Lucent Technologies
Bell Labs Innovations

DEFINITY Business Communications System
Overview

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About This Document

Purpose

This document provides general information about the components and capabilities of the DEFINITY® Business Communications System. It discusses practical and creative solutions using the system.

Intended Audience

This document is written for those who are considering the purchase of the DEFINITY Business Communications System and for Lucent Technologies representatives and distributors who need high-level information about the system and how it can be used.

Content Overview

This document discusses all DEFINITY Business Communications System capabilities available world-wide. It defines common, practical solutions and suggests unusual, creative ones.

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

This overview includes the following information:

- *Chapter 1, "Introduction,"* — outlines basic advantages, capabilities, hardware and software components, and system configurations.

- *Chapter 2, "Desktop Solutions,"* — discusses features that are available at your desktop computer or telephone.

- *Chapter 3, "Computer-Telephone Integration Solutions,"* — discusses features that merge computer and telephone functions.

- *Chapter 4, "Data Management Solutions,"* — discusses features that help you manage telecommunications information.

- *Chapter 5, "Networking Solutions,"* — discusses features that help you network your equipment and solutions.

- *Chapter 6, "Multimedia Messaging Solutions,"* — discusses features that help you handle incoming and outgoing calls efficiently through voice messaging.

- *Chapter 7, "Multimedia Video Solutions,"* — discusses features that allow you to send and receive synchronized voice and image information.

- *Chapter 8, "System Management Solutions,"* — discusses the ways in which you can manage the DEFINITY Business Communications System and related systems.

- *Appendix A, "Features"* — summarizes the features discussed in this document; this appendix also describes the features that are unique to the system.

- *Appendix B, "System Capacity Limits"* — lists the system's capacities.

- *Appendix C, "Related Documents"* — lists and describes related documents.

A Glossary, list of Abbreviations, and an Index are also provided at the back of the document.

Conventions

The following conventions are used in this document:

- The term *system* is used in general to represent the DEFINITY Business Communications System product.

- The term *switch* is used to represent other telecommunications switching products.

Trademarks

The following trademarks and registered trademarks of Lucent Technologies are used in this document:

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- AUDIX$^{®}$
- DEFINITY$^{®}$
- MERLIN$^{®}$
- PassageWay$^{®}$
- Quorum™
- TERRANOVA$^{®}$
- Voice Power$^{®}$

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Related Documents

Please see Appendix C, "Related Documents" for a detailed list of document related to the system.

Introduction

<div style="text-align: right; font-size: 2em;">1</div>

The DEFINITY Business Communications System organizes and routes voice, data, image and video transmissions (Figure 1-1). 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.
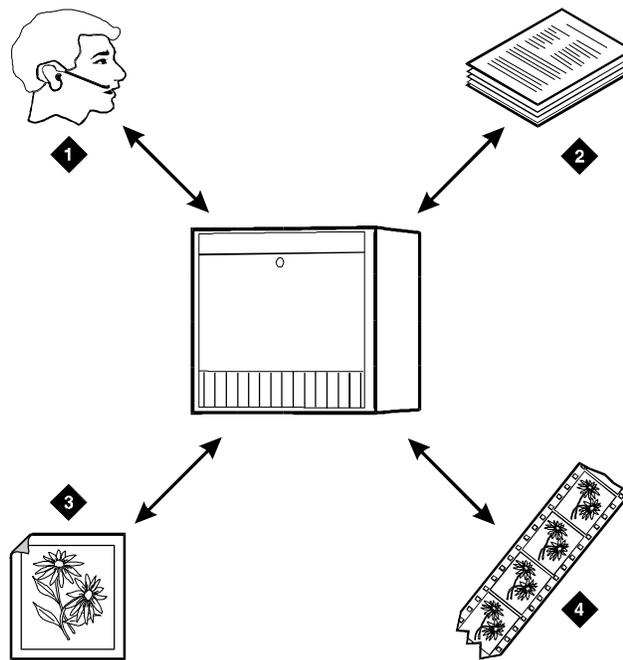
1)	Voice	3)	Image
2)	Data	4)	Video

Figure 1-1. DEFINITY Business Communications System

Advantages

The DEFINITY Business Communications System is an affordable telephony system that provides basic, single-site features on a global multimedia platform. It handles multimedia traffic as efficiently as any system available, while its state-of-the-art design further enhances Lucent Technologies' reputation for world-class reliability. The system accommodates, but also integrates, most related equipment throughout the world. Its modular design anticipates rearrangements and changes. All this translates to an exciting array of practical and creative applications for your business.

Adaptable
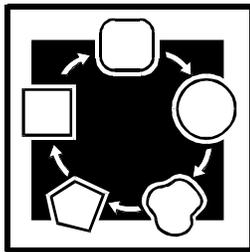
The DEFINITY Business Communications Systems' open architecture and modular design make it compatible with a wide variety of hardware and software — both Lucent Technologies tools and tools from other vendors. These may include personal and shared computers, terminals, computer networks, telephones, fax machines, and multimedia equipment. Multilingual options are available for messaging, administration, and most related applications. The system was designed to accommodate existing and anticipated global communications standards and protocols. It is adaptable to varying standards world-wide, providing efficient digital switching even when connected to conventional networks.

Expandable
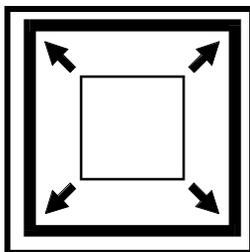
Modular port circuits, carriers (circuit shelves), and cabinets can be added to accommodate growth up to 400 stations. Each system can also be networked to other systems (Lucent Technologies systems or other types) to service many simultaneous voice, data, image and video transmissions.

Integrating
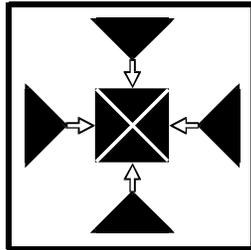
Sometimes the most important function of the system is its control and coordination of all your desktop tools and shared resources. It not only communicates with most networks and equipment throughout the world, but unifies them by translating protocols and standards as necessary. The system is designed to accommodate multimedia and network integration tools, in addition to many features that integrate computer and telephone. The system's integrating capabilities and its association with scores of leading-edge tools make it not only a good investment in itself, but it also enhances the value of your related investments.

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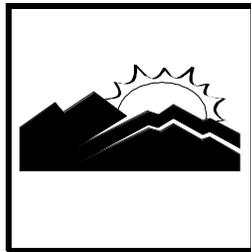
The system is inherently reliable by design. If something connected to the system should fail, the system keeps working until those systems are restored. If the disturbance is great enough that the system is disabled along with those other systems, the modular design provides you many options for getting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

State of the Art

The DEFINITY Business Communications System manages bandwidth precisely, which allows it to switch high bandwidth multimedia information while still conserving resources.

DEFINITY Business Communications System: A Global Multimedia Platform
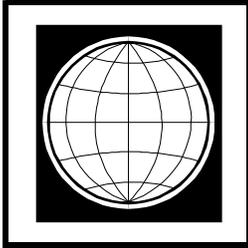
The DEFINITY Business Communications System is a global multimedia platform.

— It is compatible with a wide variety of tools, from group video systems to desktop network management applications and much more.

— It accommodates existing and emerging world-wide standards and protocols.

— Using bandwidth-on-demand, it provides the most efficient multimedia transmissions available.

— It offers multilingual options for most applications.

— It is designed to accommodate new innovations as they emerge.

— It is inherently reliable and provides many options for recovering quickly if disabled.

— It expands easily to accommodate your future needs.

World-class call features and multilanguage displays and voice prompts speed your communications with customers and associates around the globe. Messaging services enhance communication and productivity within your organization and enable business transactions across multiple time zones.

➡ 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.

For more detailed information on why the DEFINITY Business Communications System is the best choice for serving your international communications needs, see the following section.

Communications Around the World

Lucent Technologies is committed to making the features of the DEFINITY Business Communications System available globally. To meet that commitment, the system provides features that allow for differences in telecommunication standards around the world, allowing you to use the same communications system at your various locations in other countries. If you are reading this document, it is likely that the DEFINITY Business Communications System has been type-approved in your country. Check with your local distributor for more information.

Organizations that do business throughout the world are particularly interested in the following capabilities:

- Q-SIG Global Networking provides compliance with the European Computer Manufacturers Association Integrated Services Digital Network-Private Rate Interface specifications. This interface supports call setup, calling name display, and calling number display.

- The system provides flexible language displays, which allow you to select via administration the language in which messages are displayed on individual telephones.

- Telephone translation initialization, an administration feature, allows you to provide different levels of restriction and access based on who is using the telephone.

- Music-on-Hold for Analog Ports allows the music-on-hold device to be connected to analog line ports.

- World Class Routing allows flexible call routing for any type of national and international dialing plan, and consists of the following features and capabilities:

  — Flexible dialing

  — 18-digit routing

  — Automatic Route Selection (with International Direct Distance dialing calls and interexchange carrier access)

  — World Automatic Alternate Routing

  — Digit conversion

- Enhanced Integrated Services Digital Network capabilities include the following:

    — Support for either Basic Rate Interface A-law or Basic Rate Interface Mu-law companding.

    — European Computer Manufacturers Association system-to-system private networking support provides compliance with the European Computer Manufacturers Association Integrated Services Digital Network specification for basic call setup for systems in private networks.

    — Support of Integrated Services Digital Network slot maps to provide Integrated Services Digital Network capabilities in countries that require them.

    — The UK Networking feature which contains an ASN.1 Platform and three supplementary services. The European Computer Manufacturers Association specifications being used are 165 and 163/164. This feature is also consistent with other standards or agreements such as the European Telecommunications Standards Institute public network features and the Integrated Services Digital Network Private Network Specifications Forum agreements.

    — Primary Rate Interface can be carried over the PACCON instead of the processor interface board. This greatly increases the capacity of the Primary Rate Interface link, especially where Non-Facility Associated Signaling is not allowed or available.

- Digital signaling support is available for countries that require it.

- Generalized Multifrequency Compelling signaling is supported. Eighteen digits are supported on Multifrequency Compelling facilities for incoming calls. Multifrequency Russia 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 in order 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 in order 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 one-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 visually determine 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 when the numbering plan does not match the North American Numbering Plan.

- Call Detail Recording enhancements for periodic pulse metering provides periodic pulse metering pulse counts in the Call Detail Recording output record, using the pulses transmitted over trunk lines from the serving central office 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 features are described in greater detail throughout this document. See Appendix A, "Features" for a brief description of each feature available on the system.

Hardware

The Processor Port Network — a central processing unit that supervises system operation — is the main component of the system. All call processing is accomplished in this network; no additional hardware is required.

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.

Each cabinet contains 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.

The single-carrier cabinet (SCC) is the product design used for the DEFINITY Business Communications System. These cabinets are often connected together and can be stacked upon each other. They are often used by small businesses that are growing or expect to grow.
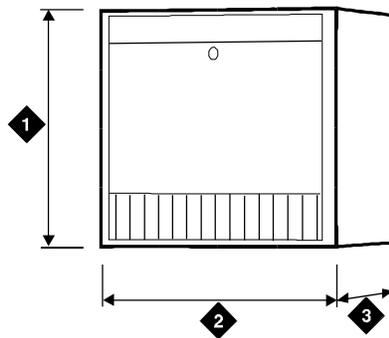
1)   51 centimeters   3)   56 centimeters

2)   69 centimeters

Figure 1-2.   Single-Carrier Cabinet

Figure 1-2 shows the front of a single-carrier cabinet with the door closed. Within each cabinet are port networks. A port network is a set of carriers served by a single Time Division Multiplexing/Local Area Network bus.

Single-Carrier Cabinets

There are two types of single-carrier cabinets used for the system:

- Control cabinet, which contains ports and the control complex (for call processing). There is a maximum of one control cabinet.

- Port cabinet, which contains ports. There is a maximum of two port cabinets.

Port Network/Carrier Summary

The relationships between port networks and carriers are summarized below.

The Processor Port Network

The Processor Port Network is the heart of the system. It contains the group of components that control, supervise, and coordinate system operations. Other components of the Processor Port Network can include port circuit equipment and packs for approximately 400 stations, depending on the configuration. The Processor Port Network uses 2 X 256 time slots to service attached equipment.

The Processor Port Network contains:

- One control carrier

- One or two port carriers (which can contain station, trunk, and service port circuit packs) in the remaining carrier positions

Switch Processing Element

The Switch Processing Element is the command center for the system. It is contained in the control carrier. It consists of the following major components:

- The processor is an Intel processor that is among the fastest available.

- There are ports provided to connect a PC using the TERRANOVA® terminal emulator, a monitoring terminal, and a dial-up modem.

- The tone clock supplies call-progress tones, dual-tone multifrequency signals, answer-back tones, trunk-transmission test tones, and system clock pulses. The tone-clock circuit pack includes either a Stratum 3 clock (optional) or combined tone detector and call classifier functions.

- A flash card storage system for customer translations.

- The packet control and packet interface provide communication and control links that go to the processor over the LAN bus. The interface supports up to 8192 ISDN D-channel signaling terminations.

- Data links to the Time Division Multiplexing bus and a link to the processor provide X.25 and ISDN protocol support (for services such as ISDN-Primary Rate Interface).

- The Switch Processing Element is contained in a single control carrier. This control carrier also has slots available for port circuit packs.

Port Carriers

The system supports a variety of trunks, telephones, data endpoints, as well as the attendant console. The following port circuit packs can be used in the system:

- *Analog line* — is supported in many configurations for many applications (basic phones, FAX machines, etc.).

- *Digital line* — for connecting digital telephones, attendant consoles, PassageWay®, and data modules that use Digital Communications Protocol.

- *Integrated Announcement* — for playing recorded announcements (messages) stored on the system to callers who are waiting in queue.

- *Auxiliary trunk* — for on-premises trunk applications to support features such as Music-on-Hold, Loudspeaker Paging, Code Calling, and Recorded Telephone Dictation.

- *Central Office trunk* — for loop-start or ground-start central office, foreign exchange, or Wide-Area Telecommunications Service (WATS) trunks.

- *Data line* — for equipment that uses Electronic Industry Association RS-232 connections and cables. The equipment must include asynchronous data units.

- *Direct Inward Dialing trunk* — for immediate-start or wink-start Direct Inward dialing trunks.

- *Direct Inward and Outward Dialing trunk* — for loop-start Direct Inward and Outward dialing trunks.

- *Basic Rate Interface line* — for connecting up to 24 ISDN-BRI-based digital telephones, Personal Computer/ISDN Platforms, and data modules on 12 2B+D interfaces.

- *24-Channel T1/DS1 interface* — for digital transmission of voice and data in the high-speed, high-capacity Data Signal Level 1 format. It supports Robbed Bit Signaling (RBS), 24th Channel Signaling, ISDN Primary Rate Interface message oriented signaling (ISDN PRI), D4 framing, ESF framing, AMI line coding (with or without Zero Code Suppression), and B8ZS line coding. Optionally, it can interface through the 120A integrated channel service unit, which is a compact, internal version of the cumbersome channel service units required between digital lines and a network. The board includes a test jack on the faceplate and, when the 120A is used, displays lights that indicate alarms and status.

- *32-Channel E1 interface* — for digital transmission of voice and data via connections to E1 lines. This board supports AMI line coding, HDB3 line coding, 75- or 120-ohm line termination, CRC-4 error checking, common channel associated bit oriented signaling (CAS), and ISDN PRI message oriented signaling. The board includes a test jack on the faceplate.

- *24/32-Channel E1/T1 interface* — optionally provides either the capabilities of the E1 interface or the T1 interface (described above).

- *Hybrid line* — for connecting hybrid and MERLIN® telephones.

- *DEFINITY® AUDIX®* — requires five slots in the Port Network for internal voice messaging capabilities (see Chapter 6, ''Multimedia Messaging Solutions'').

- *Maintenance/test* — for packet-bus maintenance and digital testing.

- *Multibutton Electronic Telephone line* — for connecting multibutton electronic-telephone sets.

- *Tie trunk* — for connecting automatic, immediate-start, E&M, wink-start, or delay-dial tie trunks, and for release-link trunks.

- *Call classifier/Tone detector* — has touch-tone and general-purpose receivers that detect call-progress tones, modem answer-back tones, transmission test tones, standard network tones, special information tones, and noise.

Configuration

The system hardware is configured in a single port network configuration (single processor port network). All functions are self-contained and no expansion networks are supported or required.

Reliability and Recoverability

The DEFINITY Business Communications System is designed to recover from a short power outage or other failure instantly, regardless of the source of the failure. Each port network includes a set of segmented, parallel buses. If one of the paired segments fails, the other bus segment continues to handle communications.

All I/O links also stay operational. Redundancy is built into the packet bus, and higher packet bus reliability can optionally be achieved by adding a maintenance/test circuit pack to each port network.

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

The system can be configured to meet the disaster recovery needs of any business. For example, the system's universal hardware and flexible software allow systems to be reconfigured quickly in emergency situations. Port networks can be added and network routing can be changed in a matter of minutes.

Connections to the DEFINITY Business Communications System

The DEFINITY Business Communications System can be connected to a wide variety of adjuncts, station equipment, and networks (Figure 1-3). This unparalleled flexibility allows you to acquire or design applications as your organization evolves.
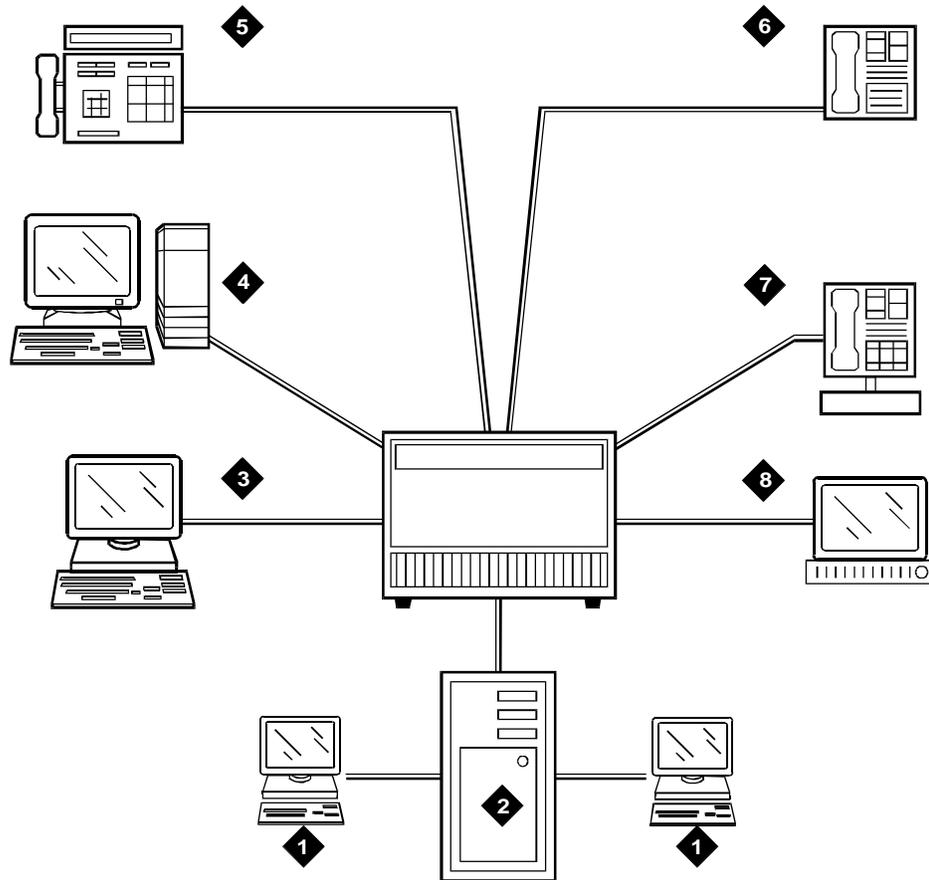
1)	Computers	5)	Attendant Console
2)	Host Computer	6)	Telephone (BRI, DCP, or analog circuit)
3)	Computer (BRI or analog circuit)	7)	Telephone with Data Module
4)	Manager Terminal (PC with TERRANOVA)	8)	Video Monitor for conferencing (BRI or DCP circuit)

Figure 1-3.   DEFINITY Business Communications System Connections

Adjunct Connections

In addition to station connections, the system includes many connections for adjunct (subordinate, related) equipment. The system provides an advanced X.25 (called BX.25) dedicated link adjunct interface, which can support a variety of adjuncts, including the DEFINITY AUDIX Multimedia Messaging System (internal).

The maintenance circuit pack provides two Electronic Industries Association (EIA) RS-232 ports: one for a PC using the TERRANOVA terminal emulator and one for a spare connection. In addition, a tip/ring connector with a built-in modem is provided for remote administration.

These RS-232 ports also work with the following typical adjuncts:

- Call Detail Recording Utilities
- Call Detail Recording printer
- System printer.

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 typical network interfaces:

- Electronic Tandem Network
- Integrated Services Digital Network-Primary Rate Interface.

Telephone Connections

All signals between analog telephones and the system are in analog form over a pair of wires. Digital DCP telephones using the Digital Communications Protocol 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.

The 8400 digital telephones automatically detect whether they are plugged into a two-wire or four-wire digital line circuit card. The 9400 digital telephones, also known as Europhones, provide inexpensive support for two-wire installations.

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.

Network Connections

Lucent Technologies is the fist vendor to provide compatibility with the Q-SIG global networking protocol. This means you can connect the DEFINITY Business Communications System with other switches throughout the world. Q-SIG Global Networking was developed to comply with the Q-SIG standards developed by the European Computer Manufacturer's Association and the International Standardization Organization. It supports the ISDN-Primary Rate Interface 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-Primary Rate Interface, 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 complete ISDN support for anywhere from 64 to 400 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 5, "Networking Solutions".

Power

The system can be powered with either 120/208/240V AC (50/60Hz) power or 48 V DC power. When commercial AC power is not available, the system will operate off DC power supplied by battery reserve.

An uninterruptible power supply can protect the system from voltage lags, over-voltage conditions, line frequency fluctuations, and power blackout of short duration. A battery backup system can be used to provide power for up to 8 hours, depending on the type and quantity of circuit packs and amount of traffic during the holdover period.

Software

All DEFINITY Business Communications Systems (and related Lucent Technologies systems) throughout the world use the same basic software. 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.

The DEFINITY Business Communications System provides a wide array of features. See Appendix A, ''Features'' for an exhaustive listing of features available on the system.

Desktop Solutions

2

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 high speed data communications and access to a variety of host and personal computers.

The DEFINITY Business Communications System 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 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.

Voice Features

With the DEFINITY Business Communications System, 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

Allows you to dial frequently called or emergency numbers with just a few button presses instead of dialing the entire number one digit at a time. You can use Abbreviated Dialing to dial both internal and external numbers of up to 36 digits. Many telephones also allow you to program abbreviated dialing buttons so you can dial frequently dialed numbers with just one button press.

Conference

Allows you to set up a conference call with up to six people. Anyone in the world with access to a telephone can participate in your conference. The Conference button on your telephone allows you to set up the conference call without the aid of an attendant.

Integrated Announcements

The system allows you to store recorded announcements (messages) internally. The announcements are digitized and stored in state-of-the-art electronic memory devices. The system's integrated announcements are as follows:

- Easy to use. Announcements can be recorded and updated from any telephone. And all announcement configuration is performed from the management interface (usually a PC using the TERRANOVA terminal emulator).

- 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 Direct Access Calling were designed to take advantage of the power of integrated announcements.

- Ideal for a global market. Since you record your own announcements, any language can be provided — even multiple languages on the same system.

- Easily expandable. In larger configurations, if you need more recording time, you can simply plug in another announcement circuit pack.

Integrated Directory

Either through voice messaging or a display telephone, the system allows you to access the names directory and retrieve an extension number. The directory contains an alphanumeric listing of all names and extension numbers connected to the system. The directory can be set up using several languages. (Contact your local distributor for a complete list of languages.)

Last Number Dialed

Allows you to redial your last call, whether it is an internal or an external call. You can press a single button to redial a number having up to 20 digits.

Leave Word Calling

Allows you to, with the touch of a button, leave a standard message ("Call me back," for example) for others on the same system.

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 a versatile secretary 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 telephone line, the system considers you "active." If you are using all your available lines, the system considers you "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 an assistant.

Call Coverage lets you redirect calls to suit any or for all of these criteria. For each telephone, you can have up to four coverage paths. A path is a set of alternate extensions that a call can be sequentially transferred to. 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 dialing an access code.

Coverage Paths for a Manager

Figure 2-1 shows four coverage paths you might need as a manager. The example assumes you:

- Receive many external calls

- Share a secretary 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.
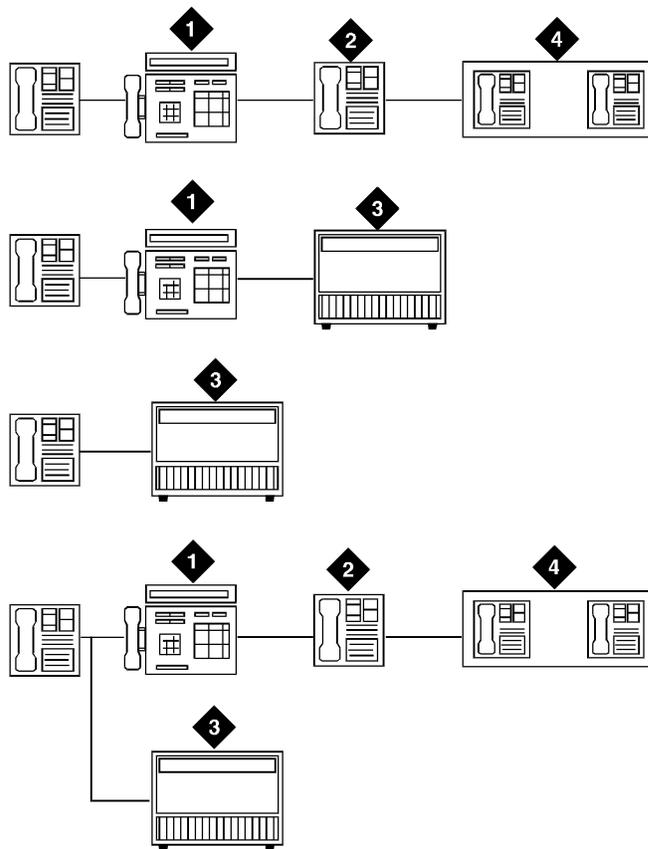
A)	External Calls: Active, Busy, Don't Answer	1) Secretary
B)	Internal Calls: Cover All	2) Clerk
C)	Internal Calls: Active, Busy, Don't Answer	3) AUDIX Voice Messaging
D)	Internal Calls: Send All Calls	4) Message Center Group

Figure 2-1.   Typical Call Coverage Options

Messaging Services

The DEFINITY Business Communications System offers the DEFINITY AUDIX voice messaging that allows you to leave, send, and receive messages quickly accurately, and conveniently. This messaging service 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 an AUDIX system is set up as the last point on a call-coverage path, as in Figure 2-1 above. 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. 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. You can even have the AUDIX automated attendant answer all calls to the company and 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 in a variety of ways, such as:

- 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, employees can dial *Speak-to-Me* and hear a synthesized voice read their messages over the phone.

These message-retrieval options can be assigned to users individually.

Telephones and Workstations

A wide variety of telephones are available with the DEFINITY Business Communications 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 terminal 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, and 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 DCP telephones using the Digital Communications Protocol 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 DEFINITY Business Communications System's family of ISDN telephones include several models which include unique features such as call logs and personal directories.

New Telephones for the Global Marketplace

With help from our many global customers, Lucent Technologies has developed new telephones, the 8400-series, the 9100-series, and the 9400-series, to meet the demand for two-wire telephones in the global marketplace.

8400-Series Telephones

The 8400 digital telephones are versatile two-wire/four-wire Digital Communications Protocol (DCP) telephones with new styling that offer new flexibility and cost savings. They automatically detect whether they are plugged into a two-wire or four-wire digital line circuit card. This is a significant benefit because it provides an easier transition to either a two-line or a four-line environment, therefore reducing wiring expenses and installation adjustments. It also allows you to save space inside the cabinet by using 16-port two-wire boards in place of 8-port four-wire boards.

In response to customers' requests, the handset has a larger mouthpiece, the telephone has raised buttons that provide improved tactile feel for easier use, and the finish is a scratch-resistant texture that preserves the telephone's appearance. Programmable speakers and microphones can be turned on or off by the system administrator.

There are several models of 8400-series telephones:

- 8403 — a three-line telephone without a display that can be wall mounted. This entry-level telephone is designed for users with basic call handling requirements. 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 off-hook dialing and listening to voice mail or broadcast messages. You can add a S201A speakerphone and use the inexpensive 8403 in a conference room.

- 8405B — a five-line telephone without a display that can be wall mounted. This telephone is for employees with call coverage responsibilities who need multiple line appearances and extensive features. It has a built-in speaker (standard models) or a two-way speakerphone (the "plus" models), and programmable keys so you can access more system features from the telephone.

- 8405D — a five-line telephone with a two-line, 24-character display. This telephone has the same features as the 8410B and can also be wall mounted.

- 8410B — a ten-line telephone without a display that can be wall mounted. This telephone is for employees with call coverage responsibilities who need multiple line appearances and extensive features. It has a built-in two-way speakerphone and programmable keys so you can access more system features from the telephone.

- 8410D — a ten-line telephone with a two-line, 24-character display. This telephone has the same features as the 8410B and can also be wall mounted.

- 8411B and 8411D — enhanced versions of the 8410B and 8410D terminals that provide a built-in RJ11C jack as an interface to analog telephone devices (such as a telecopier or a modem), and an RS232 data interface the PassageWay Direct Connection. These sets cannot be wall mounted.

- 8434DX — a thirty-four-button set with a 2-line by 40-character display. This telephone is for the busy executive or executive assistant where extensive call handling and call coverage flexibility are vital. The 8434DX has a built-in two-way speakerphone and programmable keys. A 24-button expansion module can be added.

Here are the most important features of the 8400 telephones:

Administering Speakerphones. You can administer speakerphones either through the system or through the telephone. The 8505B, 8405D, 8410B, 8410D, and 8434DX telephones can be administered as two-way speakerphones or one-way listen-only speakers, or you can disable the speakers. The 8403 can be administered as a one-way listen-only speaker or the speaker can be disabled.

Compatibility and Investment Protection. The 8400-series digital telephones are compatible with all two-wire DCP circuit packs, as well as with all earlier four-wire circuit packs. The backward and forward compatibility of these telephones protects the investment you made in your existing wiring and your existing older version four-wire systems.

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.

Doubled Capacity. When the 8400-series telephones are coupled with the new two-wire 16-port Digital Line Circuit Card, you benefit by having more capacity in each carrier. Therefore purchase of additional carriers or cabinets may not be necessary.

9100-Series Telephones

The 9100-series analog voice terminals are cost-effective and combine improved features and greater flexibility in an up-to-date, more contemporary design. Available in three models (9101, 9103, and 9110), the 9100-series analog voice terminals are more practical, more economical, and the perfect choice for anyone or any location in your business. Table 2-1 lists the features available on the different 9100-series analog voice terminals.

Table 2-1.   9100-Series Analog Voice Terminal Features

Feature	Model 9101	Model 9103	Model 9110
Data Jack	No	Yes	Yes
Flash Button	Yes	Yes	Yes
Message Waiting Lamp	Yes	Yes	Yes
Mute Button	No	No	Yes
On-Hook Dialing	No	Yes	No
Pause Button	No	Yes	Yes
Positive Disconnect	Yes	Yes	Yes
Redial	No	Yes	Yes
Repertory Dialing	No	Yes (13)	Yes (20)
Ringer Volume Control	Yes	Yes	Yes
Speakerphone	No	No	Yes
Tone/Pulse Switch	Yes	Yes	Yes

9400-Series Telephones

The 9400-series digital telephones, also known as Europhones, provide inexpensive support for two-wire installations, while still providing a European design. Three models of the 9400 telephones are available in gray and cream white. The 9403, 9410B, 9410D, and 9434 telephones are similar in design and features to the 8400-series.
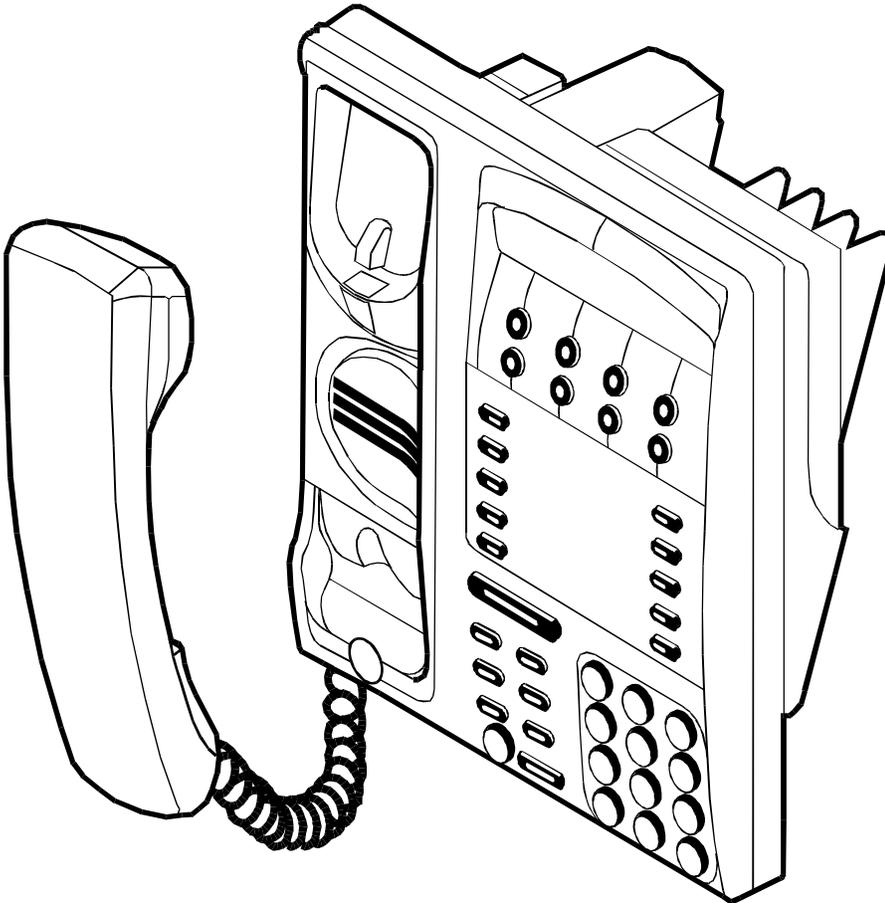
Figure 2-2.   9400-Series Telephone

Here are the most important features of the 9400-series telephones:

International Language Support. The new 9400 digital telephones are available with labels and user's guides in several languages. Because the European requirements for the lettering on the keypad are not identical, Lucent Technologies has created an overlay that holds the necessary lettering for each country. Two overlays are available: one blank and one with letters, complying with CCITT standards.

Compatibility. Like the 8400 telephones, the 9400 telephones are compatible with all 2-wire circuit packs. The connection is made via a two-wire, 16-port DCP interface card.

Solid Engineering. Compliance with the most rigid emission and electromagnetic requirements ensures the integrity of your installation.

Teleconferencing Products

How much of your business 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, easier to schedule, and travel expenses are greatly reduced. The Lucent Technologies Quorum™ and SoundStation products provide you with all the benefits of voice conferencing.

Quorum A-28 Conference Bridge

The Quorum A-28 Conference Bridge is a microprocessor-controlled analog bridge. It allows you to activate a multipoint connection of switched and private-line circuits to create a single conversation. You can set up, monitor, and control your own teleconferences through your system.

You can set up and supervise the bridge connections through an attendant console. Participants can dial into a prearranged conference at a specified time. The calls are then connected automatically, or can be screened by the attendant before being added to the conference. The attendant can also monitor the transmission quality of any or all of the active conference links. Poor connections can be identified, isolated, and redialed without interrupting the rest of the conference.

The bridge design ensures optimum sound clarity on the conference no matter how many participants are connected or where they are located. The bridge circuitry senses differences in signal strength and electronically balances voice levels. It also filters any extraneous noises that could interfere with the conference.

The Quorum A-28 Conference Bridge allows you to connect up to 28 different people on a single conference call. Or you can set up multiple conferences — up to four conferences with seven participants each.

The bridge also provides a lecture mode that you can use for one-way broadcasts. Using the Mute feature, participants can be placed in a listen-only mode, allowing one speaker to address the rest of the conference.

SoundStation Audio Equipment

Lucent Technologies' SoundStation and SoundStation EX speakerphones 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 strong or soft, or from a standing or sitting position. Integrated components and a stylish tripod design make the console an attractive yet unobtrusive conference table centerpiece.

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.

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 a conference room. It connects to an analog telephone line. The built-in keypad includes a mute button an flash key. An additional port allows you to connect the speakerphone to a tape recorder.

The system is simple to install and use. You plug the phone line into a small wall module plugged into an 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 on either side of the center console. An optional lapel microphone is available for stand-up presenters.

Computer-Telephone Integration Solutions

3

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

If these two building blocks are closely integrated, your business will realize benefits that will redefine your standards for success and customer satisfaction. The DEFINITY Business Communications System integrates data processing, data communications, and voice communications.

The following products work with the DEFINITY Business Communications System to unite your computer and telephone in powerful ways:

- PC Console

- PassageWay.

➡ **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.

PC Console

Lucent Technologies' PC Console allows your call attendants to handle incoming calls efficiently by personal computer. Using the familiar Microsoft[*] Windows[*] graphical interface, the attendants can easily keeping track of how long callers have been on hold and who they are waiting for. Attendants can monitor up to six calls at once. They need not fumble with pen and paper when handling calls, as 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 system also accommodates any language that uses the Roman alphabet and ASCII 128 character set. If a Spanish-speaking attendant takes over for a French-speaking attendant, for example, a single press of a button converts all labels, error messages and on-line help to Spanish.

[*]   Registered trademark of Microsoft Corporation.

PassageWay

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 DEFINITY Business Communications 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.

PassageWay Direct Connection

PassageWay Direct Connection links your company's desktop personal computers with an easy-to-use Microsoft Windows interface to give you greater business communications capabilities than either the telephone or the personal computer offer.

PassageWay Direct Connection provides valuable computer-telephone integration benefits, plus it is a platform bridge to a wealth of other computer-telephone integration applications. Open Application Programming Interface support and Windows Dynamic Data Exchange support allows independent software vendors or internal software developers to create new computer telephone integration applications or to enable existing applications to be interfaced to the telephone. These independent software vendors products utilize the PassageWay platform to expand the power and flexibility of computer telephone integration at the desktop.

The PassageWay Direct Connection software applications are for the individual desktop personal computer. However, using the Application Programming Interface, Dynamic Data Exchange, or Independent Software Vendors products, PassageWay Direct Connection can be linked to the your Local Area Network.

PassageWay Direct Connection is well suited for those users who are constantly conducting business using both the desktop Windows personal computer and a telephone, and want to boost their productivity and efficiency.

Here is a list of the computer telephone integration software applications included with each PassageWay Direct Connection product:

- Lucent Technologies Call is a software card-file database that allows a customer's record to be previewed, the appropriate telephone number to be autodialed and notes to be taken quickly all with just a few mouse clicks.

- Lucent Technologies Buzz instantly retrieves and displays the Lucent Technologies Call record associated with an incoming call based on the Calling Party's Telephone Display Information (Automatic Number Identification, Call Prompting, Conversant). Lucent Technologies Buzz also allows users to answer incoming calls from their personal computer with only a single mouse click.

- Log Viewer automatically maintains a record of every call either made from Lucent Technologies Call or received from Lucent Technologies Buzz.

Passageway Direct Connection supports those international companies who need A-law support and/or 8400-series telephones, which allow both two-wire and four-wire connections.

The following requirements must be met for PassageWay Direct Connection to function properly:

- An IBM Compatible personal computer with the following:

    — 386 or higher microprocessor

    — Two megabytes of RAM (4 MB recommended)

    — Two megabytes of hard disk space

    — 3.5" or 5/25" disk drive

    — An available serial port

    — Mouse or trackball

    — VGA (or higher resolution) monitor.

- Microsoft Windows 3.1 or higher (in standard or enhanced mode) or Windows for Work Groups 3.11 or higher

- A DEFINITY Business Communications System and one of the following telephones:

    — 7400-series telephone

    — 8400-series telephone

    — 9400-series Europhone.

- Local adjunct power (or closet power) for the PassageWay Direct Connection device

PassageWay FastCall for Direct Connection

PassageWay Fast Calling Feature for Direct Connection is a Windows-based application which was developed by Aurora Systems for a variety of Lucent Technologies computer telephone integration platforms including PassageWay Direct Connection, PassageWay Telephony Services, and PC/Switch. The Fast Calling Feature delivers multiple computer telephone integration capabilities for Call Center and other computer users workers in your company. These capabilities include:

- Instant account information on screen (based on Calling/Called Party and/or Caller Input Identification)

- Coordinated Voice and Data Transfer

- Outbound Preview Dialing

- Inbound Call Handling Rules (user-defined call coverage)

- Personal computer-based telephony (activation of Conference, Transfer, Drop, and Hold from the personal computer).

The Fast Calling Feature resides between PassageWay and any Windows-based application on your Local Area Network, desktop personal computer, or mainframe. This approach allows existing applications to be simply and quickly "telephony enabled" without the need for low-level software development. The Fast Calling Feature takes advantage of simple keyboard recorded scripts and macros.

The primary users of the Fast Calling Feature would be employees who use a personal computer in their work. If these users want to automate their existing Windows application to perform one of the five tasks listed above, Fast Calling can enable them to do just that.

PassageWay Telephony Services

PassageWay Telephony Services connects the DEFINITY Business Communications System and Novell[*] Networks. Developed in partnership with Novell, it allows you to integrate a variety of communications systems and software in one network. Its open architecture is based on the ECMA CSTA international standard. Besides standard Windows environments, PassageWay Telephony Services supports Windows NT, OS/2, Macintosh[†], and UnixWare[*] operating systems on a Novell NetWare[*] LAN. The Telephony Services Applications Programming Interface allows you to coordinate the applications on the network. The programming interface is supported by 30 switch vendors.

The Bulk Administration feature allows you to use existing databases to provision the telephony system. It allows you to create filters for integrating the information the system requires. It also allows you to use a variety of software for manipulating the data. The LAN Traffic Measurements Utility allows you to measure LAN traffic related to telephony.

The PassageWay Telephony Services installation programs and documentation are provided in U. S. English, French, German, Spanish, and Italian. See your local distributor for information on whether other languages are available.

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[*]   Registered trademark of Novell, Inc.

[†]   Registered trademark of Apple Computer, Inc.

Data Management Solutions

4

The DEFINITY Business Communications System is 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 DEFINITY Business Communications 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.

- The jacks, twisted-pair wiring, and optical fibre used by the premises distribution system contribute to easy installations and easy moves.

- 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. And the system's management capabilities can monitor and control your data communications.

The DEFINITY Business Communications 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

- Local area network bridging

- Information System Network 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 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.

Data Management Features

The DEFINITY Business Communications System offers a number of data management features to help control your data environment and 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 establish an end-to-end connection between two data endpoints. 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 very 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 Hotline 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 DEFINITY Business Communications System offers powerful digital interfaces for high-speed 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.

- Digital Multiplexed Interface is a high-speed, economical interface for terminal-to-host, system-to-host, and host-to-host communications.

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.

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[*]   Registered trademark of AT&T.

Digital Communications Protocol

Digital Communications Protocol (DCP), a forerunner of ISDN-Basic Rate Interface, has been the architectural foundation for both 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 for signaling and control information (or data channel). Out-of-band signaling via the data channel allows the information channels to be used for clear-channel transmission.

DCP's framing structure allows voice, data, and signaling information to be transmitted with low overhead and virtually free of errors. DCP transmits at a rate of 8,000 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.

Digital Multiplexed Interface

The DEFINITY Business Communications System supports high-speed, efficient data communications through a Digital Multiplexed Interface. Digital Multiplexed Interface allows communications between a terminal and host computer, between a system and host computer, and between two host computers.

Digital Multiplexed Interface is consistent with ISDN-PRI. 24 64-kbps channels (one of which is reserved for signaling information) transfer data between two endpoints through the system. The multiplexed channels can be sent over standard Digital Signal Level 1 facilities, allowing the host computer to be located remotely from the system.

Digital Multiplexed Interface offers two major advantages. It delivers a standard, single-port interface for linking host computers internally and externally via T1 carrier. And, since it is compatible with ISDN standards and is licensed to numerous equipment manufacturers, it promotes multivendor connectivity.

The system supports two versions of Digital Multiplexed Interface, each differing in the way information is carried over the 24th channel:

- Digital Multiplexed Interface-bit-oriented signaling carries framing and alarm data and signaling information for connections to host computers and other vendor equipment.

- Digital Multiplexed Interface message-oriented signaling, fully compatible with ISDN-PRI, uses the same message-oriented signaling format, Link Access Procedure on the D-channel, as ISDN-PRI for control and signaling. These signaling capabilities extend the advantages of Digital Multiplexed Interface-Message Oriented Signaling multiplexed communications to the public ISDN network.

Both Digital Multiplexed Interface-Bit Oriented Signaling and Digital Multiplexed Interface-Message Oriented Signaling use 23 information (bearer) channels and one signaling (data) channel multiplexed on a 1.544-Mbps Digital Signal Level 1 carrier link.

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 Core Basic Rate Interface (BRI)

World Class Core 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 new 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 types of data module:

- High Speed Links

- Data stands

- Modular-processor data module

- 7000-series data modules

- Modular-trunk data module

- Asynchronous data unit

- Asynchronous data module (for ISDN-Basic Rate Interface telephones)

- Terminal adapters.

All of these data modules support industry standards and include options for setting the operating profile to match that of the data equipment. The data modules that are currently available with the system are described below.

High Speed Links

The High Speed Link, a DCP-based data module, lets you transmit data through your system at faster speeds, allowing you to take advantage of emerging technologies such as video conferencing and Local Area Network bridging.

More and more companies are realizing that technologies and applications, such as Local Area Network-to-Local Area Network communication, video conferencing, file transfer, and Group 4 fax transmissions are necessary to compete effectively in a global market. The High Speed Link offers you a cost-effective way to manage these applications.

Used when integrated voice and data is not required, the High Speed Link has an internally timed V.35 interface for synchronous data transmissions at 56 kbps (half and full duplex) and 64 kbps (full duplex) in both switched and permanent connections.

It offers access to low-cost, dial-up communications and provides a link to high-speed network services such as AT&T's ACCUNET Switched Digital Services and Software Defined Data Network Services. When accessing these network services, the High Speed Link can communicate with applications terminating on either a digital service unit or another High Speed Link.

Its numerous testing features make fault isolation easy. For example, the High Speed Link offers a variety of client-or network-initiated loop-back tests that increase reliability, such as the capability to isolate problems remotely.

There is an interface that can be configured as an RS-366 Automatic Calling Unit or an RS-232 asynchronous data interface that supports a limited AT command set for call control.

All options are software definable and stored in nonvolatile memory. The reset options feature makes it easy to load default options. Designed to be easily upgraded, the High Speed Link has a memory cartridge interface for firmer upgrades to support new features.

The unit is externally powered, and it can be rack mounted with up to eight units and located up to 5000 feet from the system.

7000-Series Data Modules

The 7000-series data modules are designed to give you simultaneous voice and data access in a single, low-cost data module. The single DCP connection means you will never miss a voice call when you are on a data call.

The 7400B Plus and 7400A dual-function data modules provide full-duplex, asynchronous connectivity for DCP applications. They emulate the industry-standard Hayes[*] modems and work with host-connection software packages that use the Hayes command set. Priced competitively with Hayes-compatible modems (that operate at 1200-2400 bps), the modules give you a choice of transmission speeds ranging from 300 bps to 19.2 kbps.

The 7400B Plus provides integrated, simultaneous, voice/data communications over twisted-pair wiring. It is easy to install and operate and plugs into any modular DCP telephone outlet. It features two ports: a modular DCP port for a digital telephone and an RS-232 port for a personal computer or computer.

In desktop configuration, the 7400B Plus provides twisted-pair connectivity for personal computers and asynchronous computers at transmission speeds ranging from 300 bps to 19.2 kbps. It operates with all 7400-series digital telephones. On the trunk side of the system, the 7400B Plus can provide connectivity to asynchronous host computers at speeds as high as 19.2 kbps.

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.

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[*]   Registered trademark of Hayes Microcomputer, Inc.

Asynchronous Data Unit

The asynchronous data unit offers an economic alternative to data modules for connecting Electronic Industries Association RS-232 data endpoints to the system. The Asynchronous Data Unit extends the 50-foot limitation of an RS-232 interface cable up to 40,000 feet, depending on the data speed and wire gauge of the distribution system. For example, a 19.2-kbps data rate can be supported to 2,000 feet. This allows RS-232 devices (such as terminals, host computers, multiplexers, printers, and personal computers) to be located up to 40,000 feet apart for private network applications. The Asynchronous Data Unit handles standard data rates from 300 bps to 19.2 kbps and nonstandard asynchronous data rates below 1,800 bps. It provides asynchronous full-duplex operation.

The multiple asynchronous data unit is a circuit board that contains eight asynchronous data unit circuits housed in one unit. These are typically used in computer-room applications where several RS-232 connections are carried in a common cable from the host computer.

The Asynchronous Data Module is used in conjunction with the 7505, 7506, or 7507 ISDN-Basic Rate Interface telephone to support integrated voice and data. With the Asynchronous Data Module, computers or personal computers attached to the telephone can send and receive data through the system. The Asynchronous Data Module mounts in the base of a 7500-series ISDN-Basic Rate Interface telephone and provides asynchronous full-duplex operation at data rates up to 19.2 kbps.

Networking Solutions

5

The DEFINITY Business Communications System provides 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 DEFINITY Business Communications 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 Q-SIG global standard.

> **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.

Q-SIG Global Networking

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

This initial release of Q-SIG Global Networking is a foundation for the European Computer Manufacturers' Association supplementary services. It provides the call setup message and the Name and Number Identification supplementary service.

World Class Routing

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

World Class Routing is a powerful enhancement to the system's call-routing capabilities, linking 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 domestic dialing plans.

The following are key components of World Class Routing:

- Digit Conversion converts a dialed number for public network number to 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 the same routing abilities. In addition, there are a number of capabilities that enhance the flexibility of routing in supporting your domestic 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, 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 format 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, initially dialed to use Automatic Route Selection, 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 DEFINITY Business Communications System has a variety of features that enable you to manage your network resources effectively. Here are just a few examples of features that can be used to manage your network — Time-of-Day Routing, Automatic Route Selection, Facility Restriction Level, Remote Network Access, Authorization Codes, and Automatic Alternate Routing.

Time-of-Day Routing

Time-of-Day Routing allows you to select the most economical routing of Automatic Route Selection and Automatic Alternate Routing calls based on the time of day and the day of the week a call is made.

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 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. You can set up eight separate time-of-day charts to control routing at different times of the day.

Automatic Route Selection

Automatic Route Selection 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 not facility is available, the call can be queued until a facility becomes available.

Automatic Alternate Routing

Automatic Alternate Routing enables you to ensure that private network calls will be routed over the various trunking facilities available in your private network in the most effective manner possible. As with Automatic Route Selection, you set up various patterns for routing calls — in this case, with the private network. You can have up to 40 routing patterns. Each pattern includes a primary preference — the most preferred and direct route — and 15 alternate preferences. If the primary preference in a pattern is unavailable, the system searches the alternate preferences in the specified order until it finds one available.

Subnetwork Trunking

Subnetwork trunking is an Automatic Alternate Routing/Automatic Route Selection function that typically converts an on-network (private network) number to a public network number (based on patterns and preferences) for off-network routing. It can also convert a public network number to a private network number. Subnetwork trunking provides digit insertion, deletion, pauses, and/or wait for dial tone in digit outpulsing, as required, to permit calls to route to or through a remote system, over tie trunks to a private network system, or over central office trunks to the serving central office.

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 select the appropriate facility based on the type of call. Generalized Route Selection gives you the capability to not only select the optimal call routing based on the dialed number, but also 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 the decision of how a call is 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 DEFINITY Business Communications System also provides the capability to route calls based on the data format and the need for restricted or unrestricted facilities.

In order to select the appropriate trunking facility for a call, the system must know the type of call being made. In order 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. This makes life easier for your users, who do not have to worry about dialing a different access number for different call types.

Facility Restriction Level

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. Eight levels of Facility Restriction Levels can be assigned to telephones, computers, and trunk groups, using the system management tools. The system does not require the Facility Restriction Level to be in an 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 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 1,500 different authorization codes can be provided for your system at any one time. Using 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 DEFINITY Business Communications 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.

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

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

- 800-service trunks which let your business pay the charges for inbound long-distance calls so that callers can reach you toll-free

- 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.

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.

Miscellaneous Trunks

Remote-access trunks provide off-premises users with access to system features and networking.

Digital Interfaces

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 boards).

Digital Services 1 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 economic 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 DEFINITY Business Communications System offers several options in supporting the Digital Signal Level 1 interface. The options include support for voice-grade Digital Signal Level 1, alternate voice/data, and Digital Multiplexed Interface. 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 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-premise stations, and dedicated voice/data system connections

The system's DS1 interface capabilities include support for Digital Multiplexed Interface.

To achieve even greater benefits than those just listed, you can combine the DS1 interfaces and ISDN-Primary Rate Interface to give you additional capabilities. ISDN-Primary Rate Interface 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-Primary Rate Interface, 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 DEFINITY Business Communications System delivers Automatic Restoration capability with Software Defined Data Network, which restores disrupted connections between access endpoints (non-signaling trunk) and data endpoints (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.

Stratum 3 Clock

Many companies have solved their communications network needs by using high-speed digital facilities. However, many have found that standard synchronization timing systems are not accurate enough in these situations. Standard systems allow more slippage than is desirable for operations like high-speed bulk data transfer. The result is bit and frame losses which leads to the expense of lost data.

Stratum 3 clock is an optional external hardware adjunct for the system that is more accurate than standard offerings. Stratum 3 clock has a maximum of 2.5 minutes of allowable error per year. The standard clock used in the system and many systems is a stratum 4E clock, which allows approximately 17 minutes of error per year.

ISDN

The DEFINITY Business Communications System provides complete ISDN support for systems with 100 to 400 telephones. 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 — Primary Rate Interface and Basic Rate Interface.

- Primary Rate Interface is used for connecting premises equipment to the network, and acts as a powerful interface between intelligent equipment such as systems and computers.

- Basic Rate Interface is used for connecting telephones, computers, personal computers, and other desktop devices to higher-order equipment such as a system.

Both Primary Rate Interface and Basic Rate Interface are based on the same common building blocks — the use of a common interface to a transmission path that is divided into channels. Both Primary Rate Interface and Basic Rate Interface 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.

Primary Rate Interface, 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. Basic Rate Interface, 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 endpoint to endpoint 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 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 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 DEFINITY Business Communications System's support of ISDN-Primary Rate Interface, 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. For example, two systems connected by Primary Rate Interface 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.
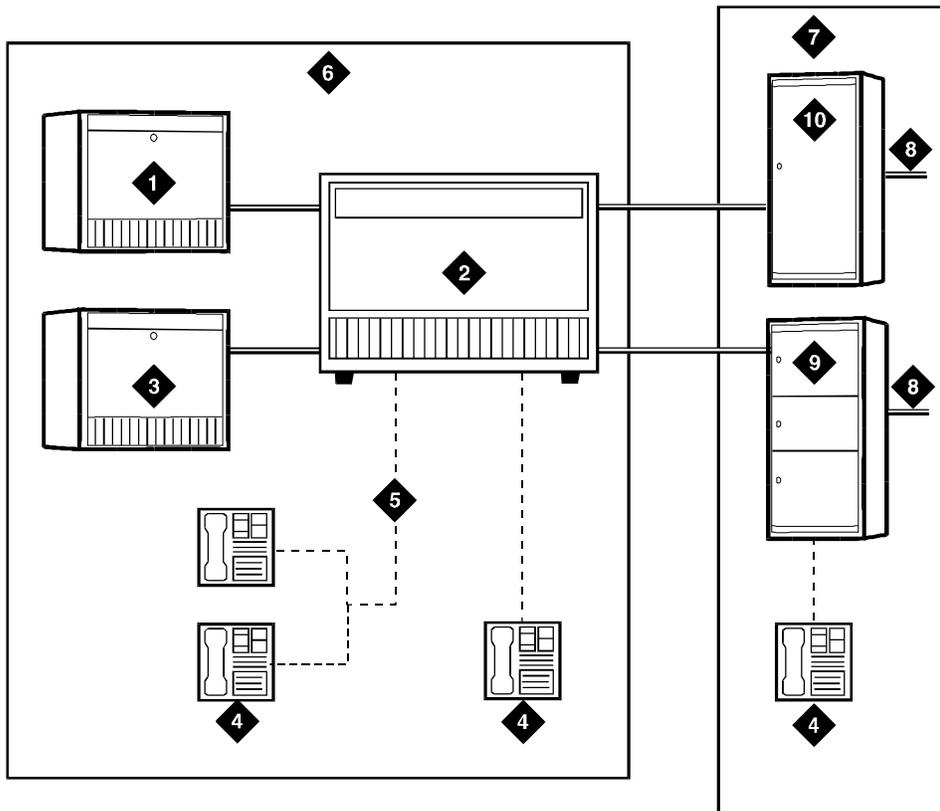
1)	DEFINITY Business Communications System	6)	Private ISDN
2)	DEFINITY Business Communications System	7)	Public ISDN
3)	DEFINITY Business Communications System	8)	Public and Private Networks
4)	Basic Rate Interface Telephone	9)	Central Office Switch
5)	Passive Bus	10)	Tandem Switch

Figure 5-1. DEFINITY Business Communications System and ISDN

The DEFINITY Business Communications 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 Primary Rate Interface 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 Primary Rate Interface D channel to supply signaling for B channels (voice and data) located on Primary Rate Interface 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

■ Sourcing

■ Consumer-to-business and business-to-business data retrieval

■ Logging for callback.

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 play with the unrestricted five-digit uniform dial plan, which allows up to five digits to be parsed for call routing.

- Extension Number Portability — The ability to keep extension numbers, and even Electronic Tandem Network and Direct Inward Dialed numbers, when employees move within the network, eliminating missed calls and saving valuable time.

- Traveling Class Marks — A mechanism for passing a caller's facility restriction level from one Extended Tandem Network system to another. Traveling Class Marks allow privilege checking to be passed across systems the through Electronic Tandem Network.

- 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.

Multimedia Messaging Solutions

6

With less than 30 percent of person-to-person business calls reaching the intended party on the first attempt, day-to-day business can be frustrating. Integration with Lucent Technologies' multimedia-messaging products can help ensure that important calls are not lost.

For nearly a decade, Lucent Technologies' voice messaging and voice response systems have provided businesses with the voice processing tools to communicate more efficiently and 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, multimedia messaging, which is much more than just an answering machine, 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 two-way calls run much longer and are devoted to business only 50 percent of the time.

DEFINITY AUDIX Messaging System

While many voice messaging systems require separate equipment and connections, the DEFINITY AUDIX system easily installs directly into a DEFINITY Business Communications System cabinet to support advanced multimedia 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 40 hours of recorded messages using a maximum of 16 ports (in two-port increments). The standard offer provides either 4 ports with 5 hours recording time or 4 ports with 10 hours of recording time. For an additional charge, you can increase the capacities to the maximums noted above. 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 and proprietary digital networking software, which allows it to exchange voice messages, subscriber profiles, and message status information with other voice messaging systems.

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

- Because it is integrated within the system, 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.

- Bypassing analog ports and digital conversions provides a more efficient, higher quality call storage process.

- Using the same terminal with look-alike screens to administer both the system and the DEFINITY AUDIX allows faster training and better performance.

- You can use the system's maintenance strategy with DEFINITY AUDIX to allow remote maintenance by the same team that maintains the system.

The entire system is contained on circuit cards, occupying five consecutive slots in a carrier on the system. All the major components are economically mounted onto the multifunction board using the latest technology in large scale integration circuit chips and in surface mount fabrication. The components mounted on the board include the central processing unit, the small computer system interface unit that supports the tape drive and hard disk, the digital signal processor complexes that do speech processing, and the time slot interfaces for the system.

In addition, an alarm board monitors the system power and environmental conditions, holds the disk drive, and includes a built-in modem for remote maintenance. The DEFINITY AUDIX system operates by emulating a digital port board.

Reliability and Security

In keeping with its commitment to assist clients in combating toll fraud, Lucent Technologies also designed the DEFINITY AUDIX system with security precautions against fraudulent access attempts. For example, the system allows a transfer of calls to only other voice mail subscribers. When a caller reaches the system and requests a transfer out, the system first checks the requested extension number against its subscriber database. If the extension number is not in the database, the transfer attempt is denied. Additionally, mailbox passwords can contain as many as 15 digits.

The alarm board on the DEFINITY AUDIX system has its own processor that allows maintenance and diagnostic access if the main processor fails. A liquid crystal display on the unit 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 built-in modem. A special alarm-originating feature helps speed problem diagnosing and correction.

Easy Installation and Expansion

Lucent Technologies specifically designed the DEFINITY AUDIX system for easy installation. There are no special power or cabling adjustments required. In fact, the system is as easy to install as a circuit pack.

System expansion is simple. All the hardware required for the full 16 ports is included in the initial DEFINITY AUDIX system. If you decide to buy fewer than 16 ports initially, you can order additional ports at any time. Lucent Technologies can then activate those additional ports through a simple software change that technicians download remotely. There is no need to modify the basic hardware, and the system still occupies only five slots.

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 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. 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 Business Communications System. Prompts are available in several languages.

The DEFINITY AUDIX system is the best voice messaging solution for international companies because it is integrated in the DEFINITY Business Communications System. Because it does not constitute a separate piece of equipment, review and approval by government agencies are often not required.

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. And the AUDIX system helps to protect sensitive information by requiring users to enter a combination of subscriber login codes and passwords to access 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, Outcalling, 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 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.

- *Priority Messaging* places important messages ahead of others.

- *Outcalling* automatically dials a prearranged phone number or pager when messages are received in a user'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 users on the system.

- *System Broadcast* capabilities are available in two forms: Broadcast Voice Mail and Login Announcement.

- *AUDIX Directory* allows users to 'look-up' the extension number of any other user by simply entering their 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 AUDIX system mailboxes for users 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 room in a subscriber's mailbox.

- *Name Record by Subscriber* lets users 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 AUDIX Voice Messaging.

- *Group Lists* allows users to create mailing lists of up to 250 people to use for broadcasting messages.

- *Message Forwarding* lets users forward messages they have received, with or without attached comments.

- *Name Addressing* allows users to enter the name of the caller if the extension number is not known.

- *Private Messaging* is a special coding feature that prevents recipients from forwarding messages.

- *Leave Word Calling* allows users to simply press a button on their telephones in order to leave a standard *call me* message on any extension.

- *On-Line Help* provides users with instant access to voiced instructions at any point in the process.

Multimedia Video Solutions

7

Your colleagues have asked to meet with you 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.

Standard business procedure would dictate that you jump on the next airplane — briefcase full of needed files and the prototype packed carefully for shipment. Hopefully, the prototype will arrive intact. And hopefully, the time away from the office will not set you too far back in your already hectic schedule.

But, now the DEFINITY Business Communications System provides an alternative to business as usual — revolutionary visual 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 visual communications. So, instead of heading to the airport, you take the prototype to the video conferencing room, feeling glad this face-to-face meeting does not involve luggage or jet-lag. And once your meeting is done, you are back at your office in a matter of minutes, ready to start implementing the decisions just made.

Besides the obvious advantage of reduced travel expenses, 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 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 the other.

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.

For additional information about Lucent Technologies' video conferencing solutions, contact the Lucent Technologies Group Video center at +1-303-488-6454 (outside of North America) or 1-888-345-5400 (in North America).

⇒ **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.

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 line of Group Video Systems, based on the PictureTel Group Video Line (System 1000, 4000, Venue 2000 and Concorde 4500), is 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 at speeds from 112 kbps up to 384 kbps. When connected to your system via either Data Communication Protocol ports or a DS1 interface, video calls are placed as easily as voice calls. And you can benefit by using your system's 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 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. Advanced video-compression technology ensures excellent picture quality across a variety of available speeds (112 to 768 kbps). 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 boardroom. All models have colour monitors, with dual monitors available on most for simultaneous viewing of video and high-resolution still images. The systems available are as follows:

- The System 1000 — Quality, inexpensive group video products:

    — Model 30 — For small group or individual meetings.

    — Model 50 — For medium to large group meetings.

- The Venue 2000 — A relatively inexpensive system with an enhanced user interface.

- The System 4000EX — The industry's most popular high performance systems:

    — Model 200 — The office system: modular, portable, and well-suited for small offices and conference rooms.

    — Model 200 — The cart system: a larger, portable system on wheels.

- The Concorde 4500 — A highly advanced system offering superior transmission quality and unparalleled ease-of-use.

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 standard and PAL, the European 625-line standard, providing global compatibility for your visual communication needs.

Desktop Conferencing Systems

Many meetings involve just two or three individuals who need to share information — whether it be text, data, graphics, or some other form of information. And with larger teams of people, these individuals may be separated in locations around the country or the world. They have the same need to meet strategic objectives, shorten project cycle time, and improve working relationships with project partners. These needs can also be satisfied through visual communications.

To satisfy these requirements, integrated telephone and PC visual communications systems are available. These desktop conferencing systems use ISDN-BRI or two switched-56 line network facilities.

Monitors and cameras create a unique desktop environment that provides you and your business with the following benefits:

- Software application screens and data can be called up, shared, changed, and annotated by video conferees while they hold a video conference.

- Geographically separated people are brought together along with the data and information they need to share at the desktop.

- Major obstacles to creative collaboration and situations that are information-intensive are overcome.

- Desktop workers can communicate with others who are using desktop conferencing or the Group Video System.

- Desktop workers can communicate with other systems supporting the ITU-T H.320 standard.

- Your system is converted into a solution for all your communication requirements — voice, data, and video.

MultiPoint Control Unit

When connecting more than two video endpoints, you can use the MultiPoint Control 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 Control Unit can operate behind any DEFINITY Business Communications System or can be directly connected to the network. The MultiPoint Control Unit can support from 4 to 64 ports in four-port increments. Those ports can then be used to connect multiple video endpoints, either Group Video System or Desktop Conferencing Systems, in a multipoint conference. Group Video System can be linked at speeds from 56 kbps to full T-1, while Desktop Conferencing Systems can be linked at speeds from 56 kbps to 384 kbps.

The MultiPoint Control Unit uses the ITU-T H.320 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 standard ensures a common level of visual, graphics, and audio quality that will satisfy your visual communication requirements.

The MultiPoint Control Unit is built on the architecture of the DEFINITY Business Communications System. The MultiPoint Control Unit sits in its own carrier and takes up approximately the same space as a single carrier cabinet. Designed for growth, the MultiPoint Control Unit's architecture allows you to add additional circuit packs and carriers as needed.

Arranging Conferences

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

You can assign a number to each conference participant and set up the MultiPoint Control 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 Control Unit can also be programmed to out-dial to the video endpoints at a designated time.

The MultiPoint Control 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.

System Management Solutions

8

The DEFINITY Business Communications System is a digital communications system 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 upon the tools that administer them.

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. Whether your system is small or large, straightforward or sophisticated, or somewhere in-between, the DEFINITY Business Communications System has the tool to effectively and efficiently manage that system. This section takes a look at the capabilities for system management.

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

We think this system management view of the DEFINITY Business Communications System 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 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 DEFINITY Business Communications System supports the following system management features:

- Local access via TERRANOVA

- Multiple concurrent administration/maintenance sessions

- Terminal administration, using administration without hardware and terminal translation initialization

- Performance measurements

- Trouble Tracker

- Call Detail Recording

- Other miscellaneous capabilities.

TERRANOVA

The DEFINITY Business Communications System uses the TERRANOVA terminal emulator installed on a PC for the standard system management tool. This gives you access to all possible administration, maintenance, traffic, and performance information on the system. The TERRANOVA software is shipped with each system.

An optional product is the WINPBX Information Management System that is used with TERRANOVA to schedule administration changes, track changes for multiple systems, create custom reports, and manage multiple locations more efficiently.

Concurrent User Sessions

In order 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. The three 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.

Terminal Administration

The system includes a number of 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. This allows a phantom extension to be administered on the system for each call type that needs to be identified.

Administration Without Hardware also supports the ability to store station templates (models). These 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 terminal types:

- Analog telephones
- Digital Communications Protocol telephones
- Hybrid telephones.

Larger configurations support additional terminal types to those listed above. These include the following:

- Attendant consoles
- Voice/computers (such as Digital Communications Protocol terminals with voice and data capabilities)
- Data modules
- ISDN Basic Rate Interface telephones and computers
- Analog queue warning ports
- Announcement circuit packs.

Terminal Translation Initialization

The system provides terminal translation initialization, 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, reducing costs for station moves. Individual lines need only be wired to the correct type of port, rather than a specific port.

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

Performance Measurements

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:

- 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 fields are as follows:

- Group Attributes report 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.

- Summary reports 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 number of calls offered, abandoned while at that coverage point, and overflowing 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.

- Processor Occupancy report provides summary information on how heavily the processor is loaded. It includes fields giving 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:

- Traffic Summary report provides a performance summary of the system with the following information:

  — 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:

1. Major alarm

2. The list of trunk groups to be studied with the detailed report were last changed

3. The list of coverage groups to be studied were last changed

4. The list of Automatic Alternate Routing/Automatic Route Selection routing patterns to be studied was last changed

These can be used to verify that your system and its users are not experiencing performance degradation due to overloaded system resources.

- 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, all remote access 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 board

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

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

- Hunt Group 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 between the options of 15 minutes, 30 minutes, or one hour. The report is divided into two sections:

    — Group Identification includes the trunk group number, size, 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 following measurements and reports are needed for engineering and load balancing a large system. These measurements include 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.

Trouble Tracker

Trouble Tracker is a Unified Network Management Architecture-compatible network management system that allows you to monitor the health of your network.

Trouble Tracker monitors alarms and trouble tickets to support a variety of Lucent Technologies and non-Lucent Technologies premises products. Trouble Tracker receives and logs alarms from a variety of different sources, creates and stores trouble tickets, forwards alarms to a variety of other locations, and provides management reports for planning and analysis.

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 DEFINITY Business Communications System has enhanced the Call Detail Recording capabilities available to you.

Call Detail Recording Features

The Call Detail Recording feature has the following new capabilities:

- Distinguish voice from data on trunk calls

- Choose whether to record the Direct Access Calling number in the "Dialed Number" field of the Call Detail Recording record, or record either the split or the agent 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 endpoints) 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

- Provide Call Detail Recording call splitting, which allows incoming and outgoing calls to be split into separate call records in order to track calls that transferred to other internal parties.

As you can see, the DEFINITY Business Communications 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/S) and any customer-provided storage device with an RS-232C interface

- Processing devices (such as the Lucent Technologies Call Accounting System Plus version 3, Cost Allocator, or host processors) that are supported over an RS-232C interface with XON/XOFF flow control

- Asynchronous ASCII printers with RS-232C interface.

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

Call Accounting System Plus Version 3

Lucent Technologies' Call Accounting System Plus Version 3 is a comprehensive telecommunications cost-management system for your small- to medium-sized business locations. Call Accounting System Plus Version 3 stores call records from the system, assigns costs to the calls, and generates standard summary, detail, account code, selection, and preselected reports.

Call Accounting System Plus Version 3 collects information about telephone calls and computes the costs using actual telephone company rates. The system provides 11 major MTS metro rate tables. If your location has special local or long distance services, Call Accounting System Plus Version 3 can also provide customized site-specific rate tables to achieve a higher accuracy level on call-pricing.

- Call Accounting System Plus Version 3 software, which runs on MS-DOS[*]-based work group system computers, offers several new features:

- The ability to record and process Automatic Number Identification information

- Enhanced reporting

- Accommodates multiple private voice networks, non-traditional dialing patterns, and other call details that require additional call processing

- Streamlined administration

- Comprehensive user assistance

Call Accounting System Plus Version 3 can be used to allocate costs, improve client invoicing, control system misuse, enhance employee management, and verify invoices.

Hacker Tracker

Hacker Tracker software automatically detects abnormal calling activities and provides real-time alarms and alerts to a your telecommunications staff.

Operating as an optional module that connects to Lucent Technologies' Call Accounting System Plus Version 3, it uses information gathered by the call accounting system to continually monitor all incoming and outgoing telephone calls and alerts security or administrators when it detects calling activity that may signal toll fraud.

Call accounting systems typically track calls and generate reports so that companies can bill long distance service back to their internal organizations. Hacker Tracker uses that information to notify your company of unusual, potentially fraudulent calling patterns.

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[*]   Registered trademark of Microsoft Corporation.

Administrators can define up to ten different conditions that will trigger an alarm. For example, an administrator may instruct the system to track certain calling patterns based on time-of-day or calls made to certain parts of the world. If the software detects a threshold violation, it automatically generates an alarm so the administrator can take immediate action to stop the fraud.

You can choose from a variety of alarm notifications. One option activates a computer's speaker and generates a "beeping" alarm tone when a security violation has occurred. Another can call someone at home by activating a digital pager.

Cost Allocator

Cost Allocator is Lucent Technologies' response to the need for a high-powered, UNIX[*] system-based, system call-accounting processing product. Targeted at the needs of the medium- to large-sized call-volume client, Cost Allocator receives, rates, and produces reports about incoming and outgoing call-detail records from multiple systems at multiple sites. Its capabilities complement the power, efficiencies, and value of the DEFINITY Business Communications System.

Cost Allocator application software, running on a 6386 WorkGroup System computer, can process information from up to 255 sites. An optional built-in poller package allows Cost Allocator to collect information from up to 90 sites, polling 7 simultaneously. When polling volumes or needs for increased flexibility warrant, the software-based Call Detail Record Poller can be used in place of the basic poller option. Call Detail Recording Poller's expanded capacities allow collection from over 250 sites, with 28 polled simultaneously.

The system produces reports in a wide variety of standard formats, and custom reports are easily tailored to provide specialized information. Reports can be displayed on a terminal, sent to a printer, and/or saved on disk. Basic types of reports, available in detailed or summary levels, include the following:

- Organizational and Account Code reports list individual station-user or authorization-code costs.

- Analysis reports offer control over types of calls and call handling.

- Selection reports help investigate system misuse by isolating calls that match customer-defined criteria.

A Custom Report Writer option allows you to access the Cost Allocator database. Once it is accessed, you can define any type of report desired.

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[*]   Registered trademark of Novell, Inc. in the United States and Other Countries, Licensed Exclusively through X/Open Company Ltd.

Call Detail Recording Unit/SE

The Call Detail Recording Unit/SE (Call Detail Recording Unit/SE) can be used when call detail record storage is more than 7,000 records.

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 DEFINITY Business Communications System.

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.

Following are some of the Call Detail Recording Unit/SE highlights:

- Stores 77,000 Call Detail Recordings for a 24-word record
- Stores 127,000 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 3,600 Call Detail Recordings per hour
- Can be remotely administered
- Provides 72 hours of Call Detail Recording 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- 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

Several other management features enhance your investment in the DEFINITY Business Communications System.

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.

Reporting Capabilities

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

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 entered. The system also supplies:

— 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 amount 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
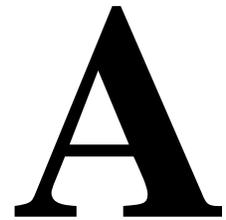
This appendix provides an alphabetically arranged description of each feature of the DEFINITY Business Communications System. Please see Appendix B, "System Capacity Limits" for feature capacities. In addition, not all system applications or adjunct applications may be available in your country. This appendix discusses all DEFINITY Business Communications System capabilities available internationally. Please check with your Lucent Technologies Account Team for further information and details about the specifics for your company and country.

All features on the DEFINITY Business Communications System are available in the standard package, except for the following, which must be ordered and initialized by Lucent Technologies personnel:

- Integrated Services Digital Network — Primary Rate Interface
- Integrated Services Digital Network — Primary Rate Interface using the Packet Controller service
- Terminal Translation Initialization.

Each feature is described briefly. Most features have many complex capabilities and options. The DEFINITY documentation set describes each feature in detail and provides complete implementation and administration information. Some features are systems of their own and have their own documentation, such as Call Detail Recording and AUDIX. See your local distributor for more information on each of these features.

In addition to the brief descriptions for each feature, this appendix contains detailed information about the features that are unique to the system that are not described in any other reference documents. These features include the following:

- Attendant Backup
- Attendant Crisis Alert
- Controlled Toll Restriction
- Direct Access Calling (similar to Call Vectoring)

Abbreviated Dialing

Provides lists of stored phone numbers that can used by telephone users and data terminal users. The user dials the list number and the one-, two-, or three-digit number associated with the phone number wanted. The number is then automatically dialed by the system. A frequently called number can be stored on an abbreviated dialing button and the user simply presses the button to make the call.

Additional TN748C tone detectors (up to five per system) may be required if the special "wait" character is used frequently. Other tone detector circuit packs may be more appropriate.

Administered Connections

Automatically establishes an end-to-end connection between two access or data endpoints. This feature provides the following enhanced capabilities: Support of both permanent and scheduled connections, Auto Restoration for connections routed over Software Defined Data Network trunks, an administrable retry interval for each administered connection, administrable alarm strategy per connection, establishment/retry/auto restoration order based on administered priority. Up to 128 administered connections can be established.

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

- Access Endpoint Circuit Packs — TN464F Digital Services 1 Interface and TN760D Analog Tie Trunk.

- Data Endpoint Circuit Packs — TN726B Data LIne or TN754B Digital LIne (supports 4-wire DCP equipment).

- Data Modules — 700A/700D Processor Data Module or Modular Processor Data Module, 700B/700C/700E Time Division Multiplexer or Modular Processor Data Module, 7400D-series telephones with Digital Terminal Data Module, Personal Computer/PBX, 510D, 515 Business Communications Terminal.

- Trunk Circuit Packs — TN464F Digital Services 1 Interface and TN760D Analog Tie Trunk.

No additional software is required.

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, 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. This feature is not available on 32-character displays. These displays default to English.

No additional software or hardware is required.

Administration Without Hardware

Allows the administration of a phone which is not physically present on the system. This feature works the same as administration with hardware. When terminals 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.

No additional hardware or software is required.

Alphanumeric Dialing

Allows data terminal users to make a call by entering a short alphanumeric name. The system then searches through an administered table and dials the longer phone number associated with that name in the table.

No additional hardware or software is required.

Alternate Facility Restriction Levels

Allows an authorized attendant on the system to quickly 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 assigned also. At a certain time, the attendant can change to the alternates, this changing the access to lines and trunks. For example, a company might want use this feature to disable most long-distance calling at night so that unauthorized staff within the building at night cannot make long distance calls.

No additional hardware is required for this feature.

Answer Detection

Provides more accurate call detail records where tone detection is possible and Network Answer Supervision is not received. This feature does not accurately detect all types of tones, especially in countries whose tone schemes are not similar to the U.S.

Requires a TN744 Call Classifier circuit pack.

Attendant Auto-Manual Splitting

Allows the attendant to announce a call or talk privately with the called party without being heard by the other party on the call. This feature provides for splitting the calling party away so the attendant can confidentially determine if the called party wants to accept the call.

No additional hardware or software is required.

Attendant Backup

Allows users to access most attendant console features from one or more specially-administered multiappearance voice terminals. Using this backup mode, users can answer calls faster, thus providing better service to your clients and employees.

Attendant Call Waiting

Allows the 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 burst tone is heard only by the called telephone user.

No additional hardware or software is required.

Attendant Control of Trunk Group Access

Allows the 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.

No additional hardware or software is required.

Attendant Crisis Alert

Provides a visual, audible, and printed record when users place a call to the local emergency service agency. This gives users the ability to assist emergency personnel when they arrive at the location by identifying where the call came from and when the call was made. This feature uses the Automatic Route Selection (ARS) feature to allow routing of any emergency service access code (such as 911) to the appropriate emergency service agency, while also identifying the call for crisis alerting.

Attendant Direct Extension Selection with Busy Lamp Field

Allows the attendant to keep track of extension status — whether the extension is idle or busy — 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. Without having to dial the extension number, the attendant can place calls to 800 extensions using the basic selector console, and 2,000 extensions using the enhanced selector console; or the attendant can place calls to 99899 extensions using the Group Select feature button.

Requires an optional selector console. No additional software is required.

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.

No additional hardware or software is required.

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 my 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 display.

No additional hardware or software is required.

Attendant Intrusion (Call Offer)

Allows an attendant to enter an existing call to inform the person being called about a message or another call. Upon intrusion, tone may be applied if administered.

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

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.

No special hardware requirements exist for this feature.

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.

There is no special hardware required for this feature.

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. Multiappearance users press the Conference or Transfer button to recall the attendant and remain on the connection when either button is used.

No additional hardware or software is required.

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.

No additional hardware or software is required.

Attendant Serial Calling

Allows the attendant to transfer trunk calls that return to the same attendant after the called party hangs up. The returned call may then be transferred to another telephone within the system and this can continue to recur. This feature is particularly useful if trunks are scarce and Direct Inward Dialing services unavailable.

No special hardware is required for this feature.

Audible Message Waiting

Places a stutter at the beginning of the dial tone when a telephone user picks up the phone. This stutter dial tone indicates that the user has a message waiting. This feature is typically 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.

No additional hardware or software is required.

Audio Information Exchange Interface

Provides an extensive message-handling system for recording and distributing spoken messages or voice messaging. AUDIX prompts users to guide them through the process of creating, sending, retrieving, answering, saving, and forwarding spoken messages. AUDIX is resident on the system. AUDIX systems can be networked through systems or other AUDIX machines. AUDIX Voice Power®, AUDIX Voice Power Lodging, and AUDIX Voice Power Auto Attendant are examples of systems that can be networked though a system using a data link. Prompt packages are available in a number of languages other than English and prompts can be further customized by your company.

For DEFINITY AUDIX, you must have five contiguous slots in the system carrier. DEFINITY AUDIX supports up to 16 ports (in two-port increments) and a total recording time of 40 hours.

Authorization Codes

Provides the ability to extend users' calling privileges. By dialing authorization codes, users can make calls that are otherwise not allowed from the phone they are using, identify certain calls for cost-accounting purposes, and use the authorization code for unique calling identification. Authorization codes can be from four to seven digits long.

Auto Start and Don't Split

Auto Start 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.

Automatic Alternate Routing

Provides alternative routing choices for private on-network calls. With this feature, the system automatically selects the preferred route (usually the least expensive) for private network calls. Automatic Alternate Routing also provides digit modification to allow on-network calls to route through the public network when an on-network route is not available. The private network that uses this feature is called an Extended Trunk Network.

Automatic Alternate Routing and Automatic Route Selection Partitioning

Provides for Automatic Alternate Routing and Automatic Route Selection to be partitioned among up to eight different groups of users within a single system. This feature allows different call routing for each group of users.

Automatic Callback

Allows internal users who placed a call to a busy or unanswered internal telephone to be called back automatically when the called telephone becomes available. A busy voice terminal becomes available when the suer hangs up after completing the current call. An unanswered voice terminal becomes available after it is used for another call and is then hung up. When a telephone becomes available, the system alerts the caller with a three-burst ring. The caller than lifts the handset and the ringing at the called telephone begins immediately.

Automatic Circuit Assurance

Assists an assigned user to identify 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 the identity of an incoming caller when a user is already on a call. The displays are administrable and can be in English, French, Italian, or Spanish. Other languages may be available; see your local distributor.

Automatic Route Selection

Provides access to public and private communications networks so that long-distance calls can be routed over the best available and/or most economical routes at the time of the day that the call is placed. Provides outgoing access to local Central Office, Foreign Exchange, Wide Area Telecommunications Service, and other long distance carriers. Provides transparent access to any Interexchange Carrier. Provides partitioning of Automatic Route Selection routing patterns.

Automatic Transmission Measurement System

Provides for voice and data trunk facilities to be measured for satisfactory transmission performance. Acceptable performance, the scheduling of tests, and report contents are administrable.

Bridged Call Appearance — MultiAppearance Voice Terminal

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.

Bridged Call Appearance — Single-Line Voice Terminal

Allows single-line telephones users to have a bridged appearance on a multiappearance phone.

Busy Verification of Terminals and Trunks

Allows attendants and users of multiappearance 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 voice terminal that is truly busy and one that only appears busy because of some trouble condition. These users can also use the feature to quickly identify faulty trunks.

Call Coverage

Provides for the automatic redirection of certain calls to a coverage path that might include such things as another telephone or group of telephones, AUDIX, hunt groups, or an attendant group. This feature is extremely versatile and permits suitable alternate answering arrangements for virtually every level of employee. 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, AUDIX, or a coverage answer group (a group of up to eight voice terminals specifically established to answer redirected calls). All group members are rung simultaneously. Any group member can answer the call.

Call Detail Recording Account Code Dialing
Forced Entry of Account Codes

Allows certain calls to be associated with a particular project or account number by forcing users to dial specified account codes before making outgoing calls. This facilitates call accounting.

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 All Calls

Allows users to forward all calls to their extension number to another extension, an off-premises number, the attendant group, or a specific attendant. This feature is activated by dialing a dial access code or by pressing a Call Forwarding button and then dialing the designated (forwarded-to) number. When Call Forwarding is activated at a telephone and a call for that telephone is forwarded, the telephone can (if administered to do so) receive a redirection notification signal that a call is being redirected.

Call Park

Allows users to put a call on hold and then retrieve the call from any other telephone within the system. This is helpful when a user is active on a call and needs to go to another location for information. Call Park also allows users to answer a call from any telephone after being paged by a telephone user or an attendant.

Call Pickup

Allows users to answer calls to other extensions within the user's specified Call Pickup group. When one member of a Call Pickup group is away and receives an incoming call, any member of the Call Pickup group can answer the call. With this feature, a user does not have to leave his or her telephone to answer a call for a nearby telephone. Instead, the user simply lifts the handset and dials an access code or presses a Call Pickup button.

Call Prompting

Allows the system to collect information from the calling party and direct the calls via Direct Access Calling. 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 handle the call in some other way (taking a message, for example).

Call Waiting Termination

Allows for 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.

Call-By-Call Service Selection

Enables a single trunk group to be used by a variety of services rather than requiring each trunk group to be dedicated to a specific service. This feature applies to integrated services digital network-private rate interface trunk groups.

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 system. For example, a calling party class of restriction could be used to prevent callers from accessing the public network.

Class of Service

Defines whether or not telephone users can access the following features and functions: Automatic Callback, Call Forwarding All Calls, Data Privacy, Priority Calling, Off-Hook Alert, or Console Permission.

Code Calling Access

Allows attendants, users, and tie trunk users to page with 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 the 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 — Terminal

Allows multiappearance 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.

Constellation Voice/Data Terminal Support

Provides support for integrating telephones with the Constellation Voice/Data Terminal. The Constellation is a voice/data workstation that provides an integrated environment for applications using digital communications protocol-based signaling and provides enhanced capability in a telemarketing environment.

Consult

Allows a covering user, after answering a call that has come to them through Call Coverage, to call the called party for private consultation. Consult can be used to let a covering user ask the principal if they want to speak with the calling party.

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 multiappearance telephones users without a display in a Coverage Answer Group to identify an incoming call to that group.

Customer-Provided Equipment Alarm

Provides you with an indication that an 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 the following: 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 digital call protocol and integrated services digital network basic rate interface telephones.

Data Hot Line

Provides for automatic placement of a data call when the originator goes off hook (usually from a data module). 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 go off hook.

Data Privacy

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data Privacy is activated when the user dials the 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. This feature is administered at the system level to selected analog and multiappearance telephones and trunk groups.

Data-Only Off-Premises Extensions

Allows users to establish data calls involving equipment that is located remotely from the system site using DATAPHONE digital service or other private line data facilities.

Default Dialing

Provides data terminal users who dial a specific number the majority of the time 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.

Dial Access to Attendant

Allows telephone users to access an attendant by dialing an attendant access code. Attendants can then extend the call to a trunk or to another telephone.

Dial Plan

The Dial Plan is the system's guide to digit translation. The Dial Plan, or first-digit and second-digit table, established during administration for each system, provides information to the system on what to do with dialed digits.

Digital Multiplexed Interface

Supports two signaling techniques, Bit-Oriented signaling and Message-Oriented signaling, for direct connection to host computers.

Digital Services 1 Trunk Service

Provides a 24-channel, 1.544 Mbps rate digital interface for a number of different types of trunks.

Direct Access Calling

Provides processing of incoming and internal calls according to a programmed set of commands. Procedure commands may direct calls to on-premise or off-premise destinations, to any hunt group or split, or to a specific call treatment such as an announcement, forced disconnect, forced busy, or delay treatment. For example, the system can collect digits from the user via Call Prompting and route calls to a destination specified by those digits. This is similar to the Call Vectoring feature on DEFINITY.

Direct Department Calling and Uniform Call Distribution

Enables direct inward access to an answering group other than the attendant even if the system does not have the direct inward dial feature. This feature provides a method of distributing calls to a group of telephones or individual attendants, data modules, data line circuit ports, or modems. Direct Department Calling routes incoming calls in a direct administered sequence. Uniform Call Distribution rings the most idle member of a group, distributing the calls evenly among the group members.

Direct Inward Dialing

Connects calls from the public network directly to the dialed extension number without attendant assistance.

Direct Inward and Outward Dialing — International

Provides a two-way service with both inward and outward dialing features. This feature allows calls from an international public exchange to be made directly to the system. This feature is a combination of the Direct Inward Dialing feature and the Direct Outward Dialing feature via a common analog or digital trunk and implies a two-way service.

Direct Outward Dialing

Allows users to access the public network without attendant assistance.

Distinctive Ringing

Helps users and attendants distinguish between various types of incoming calls by distinctive ringing patterns that can be administered for internal, external, and priority calls. In addition, there are ringing signals for Manual and Dial Intercom, Manual signaling, and Redirection Notification.

E1 Trunk Service

Provides a 32-channel, 2.048 Mbps rate digital for a number of different types of trunks.

Electronics Industry Association Interface

Provides an alternative for host connections and analog telephone users who use simple data terminals or personal computers which emulate simple data terminals.

Emergency Access to the Attendant

Provides a process to place emergency calls to an attendant. These calls can be placed automatically by the system or can be dialed by users. Emergency access calls can receive priority handling by the attendant.

End-to-End Signaling

Enables both dual tone multifrequency (touch tone) telephones and non-dual tone multifrequency telephones to send digits over a trunk after the connection has been made. For example, this feature allows the user of a rotary phone or other non-dual tone multifrequency phone to access equipment, such as AUDIX, that is controlled by dual tone multifrequency signaling. In this case, addressing digits are sent over the trunk as rotary. After addressing is complete and the call is connected, any additional digits are sent as dual tone multifrequency signals.

Enhanced Abbreviated Dialing

Supplements the standard Abbreviated Dial feature by enabling the use of an Enhanced Number List containing up to 1,000 entries. This list is in addition to the System List.

Extended Trunk Access

Provides a mechanism for routing calls that are not defined either in the first- or second-digit tables or the feature/trunk access code tables. This feature makes use of an Extended Trunk Access routing pattern and/or an Extended Trunk Access node number for determining how to route an unidentified call. The use of this feature is transparent to end users.

Extension Number Portability

Provides the ability to assign any extension to any system in an Extension Number Portability subnetwork. Within certain limits, this enables an employee who changes location to keep his or her phone number.

Facility Busy Indication

Allows users of multiappearance 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. When the lamp is not lit, the resource is idle and available.

Facility Restriction Levels and Traveling Class Marks

Provide a method of allowing certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed to use public trunks to other corporate locations, while other users may be restricted to the less expensive private network lines. This feature provides up to eight levels of restriction for users of the Automatic Alternate Routing and/or Automatic Route Selection features.

Facility Test Calls (with Security Measures)

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. There are security measures to prevent unauthorized use.

Facility and Non-Facility Associated Signaling

Provides signaling for integrated services digital network-private rate interface. Facility Associated signaling allows an integrated services digital network-private rate interface digital services 1/European 1 interface D-channel to carry signaling information for only those B-channels located on the same facility (circuit pack) as the D-channel. Non-Facility Associated signaling, on the other hand, allows a D-channel to convey signaling information for B-channels (voice and data channels) on facilities other than the one containing the D-channel. As a result, a D-channel can carry signaling information for numerous B-channels located on different digital services 1/European 1 facilities.

Forced Entry of Account Codes

This is software that forces users to enter account codes. By requiring account codes to be dialed on specific outgoing calls, this feature software provides an easy method of allocating the costs of specific calls to the correct project, department, etc. Call information is recorded by the Station Message Detail Recording feature for this purpose. An account code length can be up to 15 digits.

Generalized Route Selection

Provides voice and data call routing capabilities to select not only the least cost routing, but also optimal routing over the appropriate facilities. This feature is a capability that enhances the current Automatic Alternate Routing and Automatic Route Selection features by providing additional parameters in the routing decision.

Go to Cover

Allows users, when making a call to another internal extension, to send the call directly to the called party's coverage.

Hold

Allows telephone users 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 multifunction 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, the depression of the second call appearance would drop the first call.

Hot Line Service

Allows a single-line telephone user, by simply lifting the telephone receiver, to automatically place a call to a preassigned extension number, public or private network telephone number, or feature access code. This feature is helpful in any application where very fast service is required.

Individual Attendant Access

Allows users to call a specific attendant console. Each attendant console can be assigned an individual extension number.

Information System Network Interface

Provides an interface between the system and Lucent Technologies Information System Network via a Data Line Port in conjunction with an Asynchronous Data Unit. The Information Service Network is a packet-switched local area network that links mainframe computers, minicomputers, word processors, storage devices, personal computers, printers, and communications processors into a single system.

Integrated Announcements

Digitizes and stores recorded announcements (messages) internally within the system. Announcements can be recorded and updated from any telephone. Announcement configuration is performed through standard system management. Even a power failure will not affect the integrity of the 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. The announcements can be used with an almost endless number of applications. Announcements can be made in any language, and the modular announcement circuit packs make the system easily expandable.

Integrated 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.

Integrated Services Digital Network — Basic Rate Interface

Enables connection of the system to equipment or endpoints that supports an Integrated Services Digital Network by using a standard format called the Basic Rate Interface. This feature is a 192-kbps interface that carries two 64-kbps B-channels and one 16-kbps D-channel. Each B-channel supports voice and data, while the D-channel transports data, signaling, and other bits for framing. This feature is only available on systems that support μ law companding.

Integrated Services Digital Network — Primary Rate Interface

Enables connection of the system to an Integrated Services Digital Network by using a standard frame format called the Primary Rate Interface. The standard for this interface consists of layers 1, 2, and 3 of the Open System Interconnect model. In this interface the transmission standard for layer 1 is either Digital Services 1 (T1) or European Conference of Postal and Telecommunications Rate 1 (E1). Digital PT1 (E1) is a standard that carries traffic at a rate of 2.048 Mbps and is used in Europe and elsewhere.

The system offers several administrable protocols, each of which provides a different set of services that Integrated Services Digital Network-Private Rate Interface allows. For the U.S., users can administer either the AT&T Switched Network or National Integrated Services Digital Network-2. For other countries, the system provides Global Integrated Services Digital Network-Private Rate Interface, which includes any combination of services, including Basic Call, DDI Display, and Q-SIG Global Networking.

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 the 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 multiappearance 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 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. With this feature, an Internal Automatic Answer feature button is assigned to the user's telephone. When the feature button is pressed by the user, the feature is activated and calls will automatically be 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 users 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 users press the Last Number Dialed button or dials the Last Number dialed feature access code, the call is placed 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. This lamp is called an Automatic Message Waiting lamp since the status of the lamp is automatically controlled by the system. Leave Word Calling messages can be retrieved using a telephone display, Voice Message Retrieval, or AUDIX. Voice Message Retrieval is available in U.S. English, British English, or Italian.

List Call Forwarding

Lists all call-forwarded stations and the respective forwarding numbers, allowing the administrator to monitor outgoing calling patterns. This helps detect and prevent toll fraud.

Line Lockout

Removes single-line telephone extension numbers from service when users fail to hang up after receiving dial tone for 10 seconds and then intercept tone for 30 seconds. These intervals are administrable. The out-of-service condition lasts until the telephone user hangs up the phone.

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 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 can speak into the handset and make the announcement.

Loudspeaker Paging Access — Deluxe

Provides attendants and telephones users with integrated dial access to voice paging equipment and Call Park capabilities. When this feature is activated, the call is automatically parked. The Call Park feature does not have to be activated separately.

MERLIN/System 25 Telephone Support — 731xH Series

Enables certain MERLIN/System 25 telephones to be reused with the DEFINITY Business Communications System. The terminals that can be reused are the 7309H, 7313H, 7314H, 7315H (with built-in display), 7316H, and 7317H (with built-in display).

Malicious Call Trace

Provides a way for a called user to notify a predefined set of users that the called user may be receiving a malicious call. Any one of the predefined set of users can then retrieve certain information related to the call that will aid in tracking the source of the call. This feature also provides a method of generating an audio recording of the call. Airlines, utility companies, government offices and major corporations are a few examples of owners who may experience malicious calls. Access to a feature such as Malicious Call Trace can allow them to deter or respond appropriately to bomb threats, product tampering threats, or other possible malicious calls.

Manual Message Waiting

Allows multiappearance telephone users, by pressing a designated button on their own terminals, to light the status lamp associated with the manual Message Waiting button at another multiappearance telephone. This feature can be administered only to pairs of telephones such as 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.

Multiappearance Preselection and Preference

Provides multiappearance telephone users with options for placing or answering calls on selected appearances. Ringing Appearance Preference automatically connects the 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 the user wants to reenter a held call or activate a feature and can also be used with a feature button. For example, if an Abbreviated Dialing button is pressed, the call appearance is automatically selected and, if the user lifts the handset within 5 seconds, the call is automatically placed. The Preselection option overrides both preference options.

Multiple Audio/Music Sources

Allows you to create flexible menus of delay announcements. Up to 256 messages or music selections of varying lengths can be played based on the nature of the call, the length of the wait, or service objective. Helps decrease the number of abandoned calls and provides a flexible marketing venue.

Multiple Listed Directory Numbers

Allows a publicly published number for each incoming and two-way (incoming side) foreign exchange and local central office trunk group assigned to the system. Also allows a number of direct inward dial numbers to be treated as listed directory numbers. When a listed directory number is called, a trunk group is accessed. The trunk group then routes the call to the incoming destination designated for that trunk group.

Music-on-Hold Access

Provides music to a party that is on hold, waiting in a queue, parked, or on a trunk call that is being transferred. The music lets the waiting party know that the connection is still in effect. The system provides automatic access to the music source.

Network Access — Private

Enables calls to be connected to other switching systems and networks without having to use the public network. Calls can be connected to the following types of networks and switching systems: Common Control Switching Arrangement, Electronic Tandem Network, Enhanced Private Switched Communications Service, Tandem Tie Trunk Network, and (in Italy) Traslatore Giunzione Uscente/Entrante/Interno.

Network Access — Public

Enables calls to be connected to and from the public network. The Automatic Route Selection feature can be used to select the most-preferred route for outgoing calls to the public network. Alternatively, trunk access codes can be dialed for manual route selection. Long-distance carrier access codes can be dialed to select particular carriers.

Night Service — Hunt Group

Allows an attendant or a hunt group supervisor to individually assign a hunt group to the night service mode. All calls for the hunt group are then redirected to the hunt group's designated Night Service Extension.

Night Service — Night Console Service

Allows the attendant to direct all calls for the primary and daytime attendant to a night console. To activate and deactivate this feature, the attendant typically pushes the Night button on the principal attendant console. When this feature is activated, Night Service — Night Console Service calls to the attendant group are still handled by an attendant, even though the primary and daytime attendant consoles are out of service. Only one night console is allowed in the system.

Night Service — Night Station Service

Redirects incoming trunk calls to the attendant to designated extension numbers whenever the system is placed in Night Service. The feature is activated when the attendant has pressed the Night button on the principal console and a night console is not assigned or not activated.

Night Service — 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 telephones have not been designated to answer these calls. The incoming call activates a gong, bell, or chime, and a telephone user dials an access code to answer the call.

Night Service — Trunk Group

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 trunk group night service are in the Individual Trunk Night Service Mode so that calls coming into this trunk group are redirected to its designated night service extension. Incoming calls on other trunk groups are processed normally.

Off-Premises Station

Allows a telephone located outside the building where the system is located to be connected to the system. If central office trunks are used, the telephone must be analog and must be registered by the appropriate government agency. This feature is useful whenever it is necessary to have a telephone located away from the main location. The maximum loop distance allowed is 20,000 feet or 6,093.34 meters without repeaters.

Personal Computer Interface

Consists of the Personal Computer/Public Branch Exchange Connection, the Personal Computer/Public Branch Exchange Platforms, and the Personal Computer/Information System Digital Network Platform product family. These products are used with the system to provide users of IBM compatible PCs fully integrated voice and data workstation capabilities. Personal Computer/Public Branch Exchange Connection is a voice and data call management application. The platforms consist of personal computer interface cards and open voice and data Applications Programming Interfaces. These interfaces allow a variety of applications to operate on the personal computer and communicate with the system.

Personal Computer/Private Branch Exchange Connection

Brings the telephone and IBM-compatible personal computer into an integrated voice and data workstation. This feature is a personal computer application that runs on the Personal Computer/Private Branch Exchange platform.

Personal Central Office Line

Provides a dedicated trunk for direct access to or from the public network for multiappearance telephone users. Each Personal Central Office Line can have an appearance at up to four multiappearance phones. Users assigned this feature press the feature button to answer and place calls. An incoming call on this line rings all telephones assigned the feature, and ringing can be either audible or silent, depending on administration.

Personalized Ringing

Allows users of certain telephones to uniquely identify their own calls. Each user can choose one of a number of possible ringing patterns. The eight ringing patterns are tone sequences consisting of different combinations of three tones. With this feature, users working closely in the same area can each specify a different ringing pattern in order 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 feature provides certain telephones with the capability to access the local central office and to answer certain incoming calls during a power failure. These telephones can be used to make or answer important or emergency calls.

Priority Calling

Allows users to ring another telephone with a distinctive signal that tells the called party 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 pressing a feature button, followed by the desired extension number. A user can use Priority Calling if they have the required class of service.

Privacy — Attendant Lockout

Prevents an attendant from reentering a multiple-party connection held on the console unless recalled by a telephone user. This feature is administered on a system-wide basis. It is either activated or not activated.

Privacy — Manual Exclusion

Allows multiappearance 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.

Private Network Access (PNA) Software

Provides Automatic Alternate Routing, seven-digit uniform dialing, subnetwork trunking, and Private Network Access. This software must be used when the system is part of an Electronic Tandem Network.

Pull Transfer

Provides a convenient way to connect a party with someone better qualified to handle the call. Attendant assistance is not required and the call does not have to be redialed. This feature is an enhancement of the standard transfer operation and allows *either* the party who was originally called *or* the party to whom the held call will be transferred to complete the transfer

Q-SIG Global Networking

Provides compliance with the European Computer Manufacturers Association Integrated Services Digital Network-Private Rate Interface specifications. The Basic Call complies with standards 142 for layer 2 and 143 for layer 3 to support private network connectivity. This means it supports the Integrated Services Digital Network-Private Rate Interface connection from system-to-system as long as both systems support the same protocol. The Q-SIG Global Networking Platform provides a private network protocol for true multivendor system-to-system connectivity. This platform meets requirements for the European Computer Manufacturers Association Private Network Generic Functional Procedures for Call-Related supplementary services and is also consistent with the International Standards Organization Private Network Supplementary Services platform that is currently a draft international standard.

R2-Multifrequency Compelled Signaling

Is used primarily in national and international voice-switched networks as a robust and flexible signaling scheme for central office, direct inward dial, and tie trunks. This signaling provides a fourth addressing option, complementing the existing rotary, integrated digital services network, and dual tone multifrequency options. It can be used on both incoming and outgoing trunk calls, but only group II signaling protocol is supported on outgoing trunk calls.

Requires a TN744 Call Classifier circuit pack.

Recall Signaling

Allows users on an analog telephone to place a call on hold and consult with another party or activate a feature. After consulting with that third party, the user can conference the third party with the original party by another recall signal, or return to the original party by pressing Recall twice or by flashing the switchhook twice.

Recent Change History

Allows the system manager to view or print a history report of the most recent administration and maintenance changes on the system. This report may be used for diagnostic or information purposes.

Recorded Announcement

Provides a recorded announcement to a variety of calls: calls that cannot be completed as dialed, calls that have been in queue for an assigned interval, any calls whose destination is an announcement, or incoming calls to a user.

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 Call Coverage

Provides automatic redirection of certain calls to remote coverage points when the user and any other coverage points are unavailable by enabling a remote coverage point to be added to a user's call coverage path. Remote Call Coverage paths can be any valid dialed number up to 16 digits that begin with Automatic Alternate Routing or Automatic Route Selection.

Report Scheduler and System Printer

Allows the system manager to schedule selected reports to be printed by an asynchronous printer. Reports can be scheduled at 15-minute intervals for any combination of days of the week. A variety of reports showing system activity and status can be printed using the Report Scheduler and the System Printer.

Restriction — Controlled

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, total, station-to-station, and termination restrictions.

Restriction — Fully Restricted Service

Restricts assigned telephones from having access to public network calls. This feature is controlled by the class of restriction assigned to the calling telephone. Telephones with this class of restriction have access to internal calls only, and users cannot use authorization codes to deactivate this feature. All calls from outside the system to this telephone receive intercept tone or are routed to an attendant.

Restriction — Miscellaneous Terminal

Restricts callers at specified telephones from accessing certain other telephones. This feature is controlled by the class of restriction assigned to the calling telephone and to the telephone being called. Any class of restriction can be administered to allow or deny access to any other class of service. Restricted calls are routed to intercept tone.

Restriction — Miscellaneous Trunk

Restricts users at specified telephones from accessing certain trunk groups, such as Wide Area Telecommunications System. This feature is controlled by the class of restriction assigned to the calling telephone and to the trunk group being accessed. Any class of restriction can be administered to allow or deny access to any other class of restriction. Restricted calls are routed to intercept tone.

Restriction — Toll

Restricts users at specified telephones from placing calls that have been designated as toll calls by system administration. With this feature, a Toll Analysis table can be administered to assign certain dialed digit strings to a "toll list." A call containing one of these digit strings is designated as being a "toll call." When a user attempts to dial a toll call, he or she may not be able to place the call, depending on his or her assigned class of restriction. If a user is restricted from making an attempted toll call, the user receives intercept tone.

Restriction — Station — Inward

Restricts callers at specified telephones from receiving public network calls or calls from the attendant (including call transfers). This feature is controlled by the class of restriction of the called telephone. Denied calls are routed to intercept tone, a recorded announcement, or the attendant.

Restriction — Station — Manual Terminating Line

Restricts callers at specified telephones from receiving calls other than those from an attendant. This feature is controlled by the class of restriction of the called telephone. All other calls are routed to intercept tone, a recorded announcement, or an attendant.

Restriction — Station — Origination

Restricts callers at specified telephones from originating calls. Users can only receive calls. A user can, however, activate other features by dialing the assigned feature access code. This feature is controlled by the class of restriction of the calling telephone. If a telephone user attempts to place a call, intercept tone is received.

Restriction — Station — Outward

Restricts callers at specified telephones from placing calls to the public network. Calls can be placed to other telephone users, to the attendant, and over tie trunks. Calls can be extended to this telephone by the attendant or by other telephone users. This feature is controlled by the class of restriction of the calling telephone. If a telephone user attempts to place a call, intercept tone is received.

Restriction — Station — Public

Restricts callers at specific telephones from receiving public network calls. The users can receive network calls going through the attendant and redirected network calls. This feature is controlled by the class of restriction of the called telephone. A denied call is routed to intercept tone, a recorded announcement, or the attendant.

Restriction — Station — Termination

Restricts callers at specified telephones from receiving any calls. The restricted users can, however, originate calls. This feature is controlled by the class of restriction of the calling telephone. A denied call is routed to intercept tone.

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 when a trunk becomes available, and hears a distinctive three-burst alerting signal when called back.

Ringer Cutoff

Allows the user of a multiappearance telephone to turn certain audible ringing signals on and off. Visual alerting is not affected by this feature. When this feature is enabled, only Priority ring, Intercom ring, and Manual signaling ring at the telephone.

Rolling Average Speed of Answer

Allows calls to be routed based on a weighted average of the most recent calls versus the last time interval. Enables administrators to assign a higher handling priority for important calls.

Rotary Dialing

Enables rotary dialing telephones to be used with the system. The software recognizes that a telephone is rotary when the user lifts the handset, and expects to receive dial pulses instead of touch tones.

Security Violation Notification

Notifies a designated referral point (attendant console, display equipped telephone, or a referral call with announcements) of a security violation. This feature provides the capacity to disable a valid login ID or remote access following a security violation. This feature also provides an audit trail containing information about each attempt to access the system.

Send All Calls

Allows users to temporarily send all incoming calls directly to coverage. Send All Calls is activated by pressing the feature button or by dialing an access code. This temporarily direct all incoming calls to coverage regardless of the assigned Call coverage redirection criteria. To deactivate Send All Calls, the user presses the button a second time or dials the access code again.

Senderized Operation

Reduces the time necessary to place calls to distant locations equipped to receive dual tone multifrequency (touch-tone) signals and allows end-to-end signaling to remote computer equipment. The number dialed and end-to-end signaling digits from telephones and trunks are detected by the system and regenerated for transmission over outgoing trunks. The distant end associated with the trunk must be equipped to receive touch-tone signals.

Straightforward Outward Completion

Allows an attendant to complete an outgoing trunk call for a telephone user, without requiring the telephone user to hang up.

Subnet Trunking

Provides modification of the dialed number so an Automatic Alternate Routing and Automatic Route Selection call can route over trunk groups that end at systems with different dial plans. This feature provides digit insertion, deletion, pauses, and/or wait for dial tone in digit outpulsing, as required, to permit calls to route to or through a remote system, over tie trunks to a private network, over central office trunks to the serving central office. This feature is not required on calls to a party at the local system, but is required on calls routing to or through a remote system, regardless of the call's destination.

System-Based Bulletin Board

Provides a bulletin board on the system that allows Lucent Technologies technical personnel and your company's personnel to communicate with each other. Any administration terminal user who has appropriate permissions is able to leave and receive messages on the System-Based Bulletin Board. This feature facilitates communication between people who are responsible for administering and servicing the system.

System Measurements

Provides reports on items such as trunk group usage, hunt group usage and efficiency, attendant group activity and efficiency, and security violations. There are many reports that can be generated using this feature. All system measurement reports are on-demand reports. None are provided automatically. Reports are available through standard system management or a remote administration terminal, and can be printed if a printer is associated with the terminal. The reports can also be scheduled to print at the system printer via the Report Scheduler and System Printer feature.

System Status Report

Allows the system manager or maintenance personnel to view data associated with attendants, major and minor alarms, and traffic measurements. The information is displayed on the management terminal and presents a basic picture of the system's condition.

Temporary Bridged Appearance

Allows multiappearance telephone users in a Terminating Extension Group or Personal Central Office Line Group to bridge onto an existing group call without manually transferring the call. In addition, if a call has been answered using the Call Pickup feature, the originally called party can bridge onto the call. This feature also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it. This feature provides convenience of operation and saves time.

Terminal Translation (with Security Measures)

Allows personnel administering the system to associate or disassociate a telephone's administration-without-hardware translation to a valid port address by dialing a special digit sequence from a terminal connected to the port. This feature, and the Administration Without Hardware feature, is expected to reduce costs in the initial provisioning of a system and also help in telephone and data module moves form office to office. A security code is used to prevent unauthorized use of this feature.

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 multiappearance 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 bridge onto the call.

Time-of-Day Routing

Provides the most economical routing of Automatic Route Selection and Automatic Alternate Routing calls, based on the time of the day and day of the week that each call is made. With this feature, a company can take advantage of lower calling rates during specific times of the day and week. In addition, companies with locations in different time zones may be able to maximize the use of facilities by utilizing those in a location that has a lower rate at different times of the day or week. This feature can also be used to change the patterns during the times an office is closed in order to reduce or eliminate unauthorized calls.

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 incoming calls answered by a telephone user, but which are unanswered after being transferred. 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.

Touch-Tone Dialing

Provides quick and easy pushbutton dialing using Dual-Tone Multifrequency (or Touch-Tone) Dialing. A distinctive tone is generated when each button is pressed. If a distant switching system can accept only dial pulse (rotary dialing) signals, the system converts the touch-tone signals to the required dial pulses for transmission to the distant end.

Transfer

Allows telephone users to transfer trunk or internal calls to other telephones within the system without attendant assistance. This feature provides a convenient way to connect a party with 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. Multiappearance telephone users press the Transfer button, dial the desired extension number, and press the Transfer button again.

Transfer — Outgoing Trunk to Outgoing Trunk (with Security Measures)

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. This is an enhancement to Trunk-to-Trunk Transfer and requires careful administration and use.

Trunk Flash

Enables multifunction telephones to access "central office customized services" that are provided by the far-end central office located directly behind the system. These central office services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signals and dialing from the telephone on an active trunk call. Some versions and models of the system support "Digit 1 as Flash" as used in Italy and the United Kingdom.

Trunk Group Busy/Warning Indicators to Attendant

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 to show the attendant that the Attendant Control of Trunk Group Access feature needs to be invoked.

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.

Uniform Dial Plan

Provides a common four-or five-digit dial plan that can be shared among a group of systems. Intersystem dialing and intrasystem dialing both require four- or five-digit dialing. The Uniform Dial Plan is used with Extended Tandem Network configurations. Additionally, this feature can be used alone to provide uniform four- or five-digit dialing between two or more private switching systems without any of the configurations listed above.

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: a feature activation/deactivation button, console status button, display status button, last operation button, last voiced message button, and direct trunk group selection status button. 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 messages in the form of a voice output. Voice choices are UK English, US English, and Italian; these choices depend up on the voice synthesis circuit pack provided. Messages are protected by restricting unauthorized users from retrieving messages. This feature is activated by dialing a dial access code.

Voice Terminal Display

Provides multiappearance telephone users with updated call and message information. This information is displayed on a display-equipped terminal. The information displayed depends upon the display mode selected by the user. Users may select any of the following as the display message language: English, French, Italian, Spanish. In addition, messages can be administered on the system in a fifth language. The language for display messages is selected by each user. This feature provides an instant display of information associated with certain system features, functions, and services. Information that allows personalized call answering is available on many calls. Retrieving stored information, such as messages and directory information, is easy as well as convenient.

Voice Terminal Alerting Options

Provides certain multiappearance telephone users with different ringing patterns. This feature primarily affects audible ringing for calls directed to telephones that are off hook.

Voice Terminal Flash Timing

Allows users on an analog telephone to place a call on hold and consult with another party or activate a feature. After consulting with that third party, the user can conference the third party with the original party by another recall signal, or return to the original party by pressing Recall twice or by flashing the switchhook twice.

World Class Tone Detection

Enables the system to identify and handle different types of call progress tones, depending on the system administration. "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 board's detect mode. The "level of tone detection broadband" is used in countries that have a discontinuous dial tone.

World Class Tone Generation

Allows administrators to specify the base call progress tone set to be generated by the system. Administrator can then customize the set by selecting different values for frequency and cadency for up to 6 individually administrable tones.

Features Unique to the DEFINITY Business Communications System

The following features are unique to the DEFINITY Business Communications System:

- Attendant Backup
- Attendant Crisis Alert
- Controlled Toll Restriction
- Direct Access Calling (similar to Call Vectoring).

Attendant Backup

The Attendant Backup feature allows you to access most attendant console features from one or more specially-administered multiappearance voice terminals. Using this backup mode, you can answer calls faster, thus providing better service to your customers and prospective clients.

The recommended voice terminals are the Lucent Technologies Model 8434 and Model 8410. When calls terminate at the attendant console during normal operation, users at the backup voice terminals can answer overflow calls by pressing a button or dialing a feature access code. You can then process the calls as if you are at the attendant console. Procedures for basic feature operation are documented in the quick reference guides for each voice terminal.

When the attendant console is in the day mode (the (Night) lamp is off), you cannot answer overflow calls at the backup voice terminals until the number of calls waiting in the attendant queue has reached an administered threshold. Until the threshold is reached, the only indication the backup voice terminals receive about calls waiting in queue is when the (Queue Calls) and the (Queue Time) lamps go on. You can press the (Queue Time) button to see how long the call has been waiting, but you cannot answer the call. The (Queue Time) lamp starts flashing when the time in queue warning level has been reached (this is usually set for 15 seconds). When the calls waiting in queue threshold has been reached, the backup voice terminals will beep every 10 seconds as long as the number of calls waiting stays above the threshold. You can then answer calls using the Trunk Answer Any Station feature access code or an automatic dialing button administered with that feature access code.

When the attendant console is in the night mode (the (Night) lamp is on), all calls to the attendant console immediately beep at the backup voice terminals and the (Queue Calls) and the (Queue Time) lamps go on. You can then answer calls using the Trunk Answer Any Station (TAAS) feature access code or an automatic dialing button administered with that feature access code.

You can also install an external ringing device that sounds whenever the attendant queue limit is reached and when calls can be answered with the TAAS feature access code. This is an optional feature.

User Operation

The Attendant Backup feature has no specific user operation procedures. The user operation for basic console features using the attendant console or a backup voice terminal is documented in *DEFINITY Communications System Console Operations Quick Reference,* (555-230-890). For basic voice terminal operation, see the voice terminal user guides.

Administration

change station XXXX (**XXXX** is the extension number of the voice terminal)

- Add the following required feature buttons to the backup voice terminal:

    — **atd-qcalls** (this button causes backup voice terminals to ring when the queue warning level is reached)

    — **abrv-dial** (this is a button that is programmed with the TAAS feature access code and labeled Attendant Call Pickup)

    — **atd-qtime**

    — **night-serv** (only one backup voice terminal can have this button)

    — **ringer-off**

    — **busy-ind** (assign a busy indicator button for the attendant console extension and for the extensions of other backup voice terminals)

change attendant X (**X** is the console number; usually 1)

- Use this form to administer the attendant console parameters. On Page 1 of this form, the console `Type` field must be set to **principal**.

change feature access-codes

- Use this form to add the Trunk Access Any Station feature access code. This feature is used to answer attendant calls that overflow the attendant call waiting queue.

change system console-parameters

■ On Page 1 of this form, enter a value for the `Calls in Queue Warning` field (1-30 calls) and the `Time in Queue Warning` field (1-300 seconds).

Enter the equipment location of an unused analog circuit in the `Ext Alert Port (TAAS)` field. This identifies the port where you can connect an external ringing device to alert backup attendant personnel that there are calls waiting in the attendant queue. An analog circuit must be administered to enable the Attendant Backup feature even if you do not have a ringing device attached to the port.

change restriction cos

■ For the COS used by the backup voice terminals, enter **y** for `Console Permissions.`

> ⇒ **NOTE:**
> Make sure that the COS used by the backup voice terminals are not assigned to any other voice terminals. You can restrict users from accessing the TAAS feature access code by assigning their extensions to a COS that has the "client room" COS enabled.

change system introworks-options

■ Enter **y** in the `Day Mode TAAS Pickup & Backup Station Audible Alerting` field.

Required Hardware

The Attendant Backup feature can be used from any supported multiappearance voice terminal. The recommended models are the Model 8434 and Model 8410.

If you are using the TAAS external alerting port, you must use an approved ringing device.

Attendant Crisis Alert

The Attendant Crisis Alert feature provides a visual, audible, and printed record when users place a call to the local emergency service agency. This gives attendants the ability to assist emergency personnel when they arrive at the site by identifying where the call came from and when the call was made. This feature uses the Automatic Route Selection (ARS) feature to allow routing of any emergency service access code (such as 911) to the appropriate emergency service agency, while also identifying the call for crisis alerting.

For example, users should dial 911 to reach the local emergency service agency. When the call is placed and successfully routed to the local emergency service agency, the attendant console is notified immediately by a special emergency alerting tone and a special emergency display (the emergency call itself cannot be answered at the attendant console, but the call information is displayed). The attendant can then note the extension number and contact the appropriate personnel to assist with the emergency.

$\Rightarrow$ NOTE:
Each subsequent emergency notification is queued with a 5-second delay to allow the attendant to finish processing the current emergency notification.

The Attendant Crisis Alert feature can be used for any type of emergency such as a medical emergency, a fire in the boiler room, or a burglary.

User Operation

Other than the emergency call, which can be placed from any telephone on the system, all user operation occurs at the attendant console.

1. Someone dials the emergency services access code (for example, 911) from a telephone on the system.

   - The call is routed to the local emergency service agency. The call **does not** route to the attendant console.

   - The **Position Available** lamp goes off and the (Pos Busy) lamp goes on. This prevents new incoming calls from interrupting this emergency notification. All new incoming calls are queued and can be answered after the emergency notification is processed.

   - The (Crisis Alert) lamp flashes.

   - The special emergency alerting tone starts.

   - The following is displayed at the attendant console:

     ```
     E=  <Name>        <Ext No.>    EMERGENCY
     ```

   - The call information is logged in the system and is printed on the journal/schedule printer (if administered).

2. If you are currently on an active call, you may want to place that call on hold so you can process the emergency notification.

3. Press the [Crisis Alert] button once.

   ■ The alerting tone stops.

4. Write down the emergency information displayed on the console. Follow your local procedures for handling emergencies. Even though the console is set to "position busy," you can place calls to assist with the emergency.

5. Press the [Crisis Alert] button a second time.

   ■ The [Crisis Alert] lamp stops flashing, but remains on.

6. When you are finished handling the emergency, press the [Crisis Alert] button a third time.

   ■ The [Crisis Alert] lamp goes off.

   ■ The display goes blank.

7. Press the [Pos Busy] button.

   ■ The [Pos Busy] lamp goes off.

   ■ The *Position Available* lamp goes on.

8. You can now process other incoming calls.

Administration

change attendant 1

■ On Page 2 of this form, add the **crss-alert** feature button. Using a blank button label, create a [Crisis Alert] button label and install it on the attendant console. The [Crisis Alert] button can be added only to the attendant console, not to any of the attendant backup voice terminals.

change network ars analysis X (**X** is the first digit of the dialed digit string)

■ On this form, assign a routing pattern and the **alrt** Call Type to the desired emergency service access code. For example, if your emergency service access code is 911, assign the 911 digit string to a routing pattern and assign it the **alrt** Call Type. This takes care of the condition when the user dials 9 (for local access) and then 911. If a user dials only 911, you also want the call to route to the emergency service agency. You must assign a dialed string of 11 with a different routing pattern that removes the dialed digits 11 and inserts the dialed digit string 911. The following screens show these two examples administered as part of the ARS Digit Analysis Table.

```
change network ars analysis 9                          Page   1 of   2
                        ARS DIGIT ANALYSIS TABLE
                     Partitioned Group Number:  1      Percent Full:    6

        Dialed     Total Rte Call Nd ANI      Dialed     Total Rte Call Nd ANI
        String     Mn Mx Pat Type Num Rq      String     Mn Mx Pat Type Num Rq
911                3  3  5   alrt     n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
```

```
change network ars analysis 1                          Page   1 of   2
                        ARS DIGIT ANALYSIS TABLE
                     Partitioned Group Number:  2      Percent Full:    6

        Dialed     Total Rte Call Nd ANI      Dialed     Total Rte Call Nd ANI
        String     Mn Mx Pat Type Num Rq      String     Mn Mx Pat Type Num Rq
11                 2  2  6   alrt     n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
                                      n                                       n
```

change network route-pattern X (**X** is the routing pattern)

- On this form, assign a routing pattern for the emergency service access code.

  In this first example, Preference 1 of Pattern 5 is used when users dial 9911 (9 for the ARS access code, and 911 for the emergency service agency).

```
change network route-pattern 5                              Page   1 of   1
                         Pattern Number: 5

   Grp.   FRL NPA Pfx Hop Toll No. Del Inserted                       IXC
   No.             Mrk Lmt List Digits  Digits
1: 5       7
2:
3:
4:
5:
6:

    BCC VALUE    TSC  CA-TSC     ITC   BCIE  Service/Feature        Numbering
    0 1 2 3 4 W       Request                                       Format
1: y y y y y n   n               rest
2: y y y y y n   n               rest
3: y y y y y n   n               rest
4: y y y y y n   n               rest
5: y y y y y n   n               rest
6: y y y y y n   n               rest
```

In this second example, Preference 1 of Pattern 6 is used when users dial 911. Pattern 6 deletes the two digits dialed after the ARS access code (11), and inserts the correct digit string (911).

```
change network route-pattern 6                              Page   1 of   1
                         Pattern Number: 6

   Grp.   FRL NPA Pfx Hop Toll No. Del Inserted                       IXC
   No.             Mrk Lmt List Digits  Digits
1: 6       7                     2   911
2:
3:
4:
5:
6:

    BCC VALUE    TSC  CA-TSC     ITC   BCIE  Service/Feature        Numbering
    0 1 2 3 4 W       Request                                       Format
1: y y y y y n   n               rest
2: y y y y y n   n               rest
3: y y y y y n   n               rest
4: y y y y y n   n               rest
5: y y y y y n   n               rest
6: y y y y y n   n               rest
```

Required Hardware

There is no special hardware required for this feature.

Controlled Restrictions

The Controlled Restrictions feature allows you to activate different types of calling restrictions on user telephones. The restriction types include the following:

- Outward — The user cannot place calls to the public network.

- Station-to-Station — Users cannot place or receive calls between themselves.

- Termination — The user cannot receive any calls.

- Total — The user cannot place or receive any calls.

- Toll — The user cannot place toll calls, but can place local free calls.

The Controlled Toll Restriction feature is a new option with the DEFINITY Business Communications System and can be substituted for either Outward Restriction or Station-to-Station Restriction. Through administration, you can enable Outward/Toll Restriction, Station-to-Station/Toll Restriction, Termination Restriction, and Total Restriction.

The ways to activate controlled restrictions are as follows:

- Using a feature access code from the attendant console or from a voice terminal with console permissions, you can enable any of the controlled restrictions for a user's telephone.

- Using a feature access code from the attendant console or from a voice terminal with console permissions, you can enable any of the controlled restrictions for a group of telephones on the system. This grouping is based on the administered Class of Restriction (COR).

When a user tries to make a call from a station that is restricted, the call is routed to one of the following: the attendant, a recorded announcement, a Call Coverage path, another extension (for example, one of the backup voice terminals), or intercept tone.

User Operation

To activate a controlled restriction for one user using the attendant console or a voice terminal with console permissions, do the following:

1. Press the (Start) button or an idle call appearance button.

   ■ You hear a dial tone.

   ■ The call appearance lamp goes on.

   ■ The **Position Available** lamp goes off.

2. Dial the User-Controlled Restriction activation feature access code followed by one of these four digits:

   ■ Dial (1) for Outward/Toll Restriction

   ■ Dial (2) for Total Restriction

   ■ Dial (3) for Termination Restriction

   ■ Dial (4) for Station-to-Station/Toll Restriction.

3. After hearing the second dial tone, dial the extension number.

   ■ A confirmation tone indicates that the restriction was activated. An intercept tone indicates that the extension number already has a restriction assigned, or you dialed an improper digit.

   ■ The call appearance lamp goes off.

   ■ The display goes blank.

   ■ The **Position Available** lamp goes on.

   ■ The console returns to the normal operating mode.

To deactivate a controlled restriction for one user, do the following:

1. Press the (Start) button or an idle call appearance button.

   ■ You hear a dial tone.

   ■ The call appearance lamp goes on.

   ■ The **Position Available** lamp goes off.

2. Dial the User-Controlled Restriction deactivation feature access code followed by the digit that represents the current restriction:

   ■ Dial (1) for Outward/Toll Restriction

   ■ Dial (2) for Total Restriction

   ■ Dial (3) for Termination Restriction

   ■ Dial (4) for Station-to-Station/Toll Restriction.

3. Dial the extension number. You hear one of the following:

    - Confirmation tone if the restriction code was accepted.

    - Intercept tone if you dialed an improper restriction code number. Press ⏍Cancel⏍ and start over again.

4. Press ⏍Release⏍.

    - The call appearance lamp goes off.

    - The display goes blank.

    - The **Position Available** lamp goes on.

    - The console returns to the normal operating mode.

To activate a controlled restriction for a group of extensions, do the following:

1. Press the ⏍Start⏍ button or an idle call appearance button.

    - You hear a dial tone.

    - The call appearance lamp goes on.

    - The **Position Available** lamp goes off.

2. Dial the Group Controlled Restriction activation dial access code followed by one of these four digits:

    - Dial ⏍1⏍ for Outward/Toll Restriction

    - Dial ⏍2⏍ for Total Restriction

    - Dial ⏍3⏍ for Termination Restriction

    - Dial ⏍4⏍ for Station-to-Station/Toll Restriction.

3. After hearing the second dial tone, dial the two-digit COR number of the group.

    - A confirmation tone indicates that the restriction was activated. An intercept tone indicates that the group of extensions already have a restriction assigned, or you dialed an improper digit.

    - The call appearance lamp goes off.

    - The display goes blank.

    - The **Position Available** lamp goes on.

    - The console returns to the normal operating mode.

To deactivate a controlled restriction for a group of extensions, do the following:

1. Press the ⟨Start⟩ button or an idle call appearance button.

   ■ You hear a dial tone.

   ■ The call appearance lamp goes on.

   ■ The **Position Available** lamp goes off.

2. Dial the Group Controlled Restriction deactivation feature access code followed by the digit that represents the current restriction:

   ■ Dial ① for Outward/Toll Restriction

   ■ Dial ② for Total Restriction

   ■ Dial ③ for Termination Restriction

   ■ Dial ④ for Station-to-Station/Toll Restriction.

3. Dial the two-digit COR number of the group. You hear one of the following:

   ■ Confirmation tone if the restriction code was accepted.

   ■ Intercept tone if you dialed an improper restriction code number. Press ⟨Cancel⟩ and start over again.

4. Press ⟨Release⟩.

   ■ The call appearance lamp goes off.

   ■ The display goes blank.

   ■ The **Position Available** lamp goes on.

   ■ The console returns to the normal operating mode.

Administration

change system introworks-options

■ Use this form to assign Toll Restriction as a substitute for either Outward or Station-to-Station restriction. Enter **nothing**, **outward**, or **station-station** into the `Controlled Toll Restriction replaces` field.

If you enter **nothing**, you have access to Outward, Total, Termination, and Station-to-Station restrictions. If you enter **outward**, you have access to Toll, Total, Termination, and Station-to-Station restrictions. If you enter **station-station**, you have access to Outward, Total, Termination, and Toll restrictions.

change system feature-parameters

■ On Page 3 of this form, add the intercept treatment desired for the Outward/Toll, Termination, and Station-to-Station restricted calls in these fields:

    — `Control Outward/Toll Restriction Intercept Treatment`

    — `Controlled Termination Restriction`

    — `Controlled Station to Station Restriction`

Callers that encounter one of these restrictions can be routed to an announcement, the attendant, Call Coverage (for Termination Restriction only), an extension, or to intercept tone. If you select announcement or extension, you must enter the appropriate extension number.

If restricted calls are routed to a recorded announcement, the specific announcement must be recorded and assigned to the correct extension number.

change feature access-codes

■ Enter feature access codes for the Group-Controlled Restrictions and the User-Controlled Restrictions.

Required Hardware

There is no special hardware required for this feature.

Direct Access Calling

Direct Access Calling uses the integrated announcement hardware of the system to deliver short messages to the caller, such as "Hello. You have reached A1 Corporation. Please press 1 for the operator, press 2 to reach an employee, or press 3 for the help desk." If the caller selects 1, the call routes to the operator. If the caller selects 2, the caller is then prompted to dial the extension number. If the caller selects 3, the call routes to the companies' help desk. This automated attendant application speeds call handling and saves time for attendants.

Another application of this feature is to provide your employees a listing of services and events by way of a daily menu of offerings. Your employees can call a number and be given a menu of choices.

User Operation

The only user operation required for this feature depends on the choices given in the recorded announcements. The caller must follow the prompts and select those options chosen. In most cases, no action by the caller will result in the call being routed to the attendant console.

Administration

> **NOTE:**
> If you use a Direct Access Calling procedure to route calls to a location outside of your location, the Class of Restriction (COR) of the Direct Access procedure must route using its own ARS restricted partition to prevent toll fraud.

add feature direct-access number XXXX (**XXXX** is the extension number)

- Use this form to specify which Direct Access Calling procedure number (1 through 4) callers will access when the Direct Access Calling number is dialed. There can be four different Direct Access Calling numbers. One of the numbers is usually the published telephone number for the company. A second number could be used by employees to provide a menu of information about services and events. After you add a number, you can later change the number if needed.

change feature direct-access procedure X

■ You can assign up to four procedures that define how calls will be handled as users select the different prompts. The following example shows a Direct Access procedure. Contact your authorized dealer for support in setting up your procedures.

```
change feature direct-access procedure 1                    Page   1 of   2
                           DIRECT ACCESS PROCEDURE

 Procedure: 1                     Name auto-attd-1


 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 route-to     number 699             with cov n if digit         = 3
 08 goto         step   20  if digits            =     4
 09 goto         step   16  if digits            =     5
 10 route-to     number 0                with cov n if unconditionally
```

```
change feature direct-access procedure 1                    Page   2 of   2
                           DIRECT ACCESS PROCEDURE


 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
 20 collect      3  digits after announcement 383
 21 goto         step   13  if unconditionally
```

The procedure above does the following:

1. When a client calls your system, they receive ringback for 2 seconds.

2. Announcement 381 plays. This announcement asks them to do one of the following:

   - Press ⓪ or wait if they want the operator; if they press ⓪ or wait for the timeout, they are routed to the operator.

   - Press ① if they want the help desk; if they press ①, they are routed to extension 105, which is the help desk.

   - Press ② if they know the person's extension; if they press ②, they are routed to announcement 382, which tells them to dial the person's extension.

   - Press ③ if they want to retrieve their voice messages; if they press ③, the call is routed to the voice messaging system.

   - Press ④ if they know the department they wish to access (such as engineering); if they press ④, they are routed to announcement 383, which gives them a listing of several departments that they can dial directly.

   - Press ⑤ to start over again; if they press 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 board (TN750C) is required for this feature.

System Capacity Limits

B

The maximum parameters for the DEFINITY Business Communications System hardware and software items are listed in Table B-1. Terminal and digital station capacities are reduced by such administered items as attendant consoles.

➡️ **NOTE:**
Not all maximum capacities listed in the following table can be reached simultaneously with all configurations of the system.

Table B-1.   Capacity Limits

Feature	Capacity
Abbreviated Dialing (AD)	
AD Lists Per System	400
AD List Entry Size	24
AD Entries Per System	2,000
Auto Dialing Button	
Entries per System[1]	
Enhanced List (System List)	1
Maximum entries	2,000
Group Lists	100
Maximum entries	100
Group lists/extension	3
System List	1
Maximum entries	100
Personal Lists	400
Maximum entries	100
Personal lists/extension	3
Applications Adjuncts	
Asynchronous Links (RS232)	5
SMDR Output Devices	2
Journal:System Printer	2:1
BX.25 Physical Links	4
AUDIX Adjuncts	1
ICM Adjuncts (ISDN Gateway)	1
BX.25 Processor Channels	64
Hop Channels	64

Table B-1. Capacity Limits

Feature	Capacity
ARS/AAR	
ARS/AAR Patterns	40
ARS/AAR Table Entries (NPA,NXX,RXX,HNPA,FNPA)	2,000
Choices per RHNPA Table	12
Digit Conversion Entries	400
ARS/AAR Digit Conversion	
Digits Deleted for ARS/AAR[2]	28
Digits Inserted for ARS/AAR	18
Entries in HNPA & RHNPA Tables	1,000
FRLs	8
Inserted Digit Strings[3]	450
Patterns for Measurement	20
RHNPA Tables	32
Routing Plans	8
Toll Tables	32
Entries per Toll Table	800
Trunk Groups in an ARS/AAR Pattern	6
UDP (Entries)	240
TOD Charts	8
Attendant Service	
Attendant Consoles (day:night)	6:1
Attendant Console 100s Groups/Attendant	20
Attendant Control Restriction Groups	96
Other Access Queues	
Maximum Number of Queues	12
Maximum Number of Queue Slots	30
Size range of Reserved Queue	2-25
Reserved Queue Default Size	5
Queue Length	30
Switched Loops/Console	6
Authorization	
Authorization Codes	1500
Classes of Restriction	96
Classes of Service	16
Length of Authorization Code	4-7
Length of Barrier Code	4-7
Length of Forced Entry Account Codes	1-15
Restricted Call List	1
Remote Access Barrier Codes	10
SMDR Forced Entry Account Code List	1
Toll Call List	1
Unrestricted/Allowed Call Lists	10
Total Call List Entries	1,000
Automatic Callback Calls	240

Table B-1.   Capacity Limits

Feature	Capacity
Cabinets	
Inter-Port Network Connectivity	
Port Networks	1
Maximum Number of Port Networks/Cabinet	1
PPN	
SCC	3
Call Appearances	
Bridged Images/Appearance	7
Call Appearances/Station[4]	54
Maximum Appearances per Extension	10
Minimum Appearances per Extension	0
Total Bridged Appearances	400
Maximum Simultaneous Off-Hook per Call[5]	5
Call Coverage	
Coverage Answer Groups (CAG)	30
Coverage Paths	150
Coverage Paths Included in Call Coverage Report	15
Coverage Path per Station	4
Coverage Points in a Path	3
Max Users/Coverage Path[6]	500
Members per CAG	8
Number of Coverage Paths for which each Station can be a member	150
Call Detail Recording	
CDRU Trackable Extensions	400
Intrasystem Call Trackable Extensions	100
Number of CDRUs/System[7]	1
Maximum Number of CDR Records that can be Buffered in the System	300
Number of Records Buffered for the Primary Output Device That Will Cause Secondary Device to be Busied Out for 2 Minutes	200
Call Forwarding (Follow-me)	
Call Forwarded Digits (off-net)	16
Call Forwarded Numbers	400
Call Park	
Attendant Group Common Shared Extension Numbers	10
Number of Parked Calls	180
Call Pickup Groups	
Call Pickup Members/Group	50
Call Pickup Members/System	200
Number of Groups	100
Conference Parties	6
Simultaneous 3-way Conference Calls	161
Simultaneous 6-way Conference Calls	80

Table B-1. Capacity Limits

Feature	Capacity
Data Parameters	
Alphanumeric Dialing	
Maximum entries	50
Characters/Entry	22
PRI Endpoints (PE)	25
Access Endpoints (Number of Trunks)	100
Digital Data Endpoints	75
Dial Plan	
DID LDNs	8
Extensions	700
Feature Dial Access Codes	
Number of Access Codes	70
Number of Digits	1-4
Integrated Directory Entries[8]	407
Maximum Extension Size	5
Minimum Extension Size	1
Miscellaneous Extensions[9]	150
Names	
Number of names[10]	664
Number of characters in a name	15
Non-DID LDNs	50
Prefix Extensions	Yes
Trunk Dial Access Codes	
Number of Access Codes	65
Number of digits	1-4
Direct Access Calling (New Feature)	
Number	4
Procedure	4
Steps per Procedure	32
Facility Busy Indicators	
Buttons per Tracked Resource	100
Number of Indicators (Station and Trunk Groups)	450
Hunt Groups	
Announcements per Group	2
Announcements per System	128
Groups	24
Group Members per Group	150
Group Members per System	150
Queue Slots per Group	200
Queue Slots per System	200

Table B-1. Capacity Limits

Feature	Capacity
Intercom Translation Table (ICOM)	
Automatic/Manual and Dial	
ICOM groups per System	10
Auto/Manual	10
Dial	10
Members per ICOM group	
Auto	32
Dial	32
Members per System	320
Last Number Dialed	
Entries/System[11]	482
Number of Digits	24
Leave Word Calling (System-Based)	
Messages Stored	650
Messages per User	10
Simultaneous Message Retrievers	60
System-wide Message Retrievers	10
Malicious Call Trace	
Maximum Simultaneous Traces	16
MLDN	
Via DID	8
Via CO	50
Paging	
Code Calling IDs	125
Loudspeaker Zones	9
Personal CO Lines (PCOL)	
PCOL Appearances	4
PCOL Lines (Trunk Groups)	15
PCOL Trunks Per Trunk Group	1
Port Circuit Pack Slots[12]	
Per PPN	
SCC Standard Reliability	52

Table B-1. Capacity Limits

Feature	Capacity
Recorded Announcements	
Analog and Auxiliary Trunk Announcements	
Queue Slots per Announcement	50
Queue Slots per System	50
Calls Connected per Announcement	
Auxiliary Trunk	50
Analog Port	50
Integrated Announcements	
Integrated Announcement Circuit Packs	**5**
Channels per Integrated Announcement Circuit Pack	16
Calls Connected per Integrated Announcement	25
Recording Time (Minutes:Seconds)	
16KB recording	8:32
32KB recording	4:16
64KB recording	2:8
Integrated Queue Slots per System	25
Total Recorded Announcements	128
System Administration	
Number Of Logins	15
Admin History File Entries	50
Simultaneous Administration Command	1
Simultaneous Maintenance Command	1
Simultaneous SM Sessions	3
Printer Queue Size	50
Speech Synthesizer Circuit Packs	6
Channels per Speech Circuit Pack	4
Terminating Extension Groups (TEG)	
TEGs	32
Users That May Share a TEG	4
Time Slots	
Simultaneous Circuit-Switched Calls	180
Total Slots	512
Time Slots for Voice and Data	483
Time Slots per Port Network	512
Tone Classifiers	
Tone Receivers (General)[13]	200
TTR Queue Size	4
Prompting TTR Queue Size	80

Table B-1. Capacity Limits

Feature	Capacity
Trunks	
DS1 Circuit Packs	5
Queue Slots for Trunks	64
PRI Interfaces via PI[14]	4
PRI Interfaces via PACCON	8
PRI Temporary Signaling Connections	
TSCs in System	164
Call Associated TSCs	100
Non Call Associated TSCs	64
Administered TSCs	32
Ringback Queue Slots	64
Total PRI Interfaces[15]	8
Trunk Group Hourly Measurements	25
Trunk Groups in the System	32
Trunk Members in a Trunk Group	99
Trunks in System (including Remote Access)	120
Measured Trunks In System	120
Voice Terminals[16]	
Associated Data Modules (e.g., 7400A)	75
BRI Stations[17]	50
Digital Stations[18]	400
Display Stations	200
Stations[19]	400
Station Button Capacity Units[18]	102,800

1. There is no limit on the maximum number of auto dial buttons (other than the system limit on button capacity).
2. Plus up to seven inter-exchange carrier (IXC) digits.
3. This is the number of available 12-character inserted-digit-strings available for ARS preferences.
4. The number of call appearances is the sum of primary and bridged appearances; at most 10 can be primary. A maximum of 54 administrable buttons are supported for the 7434D terminal — 34 buttons in the basic terminal and an additional 20 buttons in the coverage module.
5. Does not apply to conferencing.
6. The maximum number of users per coverage path is equal to the number of extensions.
7. The CDRU adjunct capacity is 40,000 calls per hour.
8. The Integrated Directory Entries = Stations + Attendant Consoles.
9. Used for PCOL groups, common shared extensions, access endpoints, administered TSCs, code calling IDs, LDNs, hunt groups, announcements, and TEGs.
10. The Number of Names = number of stations + attendant consoles + trunk groups + digital data endpoints + miscellaneous extensions.
11. The Last Number Dialed Entries = Stations + Digital Data Endpoints + Attendant Consoles.
12. Only port slots are included in this count. There may be other service circuits required which would further reduce the number of port slots available.

13. The system uses the TN744 Call Classifier for basic TTR usage as well as call prompting/call classification/MFC. In addition, the TN2182 Tone/Clock/Detector will also be used for multiple tone detection functions. The number of TN748, TN420, or TN744 boards is limited only by the number of available slots, and the number of TN2182 boards is limited to one. There is a single limit on the total number of tone receiver (classifier) ports for the system: the TN748/TN420 have four ports for TTR use, the TN748/TN420 have two ports for GPTD use, the TN744 has eight ports for call prompting/call classification/MFC/TTR use, and the TN2182 has eight ports for call prompting/Call Classification/MFC/TTR use.

14. Only one Processor Interface (PI) board is supported, and therefore a total of four physical links (used for BX.25 or PRI) are available. PRI interface via the PI is not available in Germany. PRI via the PACCON must be used.

15. Since the SCC can only support one PI board, a total of four physical links (used for BX.25 or PRI) is available. When using the PACCON, the limit bounded by the DS1 CP limit.

16. The following items detract from the total number of available "Stations" on a given system: analog music-on-hold, attendants, TAAS port, stations (digital, display, BRI, and so on), analog announcements, and analog external alarm port.

17. All BRI stations can be display stations.

18. The station button capacity cannot be exceeded based on the number and type of voice terminals administered on the system. The following examples show how this capacity is determined. Unless otherwise noted, the assumption is that only 3 call appearances (CA) are assigned to the sets (except analog sets which have no call appearances).

   - Analog set: 62 units
   - Digital set with 10 buttons: 102 units
   - Digital set with 34 buttons, no display: 342 units
   - Digital set with 34 buttons, display: 472 units
   - BRI set with 17 buttons, display: 250 units

   The following is a list of specific terminal types and the button units required for each of the terminal types. **Default** means there are no nondefault buttons administered. **Loaded** means that all possible buttons are administered.

   - 8403B (default): 32 units
   - 8403B (loaded with 3 CA): 152 units

   - 8410B (default): 32 units
   - 8410B (loaded with 3 CA): 102 units

   - 8410D (default): 292 units
   - 8410D (loaded with 3 CA): 292 units
   - 8410D (loaded with 10 CA): 362 units

   - 8434D, no expansion module (default): 252 units
   - 8434D, no expansion module (loaded with 3 CA): 562 units
   - 8434D, expansion module (loaded with 3 CA): 802 units
   - 8434D, expansion module (loaded with 52 CA): 1292 units

   - 8510 (default): 190 units
   - 8520 (default): 180 units
   - 8520 (loaded with 3 CA): 350 units
   - 8520 (loaded with 20 CA): 520 units

19. Including extensions administered without associated hardware.

Related Documents

C
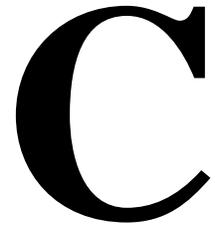
Since the DEFINITY Business Communications System is based on the Lucent Technologies DEFINITY product line, the following is a list of DEFINITY documents that you should use for reference. This appendix includes a brief description of each document. To order copies or these documents, refer to the ordering information on the back of the title page.

DEFINITY Communications System Generic 3   **555-230-204**
Feature Description, Issue 3

Provides comprehensive technical descriptions of features and parameters. Includes the applications and benefits, feature interactions, administration requirements, hardware and software requirements, and a brief discussion of data communications and private networking configurations.

DEFINITY Communications System Generic 3   **555-230-655**
Version 4 Implementation, Issue 2

Provides step-by-step procedures for preparing the hardcopy forms that correspond to the screens and are required to implement the features, functions, and services of the system. Includes procedures for completing a communications survey.

DEFINITY Communications System Generic 3   **555-230-655B**
Version 4 Implementation Blank Forms, Issue 2

Provides additional blank hardcopy forms that correspond to the screens that are required to implement the features, functions, and services of the system.

Copies of these forms are automatically included with the *DEFINITY Communications System Generic 3 Version 4 Implementation, Issue 2*, 555-230-655. Use this order number to purchase additional forms.

DEFINITY Communications System Generic 3 **555-230-206**
System Description and Specifications, Issue 4

Provides a technical description of the systems and is intended for service
personnel, sales personnel, and customers who need a comprehensive overview
of the system. Includes descriptions of hardware, software features, technical
specifications, environment requirements, maintenance requirements, and
illustration of components.

DEFINITY Communications System Generic 3 **555-230-511**
Version 4 Traffic Reports, Issue 2

Provides detailed descriptions of all the measurement, status, and security reports
available in the system and is intended for administrators who validate traffic
reports and evaluate system performance. This document was titled *System
Reports* for earlier systems. Includes corrective actions for potential problems.

DEFINITY Enterprise Communications Server **555-230-894 UK English**
Release 5 Installation and Test for Single-Carrier **555-230-895 German**
Cabinets, Issue 2 **555-230-896 French**
555-230-897 Spanish
555-230-900 Chinese

Provides procedures and information for hardware installation and initial testing of
the single-carrier cabinet systems only. The UK version will be shipped with all
single-carrier cabinet systems in the US. Some languages may not be available
until a future date.

DEFINITY Communications System Generic **555-204-105**
3i/s/vs Maintenance, Issue 8

Provide detailed descriptions of the procedures for monitoring, testing, and
maintaining the systems. Included are maintenance commands, step-by-step
trouble-clearing procedures, the procedures for using all tests, and explanations
of the system's error codes.

BCS Products Security Handbook, Issue 5 **555-025-600**

Provides information about the risks of telecommunications fraud and measures
for addressing those risks and preventing unauthorized use of BCS products. This
document is intended for telecommunications managers, console operators, and
security organizations within companies.

DEFINITY Communications System and System   **555-015-201**
75 and System 85 Terminals and Adjuncts
Reference, Issue 8

Provides descriptions of the peripheral equipment that can be used with the DEFINITY Business Communications System. This document is intended for customers and Lucent Technologies account teams for selecting the correct peripherals to accompany a system.

DEFINITY Communications System Generic 1   **555-230-701**
and Generic 3 Voice Terminal Operations, Issue 1

Provides detailed operating instructions for the system features on each type of voice terminal. Included are definitions of the voice features and user requirements.

DEFINITY Communications System Generic 1,   **555-230-755**
Generic 3, and System 75 Voice Terminal Guide
Builder, Issue 2

Provides capability to produce laser-printed documentation for specific voice terminals. The software is supported by a comprehensive user's guide and on-line help. This product requires a 386 PC, minimum of 6MB disk space, minimum of 4MB RAM, a printer supported by Microsoft GDI printer drive, and Microsoft Windows 3.1 or higher. A mouse is recommended.

DEFINITY Communications System Generic 2.2   **555-025-107**
and Generic 3 Version 2 DS1/CEPT1/ISDN PRI
Reference Manual, Issue 1

Provides a detailed technical description of digital trunks in the DEFINITY Communications Systems. This includes trunks conforming to the DS1 standard (1.544 Mbps) and the CEPT1 standard and all other methods of signaling, including bit-oriented signaling as well as ISDN-PRI signaling. This document includes background information on these topics, information on how digital trunk capabilities have been designed into the DEFINITY Communications System and information for field personnel and customers on how to provision and administer digital trunk capabilities and features. Provides both domestic and international information.

DEFINITY Communications System Generic 1   **555-230-700**
and Generic 3 Console Operation, Issue 2

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.

DEFINITY Communications System Console   **555-230-890 UK English**
Operations Quick Reference, Issue 1   **555-230-891 German**
                                                      555-230-892 French
                                                      555-230-893 Spanish
                                                      555-230-920 Chinese

Provides operating instructions for the attendant console. Included are descriptions of the console control keys and functions, call handling, basic system-troubleshooting information, and routine maintenance procedures. Some languages may not be available until a future date.

Abbreviations

A

AA
Archangel

AAR
Automatic Alternate Routing

AC
Alternating Current

ACA
Automatic Circuit Assurance

ACB
Automatic Callback

ACU
Automatic Call Unit

AD
Abbreviated Dialing

ADAP
AUDIX Data Acquisition Package

ADM
Asynchronous Data Module

ADU
Asynchronous Data Unit

AE
Access Endpoint

AIM
Asynchronous Interface Module

AIOD
Automatic Identification of Outward Dialing

ALM-ACK
Alarm Acknowledge

AMW
Automatic Message Waiting

AN
Analog

ANI
Automatic Number Identification

AOL
Attendant Offered Load

APLT
Advanced Private Line Termination

ARS
Automatic Route Selection

ASCII
American Standard Code for Information Interchange

ATB
All Trunks Busy

ATD
Attention Dial

ATMS
Automatic Transmission Measurement System

AUDIX
Audio Information Exchange

AUX
Auxiliary

AVD
Alternate Voice/Data

AWOH
Administration Without Hardware

B

BCC
Bearer-Capability Class

BCS
Business Communications Systems

BHCC
Busy Hour Call Completions

BLF
Busy Lamp Field

BN
Billing Number

BOS
Bit Oriented Signaling

BPN
Billed Party Number

BPS
Bits Per Second

BRI
Basic Rate Interface

BTU
British Thermal Unit

C

CA-TSC
Call-Associated Temporary Signaling Connection

CACR
Cancellation of Authorization Code Request

CAG
Coverage Answer Group

CAMA
Centralized Automatic Message Accounting

CARR-POW
Carrier Port and Power Unit for AC Powered Systems

CBC
Call-By-Call and Coupled Bonding Conductor

CC
Country Code

CCIS
Common Channel Interoffice Signaling

CCITT
Consultative Committee for International Telephone and Telegraph

CCMS
Common Channel Message Set

CCS
Centum (Hundred) Call Seconds

CCSA
Common Control Switching Arrangement

CDM
Channel Division Multiplexing

CDOS
Customer-Dialed and Operator Serviced

CDR
Call Detail Recording

CDRP
Call Detail Record Poller

CDRR
Call Detail Recording and Reporting

CDRU
Call Detail Recording Utilities

CEM
Channel Expansion Multiplexing

CEPT1
European Conference of Postal and Telecommunications Rate 1

CI
Clock Input

cm
Centimeter

CM
Connection Manager

CMDR
Centralized Message Detail Recording

CO
Central Office

COR
Class of Restriction

COS
Class of Service

CP
Circuit Pack

CPE
Customer Premises Equipment

CPN
Called-Party Number

CPN/BN
Calling Party Number/Billing Number

CPTR
Call Progress Tone Receiver

CRC
Cyclical Redundancy Checking

CSA
Canadian Safety Association

CSD
Customer Service Document

CSU
Channel Service Unit

CTS
Clear to Send

D

DAC
Dial Access Code

dB
Decibel

DC
Direct Current

DCE
Data Communications Equipment

DCP
Digital Communications Protocol

DDC
Direct Department Calling

DDD
Direct Distance Dialed

DID
Direct Inward Dialing

DIOD
Direct Inward and Outward Dialing

DIVA
Data In/Voice Answer

DLC
Data Line Circuit

DLDM
Data Line Data Module

DMI
Digital Multiplexed Interface

DNIS
Dialed Number Identification Service

DOD
Direct Outward Dialing

DOSS
Delivery Operations Support System

DPM
Dial Plan Manager

DPR
Dual Port RAM

DS1
Data Services Level 1

DS1C
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

DXS
Direct Extension Selection

E

E&M
Ear and Mouth (receive and transmit)

EAA
Expansion Archangel

EAL
Expansion Archangel Link

EBCDIC
Extended Binary-Coded Decimal Interexchange Code

ECC
Error Correct Code

EFP
Electronic Power Feed

EI
Expansion Interface

EIA
Electronic Industries Association

EMI
Electro-Magnetic Interference

EPROM
Erasable Programmable Read Only Memory

EPSCS
Enhanced Private Switched Communications Services

ESF
Extended Superframe Format

ETA
Extended Trunk Access

ETN
Electronic Tandem Network

ETSI
European Telecommunications Standards Institute

F

FAC
Feature Access Code

FAS
Facility-Associated Signaling

FAT
Facility Access Trunk

FAX
Facsimile

FCC
Federal Communications Commission

FEAC
Forced Entry of Account Codes

FEP
Front End Processor

FIC
Facility Interface Codes

FNPA
Foreign Numbering-Plan Area

FRL
Facilities Restriction Level

FX
Foreign Exchange

G

GM
Group Manager

GPTR
General-Purpose Tone Receiver

GRS
Generalized Route Selection

H

HNPA
Home Numbering Plan Area Code

Hz
Hertz

I

IAS
Inter-PBX Attendant Service

IC
Inter-Cabinet

ICC
Intercarrier Cable

ICDOS
International Customer Dialed Operator Serviced

ICHT
Incoming Call-Handling Table

ICI
Incoming Call Identifier

IDDD
International Direct Distance Dialing

IDF
Intermediate Distribution Frame

IE
Information Element

IMT
Intermachine Trunk

in
Inch

INADS
Initialization and Administration System

INS
ISDN Network Service

INWATS
Inward Wide Area Telephone Service

IO
Information Outlet

ISDN
Integrated Services Digital Network

ISN
Information Systems Network

ISO
International Standards Organization

ISV
Independent Software Vendor

ITP
Installation Test Procedures

ITU
International Telecommunications Union

IXC
Interexchange Carrier Code

K

kHz
Kilohertz

kbps
Kilobits Per Second

kbyte
Kilobyte

kg
Kilogram

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

lb
Pound

LDN
Listed Directory Number

LDS
Long-Distance Service

LEC
Local Exchange Carrier

LED
Light-Emitting Diode

LINL
Local Indirect Neighbor Link

LMM
Low Level Maintenance Monitor

LSU
Local Storage Unit

LWC
Leave Word Calling

M

M-Bus
Memory Bus

MA-UUI
Message Associated User-to-User Signaling

MADU
Modular Asynchronous Data Unit

MAP
Maintenance Action Process

Mbps
Megabits Per Second

Mbyte
Megabytes

MDF
Main Distribution Frame

MDM
Modular Data Module

MDR
Message Detail Record

MEM
Memory

MET
Multibutton Electronic Telephone

MFC
Multifrequency Compelled Signaling

MHz
Megahertz

MIM
Management Information Message

MIS
Management Information System

MISCID
Miscellaneous Identification

MMS
Material Management Services

MOS
Message-Oriented Signaling

MPDM
Modular Processor Data Module

MS
Message Server

ms
Millisecond

MSA
Message Servicing Adjunct

MSG
Message Service

MSM
Modular System Management

MSS
Mass Storage System

MSSNET
Mass Storage/Network Control

MT
Management Terminal

MTDM
Modular Trunk Data Module

MTP
Maintenance Tape Processor

MTT
Multi-Tasking Terminal

MWL
Message Waiting Lamp

N

NANP
North American Numbering Plan

NAU
Network Access Unit

NCA/TSC
Non-Call Associate/Temporary Signaling Connection

NCOSS
Network Control Operations Support Center

NCSO
National Customer Support Organization

NEC
National Engineering Center

NEMA
National Electrical Manufacturer's Association

NFAS
Non-Facility Associated Signaling

NID
Network Inward Dialing

NM
Network Management

NN
National Number

NPA
Numbering Plan Area

NPE
Network Processing Element

NQC
Number of Queued Calls

NSE
Night Service Extension

NSU
Network Sharing Unit

NXX
Public Network Office Code

O

OA
Operator Assisted

ONS
On-Premises Station

OPS
Off-Premises Station

OQT
Oldest Queued Time

OSHA
Occupational Safety and Health Act

OSI
Open Systems Interconnect

OSS
Operations Support System

OSSI
Operations Support System Interface

OTQ
Outgoing Trunk Queuing

P

PACCON
Packet Control

PAD
Packet Assembly/Disassembly

PC
Personal Computer

PCM
Pulse Code Modulated

PCOL
Personal Central Office Line

PCOLG
Personal Central Office Line Group

PCS
Permanent Switched Calls

PDM
Processor Data Module

PDS
Premises Distribution System

PE
Processing Element

PEC
Price Element Codes

PEI
Processor Element Interchange

PGATE
Packet Gateway

PI
Processor Interface

PIB
Processor Interface Board

PIDB
Product Image Database

PKTINT
Packet Interface

PL
Private Line

PLS
Premises Lightwave System

PN
Port Network

PNA
Private Network Access

POP
Point Of Presence

PPN
Processor Port Network

PRI
Primary Rate Interface

PROCR
Processor

PSC
Premises Service Consultant

PSDN
Packet Switch Public Data Network

PT
Personal Terminal

PTC
Positive Temperature Coefficient

PTT
Postal Telephone and Telegraph

R

R2-MFC
R2 Multifrequency Compelled Signaling

RAM
Random Access Memory

RBS
Robbed-Bit Signaling

RCL
Restricted Call List

RHNPA
Remote Home Numbering Plan Area

RINL
Remote Indirect Neighbor Link

RLT
Release Link Trunk

RMATS
Remote Maintenance, Administration, and Traffic System

RNX
Route Number Index (Private Network Office Code)

ROM
Read-Only Memory

RPN
Routing Plan Number

RS232C
Recommended Standard 232C

RS449
Recommended Standard 449

RSC
Regional Support Center

S

SABM
Set Asynchronous Balance Mode

SAKI
Sanity and Control Interface

SCC
Single Carrier Cabinet

SCD
Switch-Control Driver

SCI
Switch Communications Interface

SCO
System Control Office

SCOTCH
Switch Conferencing for TDM Bus in Concentration Highway

SCSI
Small Computer System Interface

SDDN
Software Defined Data Network

SDI
Switched Digital International

SDLC
Synchronous Data Link Control

SDN
Software Defined Network

SID
Station Identification Number

SIT
Special Information Tones

SMDR
Station Message Detail Recording

SMM
Standby Maintenance Monitor

SN
Switch Node

SNA
Systems Network Architecture

SNC
Switch Node Clock

SNI
Switch Node Interface

SPE
Switch Processing Element

SPID
Service Profile Identifier

SSI
Standard Serial Interface

SSM
Single Site Management

SSV
Station Service

ST3
Stratum 3 Clock Board

STARLAN
Star-Based Local Area Network

SVN
Security Violation Notification

SXS
Step-by-Step

SYSAM
System Access and Administration

T

TAAS
Trunk Answer from Any Station

TAC
Trunk Access Code

TC
Technical Consultant

TCM
Traveling Class Mark

TDM
Time-Division Multiplex(ing)

TDR
Time-of-Day Routing

TEG
Terminating Extension Group

TEI
Terminal Endpoint Identifier

TOD
Time-of-Day

TOP
Task Oriented Protocol

TSO
Technical Support Organization

TTR
Touch-Tone Receiver

TTT
Terminating Trunk Transmission

TTTN
Tandem Tie Trunk Network

TTY
Teletypewriter

U

UAP
Usage Allocation Plan

UART
Universal Asynchronous Transmitter

UCD
Uniform Call Distribution

UCL
Unrestricted Call List

UDP
Uniform Dial Plan

UL
Underwriter Laboratories

UM
User Manager

UNMA
Unified Network Management Architecture

UNP
Uniform Numbering Plan

UPS
Uninterruptible Power Supply

USOP
User Service Order Profile

UUCP
UNIX-to-UNIX Communications Protocol

UUI
User-to-user information

V

VIS
Voice Information System

VLSI
Very Large Scale Integration

VM
Voltmeter

VNI
Virtual Nodepoint Identifier

W

WATS
Wide Area Telecommunications Service

WCC
World Class Core

WSA
Waiting Session Accept

Z

ZCS
Zero Code Suppression

Glossary

Numerics

800 service
    A service in the USA, which allows incoming calls from a certain area or areas to an assigned number for a flat-rate charge based on usage.

A

ARS
    See Automatic Route Selection.

abandoned call
    An incoming call, where the caller hangs up before being answered.

access code
    A one-, two-, or three-digit dial code used to activate or cancel a feature, or access an outgoing trunk. The star (*) and pound (#) can be used as the first digit of an access code.

access 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 USA.

adjunct
    A processor that does one or more tasks for another processor and that is optional in the configuration of the other processor.

administer
    To access and change parameters associated with the services or features of a system.

Administered Connection (AC)
    Administered Connection is a feature that allows the system to automatically establish and maintain end-to-end connections between access endpoints (trunks) and/or data endpoints (data modules).

Administration Without Hardware (AWOH)
    Provides the ability to administer ports without the need for the associated terminals or other hardware to be physically present.

abandoned call
    An incoming call, where the caller hangs up before being answered.

American National Standard Code for Information Interchange
See **ASCII**.

analog
The representation of information by means of continuously variable physical quantities such as amplitude, frequency, and phase.

analog data
Data that is transmitted over a digital facility in analog (pulse code modulation) form. The data must pass through a modem at both ends.

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**.

angel
A microprocessor located on each port card in a processor port network (PPN). The angel uses the control-channel message set (CCMS) to manage communications between the port card and the archangel on the controlling switch processing element (SPE). The angel also monitors the status of other microprocessors on a port card and maintains error counters and thresholds. See also **archangel**.

answerback code
An assigned 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. Also called "call appearance," "line appearance," and "occurrence."

architecture
The organizational structure of a system, including hardware and/or software.

ASCII (American Standard Code for Information Interchange)
The standard code, using a coded character set consisting of 7-bit coded characters (eight bits, including parity check), used for information interchange among data processing systems, data communications systems, and associated equipment. The ASCII set consists of control characters and graphic characters.

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 transmission is advantageous when transmission is not regular (characters typed at a keyboard). Also called ''asynchronous transmission.'' See also **synchronous data transmission**.

association
An association is a communication channel between the adjunct and system for messaging purposes. An active association is one which applies to an existing call on the system or to an extension on the call.

asynchronous data unit (ADU)
A data communications equipment (DCE) type device that allows direct connection between RS232C equipment and a digital system.

attendant
A person at a console on a customer's premises who provides 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 multimedia messaging system that 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 groups
Those trunk groups where the CO processes all of the digits for the incoming call. Whenever the system determines that the CO has seized a trunk from an Auto-In trunk group, it automatically (without processing any digits) connects the trunk to the destination.

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

automatic trunk
A trunk that does not require the sending or receiving of 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."

automatic restoration
A service that restores disrupted connections between access endpoints (nonsignaling trunks) and data endpoints (devices that connect the system to data terminal and/or communications equipment). This restoration is done within seconds of a service disruption so that critical data applications can remain operational.

auxiliary equipment
Equipment used for optional 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.

B

bandwidth
The difference, expressed in Hertz, between the defined highest and lowest frequencies in a frequency range.

baud
In telecommunications applications, a unit of transmission speed equal to the number of signal events per second. See also **bit rate** and **bits per second**.

BCC

The Bearer-Capability Class (BCC) identifies the type of call, for example, voice and different types of data. Determination of BCC is based on the call originator's characteristics for non-ISDN endpoints and on the Bearer-Capability and Low-Layer Compatibility Information Elements of an ISDN endpoint.

Current BCCs are as follows:

- 0: Voice-grade data and voice

- 1: DMI Mode 1, 56 kbps data transmission

- 2: DMI Mode 2, synchronous/asynchronous data transmission up to 19.2 kbps

- 3: DMI Mode 3, 64 kbps circuit/packet data transmission

- 4: DMI Mode 0, 64 kbps synchronous data

- 5: Temporary Signaling Connection

bit (binary digit)

One unit of information in binary notation having two possible states or 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**.

bridge (bridging)

The appearance of a voice terminal's extension at one or more other voice terminals.

BRI

The ISDN Basic Rate Interface specification.

bridged appearance

A call appearance on a voice terminal that matches a call appearance on another voice terminal for the duration of a call.

buffer

(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 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.

BX.25

A Lucent Technologies 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 International Standards Organization (ISO) reference model.

bypass tie trunks

A one-way, outgoing tie trunk from a tandem system to a main system in an electronic tandem network (ETN). Bypass tie trunks, provided in limited quantities, are used as a "last-choice" route when all trunks to another tandem system 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.

cable

The physical connection between two pieces of equipment; for example, cable from a data terminal to a modem, cable between a piece of equipment and a termination field, or circuit pack I/O cables.

cable connector

A cable connector is either a jack (female) or plug (male) on the end of a cable. A cable connector connects wires on a cable to specific leads on telephone or data equipment.

call appearance, attendant console

Six buttons, labeled "a" through "f," and used to originate, receive, and hold calls. Each button has two lights to show the status of the call appearance.

call appearance, voice terminal

A button labeled with an extension number 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 or the status of the call.

call control capabilities

call control capabilities are all the capabilities (*Third Party Selective Hold, Third Party Reconnect, Third Party Merge*) that can be used in either of the Third Party Call Control ASE (cluster) subsets: Call Control and Domain Control.

call detail recording

A feature that utilizes software and hardware to record call data (same as CDRU).

call detail recording utility (CDRU)

Applications software that collects, stores, optionally filters, and outputs call detail records for direct or polled output to peripheral devices.

call reference value (CRV)

An identifier present in ISDN messages that serves to associate a related sequence of messages.

callback call

A call that is automatically returned to a voice terminal user who activated the Automatic Callback or Ringback Queuing feature.

call-waiting ringback tone

A low-pitched tone identical to ringback tone except that the tone decreases in the last 0.2-second (in the United States). A call-waiting ringback tone notifies the attendant that the Attendant Call Waiting feature has been activated and that the called user is aware of the waiting call. Tones in international countries may sound different.

carrier

An enclosed shelf containing vertical slots that hold circuit packs.

carried load

The amount of traffic actually served by traffic-sensitive facilities during a given interval.

CCS or hundred call seconds

A unit of traffic measure that is used to determine usage. In order to determine usage for a facility, it is scanned every 100 seconds. If the facility is found busy, then 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 as 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 capability is a either a request or indication of an operation. For example, a *Third Party Make Call* is a request for setting-up a call and an *Event Report* is an indication that an event has occurred.

capability groups

Capability groups are sets of capabilities, provisioned through system administration, that can be requested by an application. Each capability group may contain capabilities from several capability groups. Capability groups are also referred to, in other documentation, as administration groups or Application Service Elements (ASEs). Capability groups denote association types. For example, *Call Control* is a type of association which allows certain functions (the ones in the capability group) to be performed over this type of association.

CCITT

CCITT (Comitte Consultatif International Telephonique et Telegraphique) is now called *International Telecommunications Union* (ITU). See this name for information.

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 USA. CO codes are numbered from 200 through 999.

central office (CO) trunk

A telecommunications channel that provides access from the system to the public network through the local CO.

channel

The term channel is nonspecific and must be taken in context. Channel can refer to a circuit-switched call or a communications path for transmitting voice and/or data.

Channel can refer to a T1 or E1 facility not specifically associated with a logical circuit-switched call. In this context, a channel is analogous to a single trunk.

channel negotiation

Channel negotiation is 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 system receiving the SETUP message and ultimately to the system that sent the SETUP. Negotiation will be attempted only if the CIIE is encoded as *Preferred*.

circuit

(1) An arrangement of electrical elements through which electric current flows, providing one or more specific functions. (2) A channel or transmission path between two or more points.

circuit pack

A card on which electrical circuits are printed, and integrated circuit (IC) chips and electrical components are installed. A circuit pack is installed in a system carrier.

Class of Restriction (COR)
A feature that allows up to 64 classes of call-origination and call-termination restrictions for voice terminals, voice terminal groups, data modules, and trunk groups. See also **Class of Service (COS)**.

Class of Service (COS)
A feature that uses a number (0 through 15) to specify if voice terminal users can activate the Automatic Callback, Call Forwarding—All Calls, Data Privacy, or Priority Calling features.

common control switching arrangement (CCSA)
A private telecommunications network using dedicated trunks and a shared switching center for interconnecting company locations.

communications system
The software-controlled processor complex that interprets dialing pulses, tones, and/or keyboard characters and makes the proper interconnections 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 actual connections. A communications system provides voice and/or data communications services, including access to public and private networks, for telephones and data terminals on a customer's premises. See also **system**.

confirmation tone
A tone confirming that a feature activation, deactivation, or cancellation has been accepted.

connectivity
The connection of disparate devices within a single system.

console
See **attendant console**.

control cabinet
A carrier in a single-carrier cabinet that contains the switch processing element (SPE) circuit packs and port circuit packs. Also called "control cabinet." See also **switch processing element**.

controlled station
A station that is being monitored and controlled via a domain-control association.

coverage answer group
A group of up to eight voice terminals that ring simultaneously when a call is redirected to it by Call Coverage. Any one of the group can answer the call.

coverage call
A call that is automatically redirected from the called party's extension number to an alternate answering position when certain coverage criteria are met.

coverage path
The order in which calls are redirected to alternate answering positions.

coverage point
An extension, attendant group, or Direct Access Calling number designated as an alternate answering position in a coverage path.

covering user
A person at a coverage point who answers a redirected call.

D

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 provides the functions to make 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 basic rate interface (BRI) or digital communications protocol (DCP) interface of the system and data terminal equipment (DTE) or data communications equipment (DCE).

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 designed to transmit digital data on transmission facilities.

data terminal

An input/output (I/O) device that has either switched or direct access to a host computer.

data terminal equipment (DTE)

Equipment consisting of the endpoints in a connection over a data circuit. For example, in a connection between a data terminal and a host, the terminal, the host, and their associated modems or data modules make up the DTE. DTE usually consists of the following functional units: control logic, buffer store, and one or more input or output devices or computers. DTE can contain error control, synchronization, and telephone-identification capabilities.

D-channel backup

D-channel backup is used with Non-Facility Associated Signaling (NFAS). With D-channel backup, a primary D-channel provides signaling for an NFAS D-channel group (two or more PRIs facilities). A second (redundant) D-channel, located on a separate PRI facility of the NFAS D-channel group is designated as backup for the D-channel. The failure of the primary D-channel causes an automatic transfer of call-control signaling to the backup D-channel. When this happens, the backup becomes the primary D-channel, and when the previous primary is returned to service it becomes the backup D-channel.

delay-dial trunk

A trunk that allows dialing directly into a communications system — that is, the digits are received as they are dialed.

denying a request

Denying a Request is the same as sending a negative acknowledgement (NAK), and is done by sending an Facility Information Element (FIE) with a *return error* component (a cause value is also provided). It should not be confused with the ''denial'' event report which applies to calls.

designated voice terminal
> The specific voice terminal to which calls, originally directed to a certain extension number, are redirected. Commonly used to mean the "forwarded-to" terminal when Call Forwarding All Calls is active.

dial-repeating tie trunk
> A tie trunk that transmits called-party addressing information between two communications systems.

dial-repeating trunks
> A system tie trunk that is capable of handling system station signaling information without attendant assistance.

digit conversion
> A process used to convert specific dialed numbers into other dialed numbers.

digital communications protocol (DCP)
> A Lucent Technologies 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 multiplexed interface (DMI)
> An interface that provides connectivity between a communications system and a host computer or between two communications systems using digital signal level-1 (DS1) 24th-channel signaling. DMI provides 23 64-kbps data channels and 1 common signaling channel over a twisted- pair connection. DMI is offered through two capabilities: bit-oriented signaling (DMI-BOS) and message-oriented signaling (DMI-MOS).

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 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 processor data module (PDM) and modular processor data module (MPDM) in that it converts RS232C signals to DCP signals.

digital-to-analog converter
> A device that converts data in digital form to the corresponding analog signals. See also **analog-to-digital converter**.

digital transmission
> A mode of transmission in which the information to be transmitted is first converted to digital form and then transmitted as a serial stream of pulses.

digital trunk
> A circuit in that carries digital voice and/or digital data in a telecommunications channel.

Direct Access Calling number
> An extension that provides access to the Direct Access Calling feature on the system. Direct Access Calling allows a customer to specify the treatment of incoming calls based on the dialed number.

Direct Access Calling procedure
> A set of up to 15 commands to be performed for an incoming or internal call.

Direct Extension Selection (DXS)
A feature on an attendant console that allows an attendant direct access to voice terminals 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. DID calls to DID-restricted telephone lines are routed to an attendant or recorded announcement, depending on the option selected.

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.

disk drive
An electromechanical device that stores data on and retrieves data from one or more disks.

domain
Available domains are Direct Access Calling numbers and stations. The Direct Access Calling number domain is used only for active-notification associations, the station domain is only used for the domain-control associations.

domain-control association
A *Third Party Domain Control Request* capability initiates a unique "CRV/link number" combination, which is referred to as a domain-control association.

domain-controlled split
A split for which *Third Party Domain Control* request has been accepted. A domain-controlled split provides an event report for logout.

domain-controlled station
A station for which a *Third_Party_Domain_Control* request has been accepted. A domain-controlled station provides event reports for calls that are alerting, connected, or held at the station.

domain-controlled station on a call
A station active on a call which provides event reports over one or two domain-control associations.

E

E1
A digital transmission standard that carries traffic at the rate of 2.048 Mbps.

The E1 facility is divided into 32 channels (DS0s) of 64 kbps information numbered from 0 to 31. Channel 0 is reserved for framing and synchronization information. When a D-channel is present, it occupies channel 16.

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.

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 at the time the call is placed. Each system in the network is assigned a unique private network office code (RNX), and each voice terminal is assigned a unique extension number.

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, the automatic transfer within a communications system of a pre-defined set of central office (CO) lines to a group of answering telephones with at least one tele-phone 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."

end-to-end signaling
The transmission of touch-tone signals generated by dialing from a voice terminal user to remote computer equipment. A connection must first be established over an outgoing trunk from the call-ing party to the computer equipment. Then additional digits can be dialed to transmit information to be processed by the computer equipment.

enhanced private-switched communications service (EPSCS)
An analog private telecommunications network based on the No. 5 Crossbar and 1A ESS that pro-vides advanced voice and data telecommunications services to companies with many locations.

Erlang
A unit of traffic intensity, or load, used to express the amount of traffic it takes to keep one facility busy for one hour. One Erlang is equal to 36 CCS. See also **Hundred Call Seconds**.

extension number
A three- to five-digit number by which calls are routed through a communications system or, with a Uniform Dial Plan (UDP), through a private network. Extension numbers are primarily used for tele-phones and data terminals but can also be used with specific features.

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

facility
A general term used for a telecommunications transmission pathway and associated equipment.

facility associated signaling (FAS)
Signaling in which a D-channel carries the signaling only for those channels on the same physical interface.

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.

foreign exchange (FX)
A central office (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 central office (CO) other than its local CO.

foreign numbering-plan area code (FNPAC)
An area code other than the local area code. The FNPAC must be dialed to call outside the local geographical area.

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 two-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.

H

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.

holding time
The total length of time in minutes and seconds that a facility is used during a call.

home numbering-plan area code
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 (SCIs) whereby the SCI message passes automatically without intermediate processing through one or more intermediate SCIs.

host computer
A computer, connected to a network, that processes data from data-entry devices.

hunt group
A group of extensions that are assigned the Station Hunting feature so that a call to a busy extension will reroute to an idle extension in the group.

I

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.

information exchange
The exchange of data between users of two different systems, such as the system and a host computer, over a local area network (LAN).

information systems network (ISN)
A wide area network (WAN) and local area network (LAN) with an open architecture combining host computers, minicomputers, word processors, storage devices, PCs, high-speed printers, and nonintelligent terminals into a single packet-switching system.

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** and **Integrated Services Digital Network Primary Rate Interface**.

Integrated Services Digital Network Basic Rate Interface (ISDN-BRI)
The interface between a communications system and a 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 — an arrangement called "2B+1D." 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 terminals and digital terminals fitted with ISDN terminal adapters. See also **Integrated Services Digital Network Primary Rate Interface**.

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 Mbytes per second.

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 — an arrangement called "23B+1D," although with non- facil-
ity-associated signaling (NFAS) ISDN-PRI can include 24 B-channels and no D-channel. See also
Integrated Services Digital Network and **Integrated Services Digital Network Basic Rate
Interface**.

intercept tone
An tone that indicates a dialing error or denial of the service requested.

interface
A common boundary between two systems or pieces of equipment.

internal call
A connection between two users within a system.

International Tele-communications 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
Integrated Services Digital Network (ISDN). ITU members are from telecommunications compa-
nies and organizations around the world. See also **BX.25**.

International Telegraph and Telephone Consultative Committee
See **International Telecommunications Union (ITU)**.

in-use lamp
A red light on a multiappearance voice terminal that is illuminated 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.

ISDN Gateway (IG)
A feature allowing integration of the system and a host-based telemarketing application via a link to
a gateway adjunct. The gateway adjunct is a 3B-based product that notifies the host-based tele-
marketing application of call events.

ISDN trunk
A trunk administered for use with Integrated Services Digital Network primary rate interface
(ISDN-PRI). Also called "ISDN facility."

ISDN-PRI Terminal Adapter
A terminal adapter acts as 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.

L

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, and 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 digital signal level-1 (DS1) converter circuit packs. Lightwave transceivers convert electrical signals to light signals and vice versa.

line

A transmission path between a communications system or central office (CO) switching system and a voice terminal or other terminal.

line port

The hardware that provides the access point to a communications system for each circuit associated with a telephone and/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 Integrated Services Digital Network basic rate interface (ISDN-BRI) and primary rate interface (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 basic rate interface (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.

M

maintenance

The 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.

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.

memory

A device into which information can be copied and held, and from which the information can be obtained at a later time.

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.

modem

A device that converts digital data signals to analog signals for transmission over telephone circuits. The analog signals are converted back to the original digital data signals by another modem at the other end of the circuit.

modular processor data module (MPDM)

A processor data module (PDM) that can be configured to provide several kinds of interfaces (RS232C, RS449, and V.35) to customer-provided data terminal equipment (DTE). See also **processor data module**.

modular trunk data module (MTDM)

A trunk data module (TDM) that can be configured to provide several kinds of interfaces (RS232C, RS449, and V.35) to customer-provided data terminal equipment (DTE).

modulator-demodulator

See **modem**.

multiappearance voice terminal

A terminal equipped with several call appearance buttons for the same extension number, allowing the user to handle more than one call, on that same extension number, 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 system, another signal acknowledging the transmitted signal is received by the system. "R2" designates signaling used in the USA and countries outside the USA.

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**.

multirate

Multirate refers to the new N x DS0 service (see N x DS0).

N

narrowband

A circuit-switched call at a data rate up to and including 64 kbps.

Non-Facility Associated Signaling (NFAS)

A method that allows multiple T1 and/or E1 facilities to share a single D-channel to form an Integrated Services Digital Network primary rate interface (ISDN PRI). If D-Channel Backup is not used, one facility is configured with a D-channel, while 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), while the other facilities that share the D-channels are configured without D-channels.

On every facility, all DS0s that are not D-channels are available as B-channels. Therefore, a T1 facility without a D-channel has 24 B-channels, and an E1 facility without a D-channel has 31 B-channels.

network

A series of points, nodes, or stations connected by communications channels.

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.

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.

O

offered load

The traffic that would be generated by all the requests for service occurring within a monitored interval, usually one hour.

P

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 it 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.

PCOL

Personal Central Office Line.

primary extension

The main extension associated with the physical station set.

principal

A station that has its primary extension bridged on one or more other stations.

personal computer (PC)

A personally controllable microcomputer.

pickup group

A group of individuals authorized to answer any call directed to an extension number within the group.

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, service circuit packs, and (optionally) up to four expansion interface (EI) circuit packs. Each PN is controlled either locally or remotely by a switch processing element (SPE). See also **processor port network**.

Primary Rate Interface (PRI)

A standard Integrated Services Digital Network (ISDN) frame format that specifies a protocol used on digital circuits between two communications systems.

In North America, PRI runs at 1.544 Mbps and provides 23 64 kbps B-channels (voice or data) and one 64 kbps D-channel (signaling). In most countries outside of North America, PRI runs at 2.048 Mbps and provides 30 64 kbps B-channels (voice or data) and one 64 kbps D-channel (signaling.) The D-channel of the interface contains multiplexed signaling information for the other channels.

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. These codes are numbered 220 through 999, excluding any codes that have a 0 or 1 as the second digit.

processor carrier

See **control carrier**.

processor data module (PDM)

A device that provides an RS232C data communications equipment (DCE) interface for connecting to data terminals and host computers and provides a digital communications protocol (DCP) interface for connection to a communications system. See also **modular processor data module**.

processor port network (PPN)

A port network (PN) controlled by a switch processing element (SPE) that is directly connected to that PN's time-division multiplex (TDM) bus and local area network (LAN) bus. See also **port network**.

processor port network (PPN) control carrier

A carrier containing the maintenance circuit pack, tone/clock circuit pack, and switch processing element (SPE) circuit packs for a processor port network (PPN) and, optionally, port circuit packs.

protocol

A set of conventions or rules governing the format and timing of message exchanges to control data movement and correction of errors.

public network

The network that can be openly accessed by all customers for local or long-distance calling.

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

quadrant

A quadrant is 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).

A quadrant is considered available or idle when all six contiguous DS0s are idle. Otherwise, the quadrant is considered contaminated or partially contaminated. This is a dynamic condition; quadrants become idle and contaminated as calls are placed and dropped. Note that a T1 facility containing the primary or backup D-channel (23B + D) has a maximum of three idle quadrants. The fourth quadrant (DS0s 19-24) never has six contiguous idle DS0s because one is always allocated to the D-channel. On an E1 facility, channel 0 is reserved for framing and synchronization, and channel 16 contains the D-channel when present, but five quadrants are potentially available.

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

R2-MFC signaling

MultiFrequency compelled (MFC) signaling is a form of number signaling similar to Dual-Tone MultiFrequency (DTMF) in that tones convey the dialed number. R2-MFC is a version of MFC recommended by CCITT for signaling between a CO and a system over analog or digital CO, DID, or tie trunks.

random access memory (RAM)

A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

read-only memory (ROM)

A storage arrangement primarily for information retrieval applications.

recall dial tone

Tones signaling that the system has completed a function (such as holding a call) and is ready to accept dialing.

redirection criteria

The information administered for each voice terminal's coverage path that determines when an incoming call is redirected to coverage.

remote home numbering-plan area code (RHNPA)

A foreign numbering-plan area code that is treated as a home area code by the Automatic Route Selection (ARS) feature. Calls can be allowed or denied based on the area code and the dialed central office (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.

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 at the time the call was placed.

report scheduler
Software that is used in conjunction with the system printer for the purpose of scheduling the days of the week and the time of the day that the desired reports are to be printed.

RS232C
A physical interface specified by the EIA. RS232C transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet.

ROSE
Remote Operations Service Element is a CCITT and ISO standard that defines a notation and services that support interactions between the various entities that make up a distributed application.

S

sanity and control interface (SAKI)
A custom, very-large-scale-integration (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.

simplex system
A system that has no redundant hardware.

simulated bridged appearance
The same as a **temporary bridged appearance**, allows the station user (usually the principal) the ability to bridge onto a call which had been answered by another party on its behalf.

single-carrier cabinet
A combined cabinet and carrier unit that contains one carrier.

single-line voice terminal
A voice terminal served by a single-line tip and ring circuit (models 500, 2500, 7101A, 7103A).

small computer system interface (SCSI)
An ANSI bus standard that provides a high-level command interface between host computers and peripheral devices.

software
A set of computer programs that perform one or more tasks.

split
A condition whereby a caller is temporarily separated from a connection with an attendant. A split condition automatically occurs when the attendant, active on a call, presses the start button.

Station Message Detail Recording (SMDR)
An obsolete term now called "CDR" (see call detail recording), which is a switch feature that utilizes software and hardware to record call data.

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).

switch

Any kind of telephone switching system. See also **communications system** and **system**.

switchhook

The buttons located under the receiver on a voice terminal.

switch processing element (SPE)

A complex of circuit packs — processor, memory, disk controller, and bus-interface cards — mounted in a processor-port-network (PPN) control carrier. The SPE serves as the control element for that PPN and, optionally, for one or more expansion port networks (EPNs).

synchronous data transmission

A method of sending data in which discrete signal elements are sent at a fixed and continuous rate and specified times.

system

Any kind of telephone switching system. See also **communications system** and **switch**.

system administrator

The person who maintains overall customer responsibility for system administration. Generally, all administration functions are performed from a PC using the TERRANOVA terminal emulator software. The system requires a special login, referred to as the system administrator login, in order to gain access to the system administration capabilities.

system printer

An optional printer that may be used to print scheduled reports via the report scheduler.

system report

Provides historical traffic information for all internally measured splits.

system status report

Provide real-time status information for internally measured splits.

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 digital signal level-1 (DS1) rate of 1.544 Mbps. A T1 facility is divided into 24 channels (DS0s) of 64 kbps information numbered from 1 to 24. 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.

TAC

Trunk Access Code.

tandem switch
A system 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
A private network that interconnects several switching systems.

TEG
Terminating Extension Group.

terminal
A device that sends and receives data within a system. See also **administration terminal**.

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 system, 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 report(s).

time slice
See **time interval**.

time slot
A time slot refers to 64 kbps of digital information structured as eight bits every 125 microseconds. In the system, 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.

to control
To control means that an application can invoke Third Party Call Control capabilities using either an adjunct-control or a domain-control association.

to monitor
To monitor means that an application can receive *Event_Reports* on either an active-notification, adjunct-control, or a domain-control association.

tone ringer
A device with a speaker, used in electronic voice terminals to alert the user.

trunk
A dedicated telecommunications channel between two communications systems or central offices (COs).

trunk group
Telecommunications channels assigned as a group for certain functions that can be used interchangeably between two communications systems or central offices (COs).

U

uniform dial plan
A feature that allows a unique four- or five-digit number assignment for each terminal in a multisystem configuration.

V

voice terminal
A single-line or multiappearance telephone.

W

wide area tele-communications service (WATS)
A service in the USA that allows calls to a certain area or 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.

write operation
The process of putting information onto a storage medium, such as a hard disk.

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