on the call if he or she is actually connected to the call (in active talk or in held state). An originator of a call is always a party on the call. Alerting parties, busy parties, and tones are not parties on the call.

PBX
Private branch exchange

PCM
See pulse-code modulation (PCM).

PCOL
Personal central-office line

PCOLG
Personal central-office line group

PCS
Permanent switched calls

PDM
See processor data module (PDM).

PDS
Premises Distribution System

PE
Processing element

PEC
Price element code

PGATE
Packet gateway

PI
Processor interface

PIB
Processor interface board

pickup group
A group of individuals authorized to answer any call directed to an extension within the group.

PKTINT
Packet interface

PLS
Premises Lightwave System

PMS
Property Management System

PN
Port network

PNA
Private network access

POE
Processor occupancy evaluation

POP
Point of presence

port
A data- or voice-transmission access point on a device that is used for communicating with other devices.

port carrier
A carrier in a multicarrier cabinet or a single-carrier cabinet containing port circuit packs, power units, and service circuits. Also called a port cabinet in a single-carrier cabinet.

port network (PN)
A cabinet containing a TDM bus and packet bus to which the following components are connected: port circuit packs, one or two tone-clock circuit packs, a maintenance circuit pack, and service circuit packs. Each PN is controlled by a switch processing element (SPE).

PPM

1. Parts per million
2. Periodic pulse metering

PPN

See processor port network (PPN).

PRI

See Primary Rate Interface (PRI).

primary extension

The main extension associated with the physical telephone or data terminal.

Primary Rate Interface (PRI)

A standard ISDN frame format that specifies the protocol used between two or more communications systems. PRI runs at 1.544 Mbps and, as used in North America, provides 23 64 kbps B-channels (voice or data) and one 64 kbps D-channel (signaling). The D-channel is the 24th channel of the interface and contains multiplexed signaling information for the other 23 channels.

PRI endpoint (PE)

A PRI endpoint consists of one or more contiguous B-channels on a line-side T1 or E1 ISDN PRI facility and has an extension. Endpoint applications have call-control capabilities over PRI endpoints.

principal

A telephone that has its primary extension bridged on one or more other telephones.

principal (user)

A person to whom a telephone is assigned and who has message-center coverage.

private network

A network used exclusively for the telecommunications needs of a particular customer.

private network office code (RNX)

The first three digits of a seven-digit private network number.

processor data module (PDM)

A device that provides an RS-232C DCE interface for connecting to data terminals, applications processors (APs), and host computers, and provides a DCP interface for connection to a communications system. See also modular processor data module (MPDM).

processor port network (PPN)

A port network controlled by a switch-processing element that is directly connected to that PN's TDM bus and LAN bus. See also port network (PN).

processor port network (PPN) control carrier

A carrier containing the maintenance circuit pack, tone/clock circuit pack, and SPE circuit packs for a processor port network (PPN) and, optionally, port circuit packs.

Property Management System (PMS)

A stand-alone computer used by lodging and health-services organizations for services such as reservations, housekeeping, and billing.

PSC

Premises service consultant

PSDN

Packet-switch public data network

PTT
Postal Telephone and Telegraph

public network
The network that can be openly accessed by all customers for local and long-distance calling.

pulse-code modulation (PCM)
An extension of pulse-amplitude modulation (PAM) in which carrier-signal pulses modulated by an analog signal, such as speech, are quantized and encoded to a digital, usually binary, format.

Q

QPPCN
Quality Protection Plan Change Notice

QSIG
QSIG provides compliance to the International Organization for Standardization (ISO) ISDN-PRI private-networking specifications. QSIG is defined by ISO as the worldwide standard for private networks.

QSIG is the generic name for a family of signaling protocols. The Q-reference point or interface is the logical point where signaling is passed between two peer entities in a private network. QSIG signaling can provide feature transparency in a single-vendor or multi-vendor environment.

quadrant
A group of six contiguous DS0s in fixed locations on an ISDN-PRI facility. Note that this term comes from T1 terminology (one-fourth of a T1), but there are five quadrants on an E1 ISDN-PRI facility (30B + D).

queue
An ordered sequence of calls waiting to be processed.

queuing
The process of holding calls in order of their arrival to await connection to an attendant, to an answering group, or to an idle trunk. Calls are automatically connected in first-in, first-out sequence.

R

RBS
Robbed-bit signaling

recall dial tone
Tones signalling that the system has completed a function (such as holding a call) and is ready to accept dialing.

recorded telephone dictation ready tone
A tone that indicates a dictation machine is connected to the telephone.

redirection criteria
Information administered for each telephone's coverage path that determines when an incoming call is redirected to coverage.

Redirection on No Answer
An optional feature that redirects an unanswered ringing ACD call after an administered number of rings. The call is then redirected back to the agent.

remote home numbering-plan area code (RHNPA)
A foreign numbering-plan area code that is treated as a home area code by the Automatic Route Selection (ARS) feature. Calls can be allowed or denied based on the area code and the dialed CO code rather than just the area code. If the call is allowed, the ARS pattern used for the call is determined by these six digits.

Remote Operations Service Element (ROSE)
A CCITT and ISO standard that defines a notation and services that support interactions between the various entities that make up a distributed application.

REN
Ringer equivalency number

reorder tone
A tone to signal that at least one of the facilities, such as a trunk or a digit transmitter, needed for the call was not available.

report scheduler
Software that is used in conjunction with the system printer to schedule the days of the week and time of day that the desired reports are to be printed.

RHNPA
See remote home numbering-plan area code (RHNPA).

ringback tone
A low-pitched repeating tone that indicates to the calling party that the dialed number has been reached successfully and is ringing.

RISC
Reduced-instruction-set computer

RLT
Release-link trunk

RMATS
Remote Maintenance, Administration, and Traffic System

RNX
Route-number index (private network office code)

RPN
Routing-plan number

RS-232C
A physical interface specified by the Electronic Industries Association (EIA). RS-232C transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet.

RS-449
Recommended Standard 449

ROSE
See Remote Operations Service Element (ROSE).

S

S1
The first logical signalling channel of DCP. The channel is used to provide signaling information for DCP's I1 channel.

S2
The second logical signaling channel of DCP. The channel is used to provide signaling information for DCP's I2 channel.

SABM
Set Asynchronous Balance Mode

SAC
Send All Calls

SAKI
See sanity and control interface (SAKI).

sanity and control interface (SAKI)
A custom VLSI microchip located on each port circuit pack. The SAKI provides address recognition, buffering, and synchronization between the angel and the five control time slots that make up the control channel. The SAKI also scans and collects status information for the angel on its port circuit pack and, when polled, transmits this information to the archangel.

SAT
System access terminal

SCC
1. See single-carrier cabinet.
2. Serial communications controller

SCI
Switch communications interface

SCO
System control office

SCOTCH
Switch Conferencing for TDM Bus in Concentration Highway

SDDN
Software-Defined Data Network

SDI
Switched Digital International

SDLC
Synchronous data-link control

SDN
Software-defined network

SID
Station-identification number

simplex system
A system that has no redundant hardware.

simulated bridged appearance
The same as a temporary bridged appearance; allows the telephone user (usually the principal) to bridge onto a call that had been answered by another party on his or her behalf.

single-carrier cabinet
A combined cabinet and carrier unit that contains one carrier. See also Multicarrier cabinet.

single-line telephone
A telephone served by a single-line tip and ring circuit (models 500, 2500, 7101A, 7103A).

SIT
Special-information tones

SMDR
See Station Message Detail Recording.

SN
Switch Node

SNA
Systems Network Architecture

SNC
Switch Node Clock

SNI
Switch Node Interface

SNMP
Simple Network Management Protocol

SPE
Switch Processing Element

SPID
Service Profile Identifier

SSI
Standard serial interface

SSM
Single-site management

SSV
Station service

ST3
Stratum 3 clock circuit pack

staffed
Indicates that a split position is logged in. A staffed split agent functions in one of four work modes: Auto-In, Manual-In, ACW, or AUX-Work.

Station Message Detail Recording (SMDR)
An obsolete term now called CDR — a switch feature that uses software and hardware to record call data. See Call Detail Recording (CDR).

standard serial interface (SSI)
A communications protocol developed for use with 500-type business communications terminals (BCTs) and 400-series printers.

status lamp
A green light that shows the status of a call appearance or a feature button by the state of the light (lit, flashing, fluttering, broken flutter, or unlit).

SVN
Security-violation notification

switch
Any kind of telephone switching system. See also communications system.

switchhook
The buttons located under the receiver on a telephone.

switch-processing element (SPE)
A complex of circuit packs (processor, memory, disk controller, and bus-interface circuit packs) mounted in a PPN control carrier. The SPE serves as the control element for that PPN.

SXS
Step-by-step

synchronous data transmission
A method of sending data in which discrete signal elements are sent at a fixed and continuous rate and specified times.

SYSAM
System Access and Administration

system administrator
The person who maintains overall customer responsibility for system administration. Generally, all administration functions are performed from the Management Terminal. The switch requires a special login, referred to as the system administrator login, to gain access to system-administration capabilities.

system printer
An optional printer that may be used to print scheduled reports via the report scheduler.

system report
A report that provides historical traffic information for internally measured hunt groups.

system-status report
A report that provides real-time status information for internally measured hunt groups.

system manager
A person responsible for specifying and administering features and services for a system.

system reload
A process that allows stored data to be written from a tape into the system memory (normally after a power outage).

T

T1

A digital transmission standard that in North America carries traffic at the DS1 rate of 1.544 Mbps. A T1 facility is divided into 24 channels (DS0s) of 64 kbps. These 24 channels, with an overall digital rate of 1.536 Mbps and an 8 kbps framing and synchronization channel, make up the 1.544 Mbps transmission. When a D-channel is present, it occupies channel 24. T1 facilities are also used in Japan and some Middle-Eastern countries.

TAAS

Trunk Answer from Any Station

TAC

Trunk-access code

tandem switch

A switch within an electronic tandem network (ETN) that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.

tandem through

The switched connection of an incoming trunk to an outgoing trunk without human intervention.

tandem tie-trunk network (TTTN)

A private network that interconnects several customer switching systems.

TCP/IP

Transfer control protocol/internet protocol

TDM

See time-division multiplexing (TDM).

TDR

Time-of-day routing

TEG

Terminating extension group

tie trunk

A telecommunications channel that directly connects two private switching systems.

time-division multiplex (TDM) bus

A bus that is time-shared regularly by preallocating short time slots to each transmitter. In a PBX, all port circuits are connected to the TDM bus, permitting any port to send a signal to any other port.

time-division multiplexing (TDM)

Multiplexing that divides a transmission channel into successive time slots. See also multiplexing.

time interval

The period of time, either one hour or one-half hour, that BCMS measurements are collected for a reports.

time-out tone

Tones that indicate the user failed to dial within the preset time interval after lifting the handset or after dialing the previous digit.

time slot
64 kbps of digital information structured as 8 bits every 125 microseconds. In the switch, a time slot refers to either a DS0 on a T1 or E1 facility or a 64 kbps unit on the TDM bus or fiber connection between port networks.

TOD
Time of day

toll-free service
A service that allows incoming calls from certain areas to an assigned number a flat-rate charge based on usage. The caller pays no toll fees.

tone ringer
A device with a speaker, used in telephones to alert the user.

TOP
Task-oriented protocol

trunk
A dedicated telecommunications channel between two communications systems or COs.

trunk-data module
A device that connects off-premises private-line trunk facilities and the system. The trunk-data module converts between the RS-232C and the DCP.

trunk group
Telecommunications channels assigned as a group for certain functions that can be used interchangeably between two communications systems or COs.

TSC
Technical Service Center

TTI
Terminal translation initialization

TTR
Touch-tone receiver

TTT
Terminating trunk transmission

TTTN
See tandem tie-trunk network (TTTN).

TTY
Teletypewriter

U

UAP
Usage-allocation plan

UCD
Uniform call distribution

UCL
Unrestricted call list

UDP
See Uniform Dial Plan (UDP).

UL
Underwriter Laboratories

Uniform Dial Plan (UDP)
A feature that allows a unique four- or five-digit number assignment for each telephone in a multiswitch configuration.

UNMA
Unified Network Management Architecture

UNP
Uniform numbering plan

USOP
User service-order profile

UUCP
UNIX-to-UNIX Communications Protocol

UUI
User-to-user information

V

VAR
Value-added reseller

VIS
Voice Information System

VLSI
Very-large-scale integration

VM
Voltmeter

VNI
Virtual nodepoint identifier

voice telephone
A single-line or multiappearance telephone.

W

warning tone
A low-pitched tone heard by all parties in a Busy Verification attempt that bridges to an active call.

WATS
See Wide Area Telecommunications Service (WATS).

WCC
World-Class Core

WCR
World-Class Routing

WCTD
World-Class Tone Detection

WFB
Wireless fixed base

Wide Area Telecommunications Service (WATS)
A service in the United States that allows calls to certain areas for a flat-rate charge based on expected usage.

wink-start tie trunk
A trunk with which, after making a connection with a distant switching system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

work mode
One of four states (Auto-In, Manual-In, ACW, AUX-Work) that an ACD agent can be in. Upon logging in, an agent enters AUX-Work mode. To become available to receive ACD calls, the agent enters Auto-In or Manual-In mode. To do work associated with a completed ACD call, an agent enters ACW mode.

work state
An ACD agent may be a member of up to four different splits. Each ACD agent continuously exhibits a work state for every split of which it is a member. Valid work states are Avail, Unstaffed, AUX-Work, ACW, ACD (answering an ACD call), ExtIn, ExtOut, and OtherSpl. An agent’s work state for a particular split may change for a variety of reasons (example: when a call is answered or abandoned, or the agent changes work modes). The BCMS feature monitors work states and uses this information to provide BCMS reports.

write operation
The process of putting information onto a storage medium, such as a hard disk.

WSA
Waiting session accept

Z

ZCS
Zero Code Suppression

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DEFINITY® Business Communications System and GuestWorks Issue 6 Overview
555-231-208, Issue 1, April 2000, Comcode 108596990

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