



Overview

for

**Avaya IP600 Internet Protocol  
Communications Server**

555-233-001  
Issue 5  
November 2000

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#### Preventing Toll Fraud

"Toll fraud" is the unauthorized use of your telecommunications system by an unauthorized party (for example, a person who is not a corporate employee, agent, subcontractor, or working on your company's behalf). Be aware that there may be a risk of toll fraud associated with your system and that, if toll fraud occurs, it can result in substantial additional charges for your telecommunications services.

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Your company's "telecommunications equipment" includes both this Avaya product and any other voice/data/video equipment that could be accessed via this Avaya product (that is, "networked equipment").

An "outside party" is anyone who is not a corporate employee, agent, subcontractor, or working on your company's behalf. Whereas, a "malicious party" is anyone (including someone who may be otherwise authorized) who accesses your telecommunications equipment with either malicious or mischievous intent.

Such intrusions may be either to/through synchronous (time-multiplexed and/or circuit-based) or asynchronous (character-, message-, or packet-based) equipment or interfaces for reasons of:

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- Theft (such as, of intellectual property, financial assets, or toll-facility access)
- Eavesdropping (privacy invasions to humans)
- Mischief (troubling, but apparently innocuous, tampering)
- Harm (such as harmful tampering, data loss or alteration, regardless of motive or intent)

Be aware that there may be a risk of unauthorized intrusions associated with your system and/or its networked equipment. Also realize that, if such an intrusion should occur, it could result in a variety of losses to your company (including, but not limited to, human/data privacy, intellectual property, material assets, financial resources, labor costs, and/or legal costs).

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- System administration documents
- Security documents
- Hardware-/software-based security tools
- Shared information between you and your peers
- Telecommunications security experts

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- Avaya-provided telecommunications systems and their interfaces
- Avaya-provided software applications, as well as their underlying hardware/software platforms and interfaces
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# About This Document

## Purpose

This document provides a high-level overview of the features, components, and capabilities of the Avaya IP600 Internet Protocol Communications Server. The Avaya IP600 is based on DEFINITY systems architecture and software. For information on how to install, administer, or maintain the Avaya IP600, see the DEFINITY ONE Documents Main Menu on this CD.

This document is intended to provide an understanding of:

- Avaya IP600 components and features
- Additional solutions to further tailor Avaya IP600 to future needs

## Intended audiences

This overview provides information for the following audiences:

- Customer end users and system administrators
- Avaya account executives, representatives, and distributors who require high-level information about the system and its use

## How to use this document

This document provides a basic understanding of the components, features, and capabilities of the Avaya IP600. The information is useful in identifying applications to increase employee productivity and effectiveness.

Overview chapters are:

- [Chapter 1, Introduction](#) provides an overview of Avaya IP600, including features, hardware, and software.
- [Chapter 2, Desktop/Console Solutions](#) describes the telephones and consoles available with Avaya IP600.
- [Chapter 3, Adjuncts](#) describes the adjuncts available with Avaya IP600.
- [Chapter 4, Intuity AUDIX Messaging](#) describes the AUDIX application and features on Avaya IP600 and provides a high-level overview of application capabilities and functionality.
- [Chapter 5, Call Center](#) describes advanced call-handling applications and call center management capabilities.
- [Chapter 6, Wireless Solutions](#) describes applications that enable employees to stay in touch with co-workers and clients from both on-site and off-site locations.

- [Chapter 7, Computer Telephony Integration](#) describes the applications that enable employees to combine computer and telephone functions to access client information.
- [Chapter 8, Enterprise Class IP Solutions](#) describes the capabilities and applications that support audio/voice over a LAN or WAN.
- [Chapter 9, Telecommuting/Virtual Office](#) describes applications that enable employees to work effectively off-site.
- [Chapter 10, System Administration](#) describes applications to help manage Avaya IP600, including the DEFINITY Site Administration (DSA) tool.
- [Chapter 11, Networking](#) describes connection applications for various voice and data networks.
- [Chapter 12, SNMP Native Agent Software](#) describes the SNMP interface to the system's alarm and error tables, performance measurements, and configuration data.
- [Appendix A, Avaya IP600 Features](#) lists the features of Avaya IP600.

## Conventions used in this document

The following conventions are used in this document:

- The term *system* is used to represent Avaya IP600.
- The term *switch* is used to represent other telecommunications switching products.

## Security

Avaya IP600 security is extremely important to Avaya. See the [BCS Products Security Handbook](#) (555-025-600) and the Avaya IP600 documentation for security measures for your system.

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- CallVisor<sup>®</sup>
- DEFINITY<sup>®</sup>
- Avaya IP600 Internet Protocol Communications Server
- INTUITY<sup>®</sup>
- CentreVu<sup>®</sup>
- BCMS Vu<sup>®</sup>
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# 1 Introduction

With the use of the Internet and e-commerce, the paradigm of business is shifting from brick and mortar to the virtual enterprise. This shift is allowing small and large businesses alike to take advantage of new opportunities and find new ways to communicate with their customers.

Competition is fierce as this new technology begins to level the playing field, allowing small companies to more easily compete with larger companies without acquiring a large number of employees or large corporate infrastructure. To compete, the smaller businesses must present an image of a larger company through their sales and customer service operations, as well as their day-to-day communications with their customers. These small companies need sophisticated, highly reliable tools to effectively run their operations, improve customer service, and make the most effective use of their limited resources.

Larger companies face a similar challenge, having to present the same corporate image through all of their locations, large and small. It is critical that their customers receive the same level of service and have the same tools deployed to the field as are in use at the company's headquarters. Ensuring that the customer experience is consistent across all locations can help to improve customer satisfaction and build customer loyalty.

Thus, for both a small business and a small branch of a large corporation, the tools needed to provide the highest level of customer service and improve employee productivity are critical. The most critical of those tools are the communications systems used within the business. These systems provide not only the data and telecommunications resources for the business but also the foundation for the other service and productivity enhancing applications.

The Avaya IP600 Internet Protocol Communications Server brings together the successes of three Avaya products to create a new standard in multiservice IP telephony:

- Call processing that runs on a customized surround-supported NT operating system, co-resident with sophisticated and integrated voice/fax mail, call centers, announcements, SMDR, and Web-based system administration. Additionally, SNMP agent is supported as an option to the standard Expert systems interface.
- Complete IP Gateway and IP Gatekeeper functions that support 100% TCP/IP transport of merged voice and data to the desktop as well as to the network clouds.
- Integration with traditional DEFINITY port architecture that leverages millions of lines of proven code and proven hardware to provide world class reliability/availability and feature richness with the emerging technology of Voice over IP.

This document provides an overview of the Avaya IP600.

Avaya IP600 is a high functionality communications system for customers in the 25-40 line size or smaller with growth potential to 240 IP end points and 400 ports total. This offer supports DEFINITY® Release 9 software, INTUITY® AUDIX® Release 4.4 messaging, and DEFINITY Site Administration (DSA) Release 1.5 on a single hardware platform.

Application integration results in easy administration on the Windows NT 4.0 operating system. In addition, outside adjuncts and associated connectivity and maintenance costs are eliminated and functionality is consolidated into a single cabinet, creating a cost-effective platform.

Avaya IP600 allows for business growth without additional investment. The circuit packs, phones, and cabinet can be used seamlessly in networks with DEFINITY systems, and the port circuit packs and telephones are common across the whole IP600/DEFINITY product line.

## Avaya IP600 Communications System

Avaya IP600 supports DEFINITY ECS Release 9 software, allowing full functionality, including support for the 4600-, 6200-, 6400-, 8400- and 8500-series telephones and terminals. The following features are offered:

- DEFINITY station and trunk circuit packs, such as C-LAN and IP Media Processor
- Asynchronous Transfer Mode (ATM) and features, such as telecommuting
- Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI) access and Distributed Communications System (DCS) and QSIG private networking
- Software options, such as Co-Resident Announcements, SNMP Agent, BCMS Vu, and CentreVu CT Server
- DLG Enhancement, which allows adjunct routing to asai /adjlk TCP/IP links. This feature is co-resident on the processor and does not require any additional hardware.

See [Appendix A, Avaya IP600 Features](#) for a complete list of Avaya IP600 features.

## Announcements

The Avaya IP600 processor supports 8 ports of integrated SSP announcements. They are stored on the hard drive and can be backed up just as translations are. The processor supports 1 hour of noncompressed speech (28.8 Mbytes). You can also import \*.wav files. You can add ports using the TN750C, but the TN750C cannot use \*.wav files. For any given announcement port, up to 256 users can be connected at any one time.

## AUDIX

AUDIX provides a messaging communications solution for unified voice and fax messaging. Accessing voice mail and fax via phone, PC, laptop, and wireless saves the user valuable time. Release 4.4 of AUDIX includes several enhancements in media, access, and connectivity that offer the first truly “universal” messaging product. AUDIX provides voice, fax, and text messaging together with text-to-speech, and Message Manager functionality on a single processor. The new processor board provides DSP resources for messaging and support for TCP/IP.

In addition, the AUDIX CornerStone software base has been modified to support co-resident announcements.

## DEFINITY Site Administration (DSA)

DEFINITY Site Administration (DSA) is a Windows-based system management tool that provides an easy-to-use interface with the Avaya IP600 and AUDIX. The built-in wizards globally update records, add users, and complete other administrative and maintenance tasks. The IP600 is administered on a Windows-based computer. DSA has pull-down menus and online help. Shortcuts can be created to frequently used commands and to templates for frequently used tasks. DSA uses the Graphically Enhanced DEFINITY Interface (GEDI); however, the standard SAT (system administration terminal) interface remains available through terminal emulation.

## Call Center

Call Center functionality in Avaya IP600 consists of the following:

- Basic, Deluxe or Elite Call Center software
- Basic Call Management System (BCMS) monitoring and reporting
- Access to BCMS through DSA using SAT emulation (only one BCMS monitoring/reporting session may be active at one time)
- Scheduled printing of BCMS historical reports is not supported
- BCMS Vu support via the LAN (TN795 processor), which is a stand-alone product that connects to the Avaya IP600 system via the LAN. This option is available at additional cost and is field installable.
- CentreVu CT Server, which is offered as a stand-alone product that connects to the Avaya IP600 system via the LAN. This option is available at additional cost and is field installable.
- Call Management System (CMS) support via the C-LAN board (TN799C)
- TSAPI support via the C-LAN board or the LAN (TN795 processor)
- Passageway Direct Connect support

## Web Browser Access

Avaya IP600 can be administered through a web interface. Using a PEER WEB server through a LAN connection, the administrator can download software (Message Manager, DSA), connect to DEFINITY ECS or INTUITY, schedule a backup or look at backup results, restore from a backup, and start pcAnywhere to act as a console to Avaya IP600.

## Call Accounting

Call Accounting controls communications costs with accurate reporting on calls processed and effective cost-allocation methods. Call Accounting stores call records from phone extensions and assigns costs to the calls. The system also creates ad-hoc reports to manipulate call data and charge-back call expenses to clients or departments. Detection of toll fraud and maintenance of call records are also features of call accounting. Station Message Detail Recording (SMDR) records are written in real time to a file on the local hard disk. Because of Ethernet connectivity, this information is easily accessed from anywhere on the LAN or WAN.

## SNMP Native Agent

SNMP Native Agent is a software module loaded on all systems and available at additional cost. Native Agent provides a SNMP interface to the system's alarm and error tables, select performance measurements, and select configuration data. It also supports SNMP traps for DEFINITY alarms and restarts, INTUITY alarms, and Windows NT events.

## System Administration

Avaya IP600 offers the DEFINITY Site Administration (DSA) Release 1.5 package.

## AMIS Analog Networking

The Audio Messaging Interchange Specification (AMIS) Analog Networking feature lets subscribers exchange voice mail messages with voice messaging systems anywhere in the world, provided those systems also have AMIS analog networking capabilities. The Message Delivery feature allows subscribers to send recorded messages to any touch-tone telephone, including a residence telephone.

AMIS Networking involves:

- Establishing machine names
- Administration of dial strings for all AMIS nodes
- Administration of address ranges
- Testing with other vendors (this requires test mailboxes with password access)
- Post-implementation support for five consecutive business days, with the understanding that the translations have not been changed or modified by the customer

Customers implementing AMIS or TCP/IP networking should consider using the Node Implementation and Testing Offer, which includes administration and testing of end points. Ask your service representative for details.

**Note:** Avaya IP600's TCP/IP design is targeted for Mach 4 and above systems. Systems using IP55 TCP/IP will not function with Avaya IP600.

## Hardware

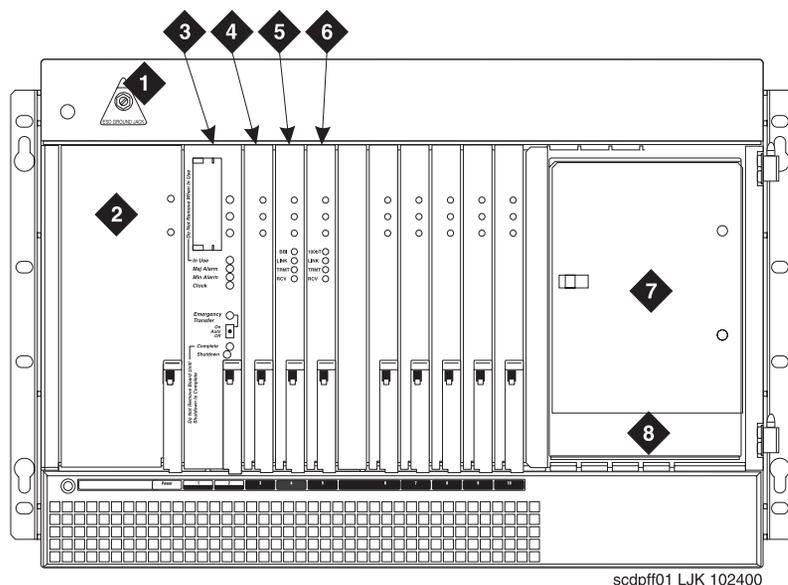
The major components of the Avaya IP600 include the following circuit packs: a TN744 call classifier, TN2302AP IP media processor (IP gateway), a TN799C C-LAN, and a TN795 processor. The processor circuit pack contains:

- Windows NT 4.0 operating system with on-board Pentium processor chip
- Secondary (Motorola) processor running application management firmware
- NT to firmware interface
- Tone clock functionality equivalent to a TN2182 circuit pack
- AUDIX hardware/software with virtual AUDIX ports
- Announcement hardware/software with virtual announcement ports

### Rack-Mounted Cabinet

The Avaya IP600 rack-mounted cabinet weighs 40-50 pounds (180-225 kilograms) fully loaded and is about 12 x 19 x 22 inches (30 x 48 x 55 centimeters — see [Figure 1](#)). It includes 10 slots for circuit packs and a power supply and supports up to 400 ports (may require a second cabinet). The TN795 circuit pack must be in slot 2. The cabinet is designed for rack mounting but can be floor mounted.

Figure 1. Avaya IP600 Rack-Mounted Cabinet



- |   |                                 |   |   |
|---|---------------------------------|---|---|
| 1 | Grounding Receptacle            | 6 | TN2302AP IP Media Processor   |
| 2 | 650A Power Supply               | 7 | Storage Area (containing grounding wrist strap, backup PCMIA flashcard, and documentation library CD) |
| 3 | TN795 Processor                 | 8 | Fiber pass-through area   |
| 4 | TN744D Call-Classifier-Detector |   |   |
| 5 | TN799B/C C-LAN                  |   |   |

*Slots 1 and 2 are reserved for the TN795 circuit pack. Slot 3 is recommended for the TN744D circuit pack, but any other slot is acceptable.*

## Avaya IP600 Site Requirements

Avaya IP 600 is designed to be mounted in a standard 19-inch (48 cm) data rack that has been properly pre-installed and secured as per the EIA 464 (or equivalent) standards. The Avaya IP600 cabinet can be front mounted (as shipped) or mounted at its midpoint.

The customer is responsible for providing the rack and having it installed and secured prior to Avaya IP600 installation. This also applies to providing AC power to the rack. The technicians trained to install the Avaya IP600 do not typically have the tools or proper training for data rack installation.

Installation requires 1 foot (30 cm) of clearance in the rear, and 18 inches (45 cm) of clearance in the front, which is consistent with the EIA 464 data rack standards. In a two-cabinet configuration, the dimensions of the TDM/LAN cable require that the B cabinet be mounted directly over the A cabinet (flush).

The Avaya IP600 should be installed in a well-ventilated area. Maximum equipment performance is achieved at an ambient temperature between 40 and 120 °F (4 and 49 °C) for a short-term operation (not more than 72 consecutive hours or 15 days in a year) and up to 110 °F (43 °C) for a continuous operation.

The relative humidity range is 10% to 95% at up to 84 °F (29 °C). Above this, maximum relative humidity decreases from 95% down to 32% at 120 °F (49 °C). Installations outside these limits may reduce system life or affect operation. The recommended temperature and humidity range is 65 to 85 °F (18 to 29 °C) at 20% to 60% relative humidity.

The other Environmental Considerations and System Protection requirements described in the DEFINITY ECS Systems Description under “Site Requirements” apply to the Avaya IP600 as well.

**Table 1. Cabinet Power Source Information**

<b>Cabinet Style and Power<sup>1</sup> Distribution Unit</b>	<b>Power Sources</b>	<b>Power Input Receptacles</b>
Rack Mount Cabinet. AC power supply (650A integrated power supply)	Single phase 120 VAC with neutral Single phase 240 VAC with neutral	120 VAC, 60 Hz NEMA 5-15R 240 VAC, 50 Hz IEC 320 Japan installations use country specific receptacles for 100 and 200 VAC, 50/60 Hz

<sup>1</sup>There is no integrated DC power supply. DC rectifiers can be used if desired; follow manufacturer’s instructions.

**Table 2. Circuit Breakers for AC-Powered Cabinets**

<b>Cabinet Type</b>	<b>Circuit Breaker Size</b>
Rack Mount Cabinet (120 VAC) 60 Hz	15 A
Rack Mount Cabinet (240 VAC) 50 Hz	10 A

## Reliability

High reliability and availability has been a cornerstone of DEFINITY systems. The hardware is designed to detect and correct errors as they occur, to minimize the number of components that cause system outage, and to simplify fault isolation to a replaceable component. Error detection and correction, system reconfiguration, and alarming escalation paths provide necessary performance elements. The software is designed to recover from intermittent failures and to continue providing service with a minimum of disruption.

The maintenance subsystem manages three categories of maintenance objects: hardware maintenance objects (MOs), software processes, and data relationships. Hardware MOs are tested, alarmed and removed from service by the software. When the problem is isolated, the object is replaced. If a software process encounters trouble, it is recovered or restarted. Data relationships are audited and corrected.

Studies have shown that 95% of problems experienced by DEFINITY systems are self-corrected and occur without affecting the customer. All systems are provided with remote diagnostics capability, which enables rapid troubleshooting and maintenance in the cases where the system cannot heal itself. This whole sophisticated maintenance management implementation is the prerequisite to the 99.99 - 99.999+% availability of DEFINITY systems.

The maintenance philosophy is carried forward into the Avaya IP600, with its new subsystems maintenance management needs added. For example, the following design elements help assure high availability of the Windows NT operating system:

- A secondary on-board processor complex supports initialization, monitoring, and recovery functions for all applications running on the Windows NT OS. The secondary processor takes corrective action when problems are detected in a way to minimize user impact.
- DiskKeeper code is incorporated and runs regularly to eliminate disk fragmentation problems.
- Applications running on the OS are thoroughly pretested to assure proper performance. This OS is closed to any applications other than the manufacturer-provided ones to avoid interference of operation.
- The Windows NT event log is proactively scanned for potential service affecting items. If found, alarms are generated, and, if necessary, a service technician is dispatched.

As another example, the new cabinet uses a three fan, hot replaceable, assembly. The fans automatically sense temperature and adjust their operating speed accordingly. If one fan fails, the other two speed up and are more than adequate to provide sufficient cooling for weeks (or more). In parallel, an alarm is created that dispatches a technician to replace the fan unit.

Coresident applications running on the Windows NT OS reduce cost and complexity by eliminating unnecessary boxes, cabling, and administration tasks. The result is a system that is easier to install and configure than traditional solutions with less risk of error. The integrated DSA tool simplifies the task of configuring the Avaya IP600 and AUDIX, which in turn reduces the likelihood of down time from administration errors.

Avaya IP600 also provides

- System survival of minor power disruptions without service interruption
- Automatic restoration of the last saved version following a power outage
- Scheduled centralized backups of critical system information at remote sites. In an emergency, multiple copies of translations, AUDIX subscriber information, and the Windows NT registry are available. Saved information can be quickly restored.
- IP trunk fail-over to the PSTN (QOS thresholds can be set to drive shift to and from PSTN)
- Option of Emergency Transfer equipment that cuts up to 6 analog lines directly through to CO analog trunks.

Anticipated availability of the Avaya IP600 is 99.99%.

## Incomparable service

Avaya IP600 customers enjoy incomparable service. Avaya's knowledgeable sales associates and technicians are thoroughly trained in remote alarming, expert systems, troubleshooting, and disaster-recovery procedures.



# 2 Desktop/Console Solutions

The communications needs of people in your company vary widely. Some may require only basic telephone service. Others may need effective messaging services or high-speed data communications and access to a variety of host and personal computers.

Avaya IP600 brings voice communications, data communications, and messaging together on the desktop, which enables you to customize the types of service for various users.

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

## Telephones for the global marketplace

A wide variety of telephones, ranging from basic single-line to sophisticated digital service that integrates voice and data communications, are available with Avaya IP600. You can incorporate a mixture of telephone types based on user job function. All of the telephones are easy to use and provide the ability to tap into the power of Avaya IP600.

Avaya IP600 supports four basic families of telephones—Internet Protocol (IP), Digital Communications Protocol (DCP), Analog, and Basic Rate Interface (BRI). These terms describe how each type of telephone communicates with the Avaya IP600. 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.

The IP, DCP, and BRI sets use digital transmission for integrated voice, data, and control signals. These telephones provide a rich array of time-saving and value-adding features.

With help from our many global customers, Avaya has developed the 4600-, 6400-, 8400-, and 6200-series telephones to meet the demand for telephones in the global marketplace.

### 4600-Series IP Telephones

The 4600-series IP telephones are highly integrated, high function standards-based IP end points. They are designed for superior audio quality with full duplex speakerphone and echo cancellation capability. The telephones have two 10/100 Mbps automatic negotiating Ethernet connections: one for the phone and one for a PC. "Blackburst" technology is used to assure that the voice packets receive sufficient priority. G.711, G.729A/B and G.723.1 compression schemes are all supported. (Availability of the compression algorithms is delayed but will be provided to all existing customers free of charge when available). A universal serial bus and an infrared port are built in, ready to support future applications.

These telephones emulate the DCP 6400-series telephones and provide all of the same features except for the group listen speakerphone feature. For a complete list of telephone features, see [Appendix B, Telephone Features](#).

The 4600-series IP telephones can be used with static or dynamic addressing. Dynamic addressing requires a Dynamic Host Configuration Protocol (DHCP) server. Dynamic addressing is one key to reducing phone reconfiguration expenses incurred due to moves. Trivial File Transfer Protocol (TFTP) is supported and allows firmware to be upgraded over the LAN. A TFTP server will be provided free of charge to customers so that they can benefit from future upgrades.

Avaya IP600 supports the following 4600-Series telephones:

- 4624 IP telephone

The 4624 IP telephone is a digital, multi-line IP telephone that has 24 call appearance/ feature buttons and a 2-line by 24-character display. This telephone is designed for the busy executive or executive assistant who requires extensive call handling and call coverage flexibility. The 4624 has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the four display-associated soft keys. The 4624 has a built-in 2-way speakerphone and can be wall mounted.

- 4612 IP telephone

The 4612 IP telephone is a digital, multiline IP telephone that has 12 call appearance/ feature buttons and a 2-line by 24-character display. The 4612 has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the four display-associated soft keys. The 4612 has a built-in 2-way speakerphone and can be wall mounted.

- 4606 IP telephone

The 4606 IP telephone is a digital, single-line IP telephone with 6 call appearance/feature buttons and a 2-line by 16-character display. The 4606 has a built-in 2-way speakerphone and can be wall mounted. There are no soft keys associated with the display.

### Requirements

The 4600-Series telephones require the TN2302 IP Media Processor circuit pack for the audio capability. They also require the TN799C Control-LAN (C-LAN) circuit pack for the signaling capability.

## IP Softphones

IP Softphones extend the level of DEFINITY services. They turn a PC or a laptop into an advanced telephone. Users can place calls, take calls, and handle multiple calls on their PCs. For a discussion of the types of Softphones available with IP Solutions for Avaya IP600, see [“Avaya IP Softphones” on page 66](#).

The following DCP telephones are available:

- 6400-Series digital telephones
- 8400-Series digital telephones

**6400-Series digital telephones**

The 6400-Series digital telephones are versatile 2-wire DCP telephones that support all of the key/hybrid features of Avaya IP600. These telephones have a new, global design and include the following additional features:

- Date and time display
- Feature button that allows switchhook control of a headset
- Conference, Transfer, Hold, and Last Number Dialed fixed feature buttons
- Group Listen capability that allows you to use your handset or headset while others in the room listen via a speakerphone. This 2-way handset, 1-way speaker mode allows you to serve as a spokesperson for a group.
- Station User Administration capability that allows you to program certain features on the telephone yourself.
- Whisper Page that allows an assistant to announce a second call to a company official during an active call on the official's telephone. The announcement is heard only by the official.
- Auto Call Times that allows each call to be timed automatically upon answer. The elapsed call displays on the telephone. The timer is stopped automatically when a call is ended or placed on hold.

The 6400 Tip/Ring Module enables a 6400-series analog adjunct, such as a fax machine or modem to operate independently on 12 channels with its own extension.

**6400-Series DCP telephones**

Avaya IP600 supports the following 6400-Series DCP telephones:

- 6402 telephone

The 6402 telephone is a digital, single-line DCP telephone without a display; it can be wall mounted. This cost-effective, entry-level telephone is designed for users with basic call handling requirements. The 6402 is ideal for areas where there is minimum use, such as reception areas, copy rooms, file rooms, or warehouse locations. This telephone has a Feature button for accessing up to 12 system features and a built-in, 1-way (listen-only) speakerphone that facilitates off-hook dialing and listening to voice mail or broadcast messages.

- 6402D telephone

The 6402D telephone is a digital, single-line DCP telephone with a 2-line by 16-character display. This telephone has a Feature button for accessing up to 12 system features. The 6402D has a 1-way (listen-only) speakerphone and can be wall mounted. There are no soft keys associated with the display.

- 6408+ telephone

The 6408+ telephone is a digital, multiline DCP telephone that has 8 call appearance/feature buttons. This telephone has no display and can be wall mounted. The 6408+ is designed for users who need multiline appearances and extensive features. The 6408+ has a built-in 2-way speakerphone and programmable keys so users can access more system features from the telephone.

- 6408D+ telephone

The 6408D+ telephone is a digital, multiline DCP telephone that has 8 call appearance/feature buttons and a 2-line by 24-character display. This telephone is designed for users who need multiple line appearances and extensive features. The 6408D+ has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the four display-associated soft keys. The 6408D+ has a built-in 2-way speakerphone and can be wall mounted.

- 6416D+ telephone

The 6416D+ telephone is a digital, multiline DCP telephone that has 16 call appearance/feature buttons and a 2-line by 24-character display. This telephone is designed for users with call coverage responsibilities who need multiline appearances and extensive features. The 6416D+ has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the four display-associated soft keys. A 24-button expansion module can be added to provide 24 additional auxiliary buttons. (The 24-button expansion module requires power from the station or the closet.) The 6416D+ has a built-in 2-way speakerphone and can be wall mounted when not used with the 24-button expansion module.

- 6416D+M telephone

The 6416D+M telephone is a digital, multiline DCP telephone that has 16 call appearance/feature buttons and a 2-line by 24-character display. This telephone is designed for users with call coverage responsibility who need multiline appearances and extensive features. The 6416D+M has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the four display-associated soft keys.

A 24-button expansion module can be added to provide 24 additional auxiliary buttons. (The 24-button expansion module requires power from the station or the closet.) The 6416D+M allows you to install a 100A Tip/Ring module, providing a connection between the telephone and such analog adjuncts as modems, fax machines, analog conference-quality speakerphones, answering machines, and TDD machines commonly used by the hearing impaired. The 6416D+M has a built-in 2-way speakerphone and can be wall mounted. The 6416D+M also has a built-in headset jack.

- 6424D+ telephone

The 6424D+ telephone is a digital, multiline DCP telephone that has 24 call appearance/feature buttons and a 2-line by 24-character display. This telephone is designed for the busy executive or executive assistant who requires extensive call handling and call coverage flexibility. The 6424D+ has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the four display-associated soft keys. The 6424D+ has a built-in 2-way speakerphone and can be wall mounted. A 24-button expansion module can be added to provide 24 additional auxiliary buttons. (The 24-button expansion module requires power from the station or the closet.)

- 6424D+M telephone

The 6424D+M telephone is a digital, multiline DCP telephone that has 24 call appearance/feature buttons and a 2-line by 24-character display. This telephone is designed for the busy executive or executive assistant who requires extensive call handling and call coverage flexibility. The 6424D+M has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the four display-associated soft keys. The 6424D+M allows you to install a 100A Tip/Ring module, providing a connection between the telephone and such analog adjuncts as modems, fax machines, analog conference-quality speakerphones, answering machines, and TDD machines commonly used by the hearing impaired. The 6424D+M has a built-in 2-way speakerphone and can be wall mounted. A 24-button expansion module can be added to provide 24 additional auxiliary buttons. (The 24-button expansion module requires power from the station or the closet.) The 6424D+M also has a built-in headset jack.

### Requirements

The 6400-Series telephones are supported by the following 2-wire DCP circuit packs:

- TN2224 (24-port circuit pack)
- TN2214 (international 24-port circuit pack)

### 8400-Series digital telephones

The 8400-Series telephones are versatile 2-wire/4-wire DCP telephones that offer flexibility and cost savings and support most of the key/hybrid features of Avaya IP600. (Table 3 on page 16 shows the differences between the 8400-Series telephones and 6400-Series telephones.) These telephones detect automatically whether they are plugged into a 2-wire or 4-wire digital line circuit card.

**Note:** The 8400-Series telephones are not offered with Avaya IP600 systems sales. However, you can purchase these telephones from Avaya.

Avaya IP600 supports the following 8400-Series telephones:

- 8403 telephone

The 8403 is a 3-line telephone without a display that can be wall mounted. This telephone has a built-in, 1-way (listen-only) speakerphone and three programmable buttons.

- 8405B telephone

The 8405B is a 5-line telephone without a display and that can be wall mounted. The 8405B has a built-in 1-way speaker and programmable keys.

- 8405D+ telephone

The 8405D+ is a 5-line telephone with a 2-line, 24-character display that can be wall mounted. This telephone has a built-in 2-way speaker and programmable keys.

- 8410B telephone

The 8410B is a 10-line telephone without a display that can be wall mounted. The 8410B has a built-in 2-way speakerphone and programmable keys.

- 8410D telephone  
The 8410D is a 10-line telephone with a 2-line, 24-character display. The 8410D has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the four display-associated soft keys. This telephone can be wall mounted.
- 8411D telephone  
The 8411D is a 10-line telephone with a 2-line, 24-character display. This telephone is an enhanced version of the 8410D telephone that has a built-in RJ11C jack, which provides an interface to analog telephone devices (such as a telecopier or a modem) and an RS232 data interface for PassageWay Direct Connection. The 8411D has a built-in 2-way speakerphone and programmable keys. The 8411D has 12 additional features that are accessible via the 2-line by 24-character display and are selected by the 4 display-associated soft keys. This telephone cannot be wall mounted.
- 8434DX telephone  
The 8434DX telephone is a 34-button telephone with a 2-line, 40-character display. The 8434DX has a built-in 2-way speakerphone and programmable keys. The 8434DX has 12 additional features that are accessible via the 2-line by 40-character display and are selected by the four display-associated soft keys. A 24-button expansion module can be added. (The 24-button expansion module requires power from the station or the closet.)

**Table 3. Differences between the 6400-Series telephones and 8400-Series telephones.**

<b>Feature</b>	<b>6400-Series telephones</b>	<b>8400-Series telephones</b>
Whisper Page	Yes	Yes
Group Page	Yes	Yes
Bridged Appearance	Yes	Yes
Personal CO Line Appearance	Yes	Yes
Directed Call Pick-up	Yes	Yes
Group Listening	Yes	No
Station User Administration	Yes	No
Time/Day Default	Yes	No
Adjustable Display	Yes	No
Pull-out Tray	Yes	No
Headset without handset offhook	Yes	No

1 of 2

**Table 3. Differences between the 6400-Series telephones and 8400-Series telephones.**

Feature	6400-Series telephones	8400-Series telephones
Dual-Purpose NEXT button	Yes	No
Auxiliary Jack	No	Yes
Tip/Ring Interface	6416D+M and 6424D+M only	Yes (8411)
2- and 4- wire	2-wire only	Yes
RS-232 CTI Interface	No	Yes (8411)
AD Labeling	Yes	Yes
Active Dialing	Yes	Yes
Context-Sensitive Help	Yes	Yes
Automatic Timer	Yes	No

2 of 2

**Requirements**

The 8400-Series telephones are compatible with all 2-wire and 4-wire DCP circuit packs.

**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 DEFINITY system are analog 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 Avaya IP600 voice features either by entering access codes from your touch-tone keypad or by pressing feature buttons.

Avaya IP600 supports the 6200-Series analog telephone.

**6200-Series analog telephones**

The 6200-Series telephones are single-line, analog telephones.

Three 6200 telephones are available:

- 6210 telephone

The 6210 telephone is a single-line analog telephone that can be wall mounted. This telephone has a built-in Data jack that allows a user to bridge a fax machine, modem, or laptop computer onto the single analog line.

- 6218 telephone

The 6218 telephone is a single-line analog telephone that can be wall mounted. This telephone has a built-in Data jack that allows a user to bridge a fax machine, modem, or laptop computer onto the single analog line. The 6218 also has 8 speed dial buttons and a 2-way speakerphone. The 6218 telephone is available in the United States only.

- 6220 telephone

The 6220 telephone is a single-line analog telephone that can be wall mounted. This telephone has a built-in data jack that allows a user to bridge a fax machine, modem, or laptop computer onto the single analog line. The 6220 also has 10 speed dial buttons and a 2-way speakerphone.

**Telephone Features**

For additional information on digital telephone features, see [Appendix B, Telephone Features](#).

## Wireless Handsets for X-Station Mobility

X-Station Mobility allows remote users to access switch features. That is, X-Station Mobility allows certain OEM wireless telephones remoted over a PRI trunk interface to be controlled by the Avaya IP600 as if the telephones were directly connected to the switch. The telephones are administered to be of the type “X-Mobile” and have additional administration information on the station form that assigns the capabilities of a remote station to the associated PRI trunk group. The wireless telephones thus have access to such DEFINITY features as call-associated display, bridging, message waiting, call redirection, and so forth.

The X-Station Mobility feature offers the following enhancements:

- **Cluster ID Administration** — assigns and reuses cluster IDs based on the radio controller. It significantly eases provisioning of DEFINITY Wireless Business Systems (DWBS) having more than 32 radio controllers.
- **Increased Radio Controller Capacity** — increases the maximum number of DWBS radio controllers from 50 to 150 to support customers with large, multiple-building locations. Associated radio controllers can be placed in different port networks. However, the radio controllers must be isolated from one another to avoid interference.
- **Large Displays** — provides for all of the Avaya IP600 call information to be displayed. The information can be formatted to fit a variety of display dimensions, and formatting is administered through a field on the station form.

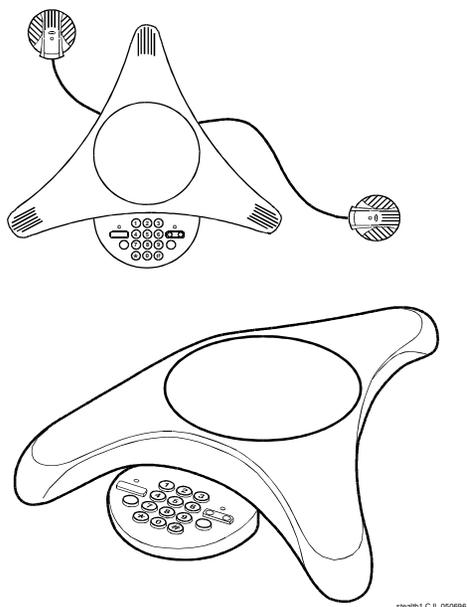
## Teleconferencing Products

### SoundStation Audioconferencing Systems

Avaya's SoundStation and SoundStation EX Audioconferencing Systems enable a group of people in a conference room to share their conversation with others through a telephone connection. The Soundstation equipment permits natural conversation among many people — whether strong or soft, or from a standing or sitting position.

- **SoundStation** — Three microphones and a digitally tuned speaker 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 and a flash key. An additional port allows you to connect the speakerphone to a tape recorder.
- **SoundStation EX** — This system includes all the features and functions of the SoundStation. It accommodates larger conferences by including two palm-sized external microphones that can be positioned up to 6 feet (1.8 m) on either side of the center console. An optional wireless microphone is available for stand-up presenters. See [Figure 2](#) for an illustration of the SoundStation EX with External Microphones.

**Figure 2. Soundstation EX with External Microphones**



## Attendant Consoles

To increase the effectiveness of attendants handling calls, Avaya IP600 offers the following tools:

- DEFINITY 302C Attendant Console (requires connectivity to a 2-wire DCP circuit pack)
- DEFINITY PC Console Release 2.0 or later

### DEFINITY Attendant Console

The DEFINITY Attendant Console is a digital call-handling station with push-button control that enables call attendants to answer, place, and manage calls and monitor selected system operations. The Attendant Display shows call-related information that helps the attendant operate the console. Attendants may select one of several available display languages. For more information, see [Appendix C, Attendant \(Operator\) Features](#).

### DEFINITY PC Console

The DEFINITY PC Console is a software application that enables call attendants to handle incoming calls efficiently using a personal computer. Using the familiar Microsoft Windows interface, attendants can easily track how long callers have been on hold and for whom they are waiting. Attendants can monitor up to six calls at once. They need not fumble with pen and paper when handling calls, as they can make notes on their computers about what each caller needs. Having the call-processing software on the same computer with spreadsheet, word-processing, or other software enables 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. Online telephone identification enables attendants to identify employees quickly. Calls are transferred with the press of a button. Online help makes it easy for attendants to remind themselves how to use the system.

The PC Console is easily customized, so even if attendants from different shifts share the same computer, they can each preserve their preferences in the call-processing environment. The PC Console is available in English, Dutch, Spanish, French, German, Italian, and Portuguese. For example, if a Spanish-speaking attendant takes over for a French-speaking attendant, a single press of a button converts all labels, error messages, and online help to Spanish.

**Requirements**

The following requirements must be met for the PC Console to function properly:

- An IBM-compatible personal computer with:
  - ~ A Pentium-based, 100-megahertz or higher processor
  - ~ A minimum of 16 megabytes (MB) of random-access memory (RAM)
  - ~ A minimum of 4 MB of read-only memory (ROM)
  - ~ A 3.5-inch diskette drive
  - ~ An available COM port
  - ~ Sufficient hard disk space. The space required to support PC Console depends on the number of users you are supporting, the amount of information stored for each person, and whether you will include each person's photograph in PC Console.
- Any of the following operating systems:
  - ~ Microsoft Windows 95
  - ~ Microsoft Windows NT
  - ~ Windows 98
- A 2- or 4-wire DCP telephone with a PassageWay<sup>®</sup> adapter, an 8411 telephone, a CALLMASTER<sup>®</sup> IV, a CALLMASTER VI, a 6424D+M telephone, or a 6416D+M telephone.
- Local adjunct power (depending on your telephone)
- PassageWay direct connection



Avaya IP600 provides the following equipment to supplement the services and features of your system and telephones:

- Power systems
- On hold and delayed announcement systems
- Headsets
- Audio and visual paging
- Alerts and sensors
- Speakerphones
- Security devices
- Call Accounting systems

## Power Systems

Avaya offers the following solutions to provide power for equipment and protection from power disturbances or disasters:

- Online Uninterruptible Power Systems (UPS)

A UPS safeguards your Avaya IP600 and associated applications from utility power irregularities. During a power failure, the UPS battery activates and supplies power for a limited amount of time. This line of UPSs offers advanced battery management and hot-swappable extended battery modules and comes in a 2U-high rack-mounted unit.

- Surge Protectors

Surge protectors help protect PCs, fax machines, and other equipment from electrical surge damage. Alternating Current (AC) Protectors prevent voltage surges from entering the system via the AC utility line. Line Protectors prevent voltage surges from entering the system via incoming central office lines or via wiring for phones that extend to or from another building.

- Terminal Power Supplies

Terminal power supplies provide local power for phones and adjuncts that require additional power, such as DCP phones with headset adapters or adjunct speakerphones. Types of power supplies include

- ~ a desktop AC power module (North American standards)
- ~ 1151A global AC Ethernet-powered module battery backup (optional)

## On hold and delayed announcement systems

Avaya offers the following external announcement systems for Avaya IP600:

- **Magic On Hold Express Systems**

Magic On Hold Express systems provide businesses with fully customized, professionally produced announcements for customer-specific “on hold” environments. The announcements are delivered to your company from the production studio. Production options include legally licensed background music and/or customized information messages that play when a caller is placed on hold or in queue.
- **Magic On Hold Systems**

Magic On Hold systems provide businesses up to 3 minutes of continuous radio programming for customer-specific requirements. Production options include legally licensed background music and/or customized information messages that play when a caller is placed on hold or in queue.
- **Professional Announcement Recordings**

Professional Announcement Recordings (PARs) enhance Auto Attendant, Automatic Call Distribution (ACD), and Integrated Voice Response (IVR) applications. PARs greet and guide business callers using crisp, clear, concise voice messages that optimize a caller’s personal perception of automated communications. Callers hear professional productions that deliver important information.
- **Delay Announcement Systems**

Delay announcement systems serve announce only, information announcement, in-queue announcement, and broadcast messaging for businesses with automatic messaging applications. These systems occupy callers during the call process.

## Headsets

Headsets help increase productivity in telephone-intensive work assignments. Headsets are proven to reduce neck strain and muscle tension when telephones are used at least 3 hours per day. Avaya provides a complete product line for Call Center, traditional business office, computer telephony, and mobile applications.

## Audio and visual paging

Avaya's overhead voice paging equipment allows telephone users to make announcements by speaking into a telephone handset. Avaya IP600 supports as many as 9 paging zones, and 1 zone can be set up to activate every zone at the same time. A zone is the location of the loudspeakers: for example, conference rooms, warehouses, or storerooms.

Visual paging includes indoor LED message display signboards, wireless keyboards for sign programming, connector kits for integrating with the PagePac Plus equipment, and optional software for "ad-hoc" visual message programming.

## Alerts and sensors

Avaya provides a complete product line of alerts and sensors for your business. With alert devices, you can select the type of sound for incoming calls, such as bell, horn, or chime sounds. Or, you can use visual signals such as flashing lights to indicate a ringing call, a voice mailbox message, or a voice paging message.

Sensors detect and analyze central office ringing signals to determine if the signal is a standard voice, data, or fax call. Once it determines the type of signal, the sensor device routes the call to the appropriate end point.

## External speakerphones

External speakerphones provide total telephone operation without using a handset. Turning on a speakerphone is equivalent to lifting a handset when placing or answering a call. Turning off the speakerphone is equivalent to hanging up the handset. Although the majority of Avaya phones have built-in speakerphones, external speakerphones are preferred for applications such as conference calls.

## Security devices

Avaya provides the following security devices:

- Access Security Gateway (ASG)

The Access Security Gateway (ASG) is a built-in authentication feature that offers a more secure alternative to static login passwords for remote access to Avaya IP600. Using an encryption algorithm, the Access Security Gateway uses session-based challenge and response technology to limit access to the Avaya IP600 system's remote maintenance and administration port, system administration terminal, and NET CON channels. See *Installation and Upgrades for DEFINITY ONE Communications System Release 9 and Avaya IP600 Internet Protocol Communications Server* (555-233-109) for information on how to administer this feature.

- Remote Port Security Device

The Remote Port Security Device (RPSD) is a single-line dial-up port protection system that prevents unauthorized access to a host resource. Host resource dial-up ports are protected by the installation of the RPSD lock on the analog telephone line leading to the port. Access is provided only when the calling party uses the RPSD key, a unit that is installed on the analog telephone line at the calling party's end.

The RPSD works with all data communications protocols and can be used in the following applications:

- ~ Protecting organizations with remote and home offices that communicate over the public telephone network via dial-up lines
- ~ Safeguarding companies that administer their communication and voice processing systems remotely from a centralized site, helping to ensure that critical network routing information, traffic data, and PBX feature translations are not compromised
- ~ Controlling dial-up access by a supplier that provides remote maintenance services ensuring that only the service provider has access to the maintenance ports.

## Call Accounting Systems

**Note:** Call Accounting Systems (CAS) do not run co-resident on the Avaya IP600 platform. They are supported ONLY if they can interface to the Avaya IP600 system for CDR records. Also, they must be able to retrieve data over LAN.

Avaya provides the following call accounting systems to reduce service expenses, optimize resources, assign costs, and identify abuse. These products aid in clearly understanding these issues and communicating them to others.

- DEFINITY Network Telemangement (DNT)

DEFINITY Network Telemangement is a state-of-the-art, multi-user telemangement system. Designed in the industrial-strength, on-line, dynamic server-based Informix NT, this application is unmatched in its performance and speed.

DEFINITY Network Telemangement provides a full 32-bit seamless application for client/server environments that use local area networks (LANs) and wide area networks (WANs). DEFINITY Network Telemangement is ODBC-compliant and uses MAPI/TAPI conventions. DEFINITY Network Telemangement is a fully distributed network-based product.

DEFINITY Network Telemangement can support as many simultaneous users as required and is offered in various modules that include call management (call accounting), asset management (inventory control), and service management (work/service orders).

- Call Accounting System for Windows (CFW)

The Call Accounting System for Windows (CFW) allows you to generate comprehensive and accurate accounting reports using the Microsoft Windows 98 or 95 environment. Detailed or summary reports can be expressed in 2- or 3-dimensional color charts and graphs, or in text files suitable for downloading to other applications. The optional toll-fraud detection module enables you to detect fraudulent use of your long-distance services.

You can generate reports that identify:

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

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

You can define up to 5 levels of reporting hierarchy for assignment of costs. The system archives your data for one accounting period. A flexible markup capability allows service businesses to adjust call pricing for each client.

CFW can generate 20 standard historical or real-time reports from as many as 100 locations and 10,000 stations. An individual system is capable of polling different types of call detail storage units or other CFW systems. The remote systems forward call records and alarms as they are generated.

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

CFW is widely compatible and requires little maintenance, even while collecting data, generating reports, and managing remote data collection sites.

- Call Accounting System XP (CAS XP)

Call Accounting System XP has equivalent functionality to the call accounting portion of DEFINITY Network Telemangement. Call Accounting System XP is a LAN/WAN-based application that supports customers with multi-user requirements. It is offered at a base size of 50 stations, but can be upgraded to 10,000 stations. As an option, you can purchase an Internet module that provides employee Internet usage reports.

CAS XP is supported with a mandatory professional service offer for on-site installation, initialization, and training provided by the vendor. CAS XP supports up to 100 sites, 10,000 stations, and 5 simultaneous users. Hacker Tracker is also an available option. CAS XP runs on Windows 98, Windows NT 4.0, and Windows 2000.

# 4 INTUITY AUDIX Messaging

Fewer than 30% of person-to-person business calls reach the intended party on the first attempt. This makes the day-to-day business communications frustrating and can impact productivity. Integration of communications with Avaya's INTUITY AUDIX messaging system helps ensure that calls are not lost.

In addition to call-answer capability, AUDIX provides new opportunities through multimedia messaging. A multimedia message can include text, voice, and fax components. Multimedia messaging allows users to mail a single message to persons on a mailing list, send a message with multiple components to other subscribers, or categorize and store messages for later reference.

This section provides a high-level overview of AUDIX and describes its application and features within Avaya IP600. This section includes:

- Application overview
- Additional sources of information
- Accessing AUDIX administration
- Features of AUDIX

## Application overview of AUDIX

AUDIX is a multimedia messaging application on the Avaya IP600 platform that allows users to integrate voice, text, fax messages, and binary files into a single message.

For example, a sales manager who wants to inform a distributed sales force of a new compensation plan can send a message with both voice and text. The voice component might be

“This message is going to all members of the Northeast Sales region. Congratulations on your excellent results last year. As of January 1st, the compensation plan for new product sales will be changed. Please print the attached text message for detailed information.”

The text message could be created in Message Manager and specify the plan details.

## Additional sources of information

The following additional information for administering AUDIX is available:

- AUDIX System Administration documentation on the *DEFINITY ECS Documentation Library* CD
- The Message Manager Installation chapter in *Installation and Upgrades for DEFINITY ONE™ Communications System Release 9.0 and Avaya IP600 Internet Protocol Communications Server* (555-233-109)
- AUDIX help topics in the DEFINITY Site Administration (DSA) online help application
- Command Line Administration Quick Reference on the *DEFINITY ECS Documentation Library* CD

The *DEFINITY ECS Documentation Library* CD contains the following information:

- Messaging Solutions Quick Reference Guide
- AUDIX Wallet Card
- Message Manager Quick Reference Guide

## Accessing AUDIX Administration

AUDIX Administration tools can be accessed one of two ways:

- Through the DEFINITY Site Administration (DSA) application
- Dialing directly to the AUDIX application using Telnet or a terminal emulator that uses Telnet

## Using DEFINITY Site Administration (DSA)

To establish a connection for AUDIX administration:

1. On DSA window browser, click **Tasks** tab, then click **Add System**.
2. Click **Add Voice Mail System**.
3. Choose a name for AUDIX in DSA in the System Name field.

**Note:** Your telecommunications manager can assign a name or you can choose a name. The name will appear in the **Tree** tab.

4. Indicate the connection method:
  - ~ Modem or data module
  - ~ Direct serial port connection
  - ~ LAN connection
5. When prompted by the Add Voice Mail System Wizard, provide additional information about the connection.
6. Indicate automatic or manual DSA login to AUDIX each time you login.
7. If DSA automatically logs in, enter the AUDIX login and password information.

## Review and test the DSA connection

To review and test the DSA connection:

1. Review the Voice Mail System Summary and make any needed corrections.
2. Click **Test** to try the connection.

If the connection works, DSA displays the login prompt or the AUDIX Command Prompt screen. If the connection does not work, DSA displays an error dialog box with troubleshooting information.

3. Click **Next** and **Finish**.
4. Click the **Tree** tab and confirm that it displays in the tree.

To later change the voice mail system or connection information, right-click **Voice Mail System** in the DSA **Tree** tab and choose properties.

Add as many systems as desired to DSA. If connecting to systems directly using serial ports, you can connect as many switches or AUDIX systems that have ports. If connecting to systems over a network, you can connect to as many systems as needed.

To connect to AUDIX administration:

1. On the DSA browser pane of the DSA window, click the **Tree** tab.
2. Right-click the AUDIX system you want to administer.
3. In the pull-down menu, select **4410 Emulation** or **513 Emulation**.
4. At the `login:` prompt enter login/user name.
5. At the `password:` prompt, enter your password.
6. At the `TERM:` prompt, click **F7** (Continue).

The system displays the **AUDIX Command Prompt** screen.

7. For more information see the following documentation on the *DEFINITY ECS Documentation Library* CD.
  - ~ AUDIX System Administration
  - ~ Command Line Administration Quick Reference
  - ~ Online help topics available from AUDIX administration screens

## Using Telnet to access AUDIX Administration

To access AUDIX via Telnet or another terminal emulator:

1. Set up a connection to local Telnet or other terminal emulator using the following information:
  - ~ Local machine name for host name
  - ~ Specified port number for port
  - ~ VT100 for Term type
2. Once the connection is established, enter login/user name at the Telnet login: prompt.
3. Enter a password at the password: prompt.
4. At the TERM prompt: enter a terminal type, such as vt100, 4410, or 514.

The system displays the **AUDIX Command Prompt** screen.

For a more information about using Telnet, see Chapter 2, “Connectivity and Access” in *Installation and Upgrades for DEFINITY ONE™ Communications System Release 9.0 and Avaya IP600 Internet Protocol Communications Server (555-233-109)*.

## Features of AUDIX

The following AUDIX features allow users to send, receive, and organize voice, text, and fax messages:

- Voice Messaging
- Voice Mailbox
- Transmission Control Protocol/Internet Protocol (TCP/IP)
- Message Manager
- Fax Messaging
- Automated Attendant
- Bulletin Board
- CornerStone Software

### Voice Messaging

The AUDIX Voice Messaging software permits recording and exchanging voice messages with other users. It contains stored voice prompts that help users create, send, retrieve, answer, save, or forward spoken messages. The feature also answers calls for users who are busy or unavailable. In addition to a personal answering service, AUDIX can be used as a messenger to individuals or groups, as an information service, as an office receptionist, or as an automated attendant.

Users and callers instruct the AUDIX Voice Messaging feature by pressing touch-tone keys in response to detailed voice prompts.

Nuances and inflection are integral parts of person-to-person communication. The AUDIX software uses a high-quality voice-encoding algorithm known as Code-Excited Linear Prediction (CELP) to capture the nuances and subtle inflections of the human voice.

Voice Messaging is similar to an electronic mail system in that messages can be sent to other individuals or groups without directly calling the recipient. The message is stored in the recipient's AUDIX mailbox. Recipients can access stored messages at their convenience.

Voice Messaging enables a user to:

- Send messages to other AUDIX and Message Manager users
- Listen to messages received from other AUDIX and Message Manager users
- Forward messages received with comments attached
- Reply to messages received from other AUDIX and Message Manager users
- Create mailing lists containing up to 250 recipients

In addition to basic capabilities, the Outcalling function of AUDIX Voice Messaging allows the feature to:

- Automatically place a call from AUDIX to a user when messages are waiting
- Specify the number to be called by AUDIX when messages are waiting (may be an office, home, car, or pager)

## Call Answer

Call Answer answers a call and records a message when the user is unavailable. This function enables the Voice Messaging feature to:

- Answer incoming calls
- Create personal greetings for incoming calls
- Disable call answer so that a caller hears a greeting, but cannot leave a message
- Customize a set of standard greetings
- Record up to 9 different personal greetings using the Multiple Personal Greeting function
- Play a single greeting for all calls, or assign various personal greetings to play in response to different types of calls, for example, internal and external, busy and no answer, or out-of-hours

## Voice Messaging languages

The AUDIX Voice Messaging feature provides a standard American English announcement set. The announcement set can be replaced or augmented with a number of options, including non-English languages and Telecommunications Device for the Deaf (TDD). For the most recent list of language alternatives, contact your Avaya account representative.

**Multilingual support**

Optional multilingual functions allow callers to interact with the AUDIX application using different languages. For example, callers can follow voice prompts in languages that may or may not match the language of the people they are calling. An administrator can install up to 9 languages on the AUDIX application and operate them simultaneously.

Users can also record personal greetings in two different languages. Prompts are delivered in the selected languages.

**Customized announcements**

Announcements comprise sets of spoken instructions or voice prompts in the AUDIX Voice Messaging application. For example: “To access your mailbox, press star R.”

**Voice mailbox**

A mailbox is a storage area on a computer disk for messages, personal greetings, and mailing lists. AUDIX users acquire a mailbox when they are administered on the application. Each user accesses this mailbox with a private password.

Callers can leave messages in a user’s mailbox, but cannot perform other operations related to the user’s mailbox. After a user logs in, the feature voices the name of the user and reports the number of new messages received. Each message consists of a message header and a message body.

**Incoming mailbox**

Mailboxes are divided into two sections, the incoming mailbox, and the outgoing mailbox. The incoming mailbox receives messages from other users, from the AUDIX application, and from callers redirected to the mailbox because no one answered. The user can save, delete, reply to, or forward messages. There are three categories of incoming messages: New, Unopened, and Old. [Table 4](#) describes each category.

**Table 4. Incoming mailbox categories**

Category	Description
New	A message and header that a user has not listened to. The Message Waiting Indicator (MWI) on the user’s telephone turns on when a new message is present and turns off after the user has listened to it.
Unopened	A message where the header, but not the message itself, has been listened to. The MWI does not stay on for this type of message.
Old	A message that the user has listened to but not deleted.

The system administrator can set the order in which these categories are played to the user.

**Outgoing mailbox**

The outgoing section of a mailbox stores messages that users create, send, or forward. In most cases, messages remain in the outgoing section until delivered. [Table 5](#) describes the outgoing mailbox categories listed in default order. The AUDIX administrator can change this order.

**Table 5. Outgoing mailbox categories**

Category	Description
Filed	Messages that users create and save in the outgoing section of a mailbox. Users can later access these messages to modify them, address and send them again, or delete them.
Undelivered	Messages that have not been sent or messages scheduled for delivery at a future date or time. Users can review, change, or cancel messages and their addresses at any time before delivery.
Nondelivered	Messages that AUDIX could not deliver. The application attempts to deliver a message 10 times (or the administered number of times), then places the message in this category. This usually indicates that the intended recipient's incoming mailbox is full, that the recipient's application cannot recognize or accept a message component (for example, is not fax-enabled), or that there were transmission problems (for example, with an AMIS analog line).
Nondeliverable	Messages defined as "nondeliverable" can be rescheduled for delivery with a new address, or altered to allow forwarding, if needed.
Delivered	Message headers that identify messages delivered but not yet listened to or that identify messages containing nondeliverable components. The latter type of message header is an <i>Incomplete Delivery</i> header. For example, if a message contains more than the four components allowable (that is, a voice, fax, text, and file attachment), the additional components are not delivered, and the message header indicates that a component was not delivered.
Accessed	Message headers that identify messages that have been listened to. A message is considered accessed even if only the header has been listened to.

**TCP/IP**

AUDIX Transmission Control Program/Internet Protocol (TCP/IP) provides the ability to exchange messages with subscribers on other AUDIX systems. The remote system can be next to or geographically distant from the local Avaya IP600 system.

AUDIX TCP/IP uses the proprietary AUDIX digital protocol to exchange messages, user profiles, and message-status information with other machines. The digital protocol uses a digital file format, similar to a data-file transfer between two computer systems, to transmit the information. Digitally transmitted messages are communicated quickly and with excellent sound quality.

TCP/IP allows you to exchange voice, fax, text messages, and attached files from other AUDIX systems. This enables a user to:

- Address messages by name only, known as name addressing. This function applies *only* to administered remote recipients. *Administered* refers to remote users who have been entered in the database of the local application.
- Include the names and telephone numbers of remote recipients in personal mailing lists. Nonadministered remote recipients can be included only by telephone number.
- Hear the spoken name of the intended recipient. If the administrator has not recorded these names, the user hears only the remote mailbox ID.
- Access the names and number directory ( \*  \*  N) to look up telephone numbers by name.
- Assign aliases to remote recipients on systems administered for AUDIX TCP/IP. Administered remote recipients can be included by name or by telephone number. Nonadministered remote recipients can be included by telephone number only.
- Use automatic addressing to respond to incoming messages.

TCP/IP enhances AUDIX Messaging in many ways:

- Customers who exceed the capacity of one AUDIX application at a single location can network multiple machines. This enables users to exchange messages as if they were on the same machine.
- Customers with business offices in more than one location, whether in the same building or in different cities, can exchange messages with every location.

The following functions can be used for messages exchanged between remote users:

- The ability to play a recorded name, when addressing or receiving a message, if a name is recorded for the remote user
- The ability to forward messages to one user or a group of users, respond to messages, and create group mailing lists

The following are additional sources of information:

- AUDIX System Administration documentation on the *DEFINITY ECS Documentation Library* CD
- Online help topics available from the TCP/IP browser screens

## Avaya Message Manager

Avaya Message Manager is a combination of communications modules that function as one software feature. Users can create, send, and receive compound messages containing multiple media types: voice, fax, text, or file attachments to other users inside or outside the corporate environment.

Message Manager is a Windows-based graphical user interface (GUI) that allows AUDIX application message headers to be viewed on a PC screen through a local area network (LAN) connection. The AUDIX application is called the “AUDIX server” when it connects to a LAN.

What distinguishes Message Manager from ordinary voice messaging products is the way users interact with the feature. Users access information visually, instead of listening to voice prompts and using a touch-tone keypad. Viewing message headers on-screen is faster for users because they can quickly view who called, when, and why, without having to listen to prompts, press keys, or remember instructions. The on-screen information helps users access and prioritize important data, more easily develop mailing lists, and track multiple personal greetings.

Message Manager is available in the following languages:

- English
- French
- Spanish
- Brazilian Portuguese
- German
- Dutch

Additional languages are being considered for future releases.

Message Manager includes the basic functions listed in [Table 6](#).

<b>Function</b>	<b>Description</b>
Send messages to multiple recipients	You can create and send a message with one or more message components to one or several people. The message is delivered as soon as possible or can be scheduled for a later delivery time.
Addressing	You can send the message to just one person, to a list of people, or to someone who is on a remote AUDIX system.
Send faxes	The fax software for Message Manager is used to create and send a new fax message. Creating a new fax is similar to printing a hard copy of your work in another program.
Fax from other applications	Although faxes can be stored in and sent from Message Manager, creating and sending a new fax is actually done from any other Microsoft Windows application that allows printing.
Create a custom fax cover page	You can use the Fax Cover Page Designer to add text or bitmap graphics to the fax cover page. You can also use the Designer to change the location and size of the Message Manager text display areas.
Use the Outgoing Folder	After a message is sent, you can check delivery status by opening the Outgoing Folder. The Outgoing Folder lists all messages sent, the time sent, and whether the recipient has received or accessed the message. Additional information is available by double-clicking a message header in this folder.

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**Table 6. Message Manager functions and descriptions**

Function	Description
Build Personal Phonebook	You can use the Personal Phonebook in Message Manager to store “cards” with the addresses of AUDIX subscribers, as well as other numbers and notes. As subscribers are added to the Phonebook, you can quickly add them to an address list. The Personal Phonebook is stored on your PC and can be used while working offline.
Build AUDIX lists	With AUDIX lists, you can store the addresses of sets of people you want to send simultaneous messages to, such as a project team or a corporate department. You can quickly address a message to an entire address list. AUDIX lists are stored on the AUDIX server and are not available offline.
Work offline	If you work away from the office, you may want to edit messages you have received or compose new messages, then later log in and send them during a single call. This saves toll charges because an AUDIX server connection is not required, except when ready to send or receive messages.
Minimize or lock Message Manager	<p>You can minimize Message Manager and still be notified of new messages. Log in to Message Manager, then use standard Windows techniques to minimize the program and keep it active. Later, you can restore the program to retrieve messages or to create and send new messages.</p> <p>A Lock function provides additional security. When you click the Lock button, the application is minimized and requires your AUDIX password to be restored. Locking Message Manager prevents others from accessing your AUDIX mailbox. This function is inactive while working offline.</p>
Record your name or greetings	When you install Message Manager, you can use the name and personal greeting that were recorded through the AUDIX telephone interface. You can also select a menu option to record your name or display a screen to record and manage greetings. The AUDIX server uses the choices you make in Message Manager for playing names or greetings to your callers.
Outcalling	If you are away from the office, you can be notified of new AUDIX messages. Use the Outcalling function to enter a telephone number that the AUDIX server then dials to notify you of new messages.
Sound card	Message Manager uses an audio connection to your telephone to play or record voice messages or greetings. However, you can use your computer’s sound card with speakers and a microphone instead. This is the only way to play or record voice messages while working offline.

2 of 2

The following information is available to Message Manager users:

- Message Manager Quick Reference Guide, available on the *DEFINITY ECS Documentation Library* CD
- Message Manager online help, available by selecting Contents from the Message Manager Help menu
- A file customized just for your site, described in the Updating Your Site-Specific Information section of the Message Manager Installation chapter in *Installation and Upgrades for DEFINITY ONE™ Communications System Release 9.0 and Avaya IP600 Internet Protocol Communications Server* (555-233-109).

## FAX Messaging

The Avaya FAX Messaging feature combines the send and receive capabilities of a stand-alone fax machine or fax modem on a PC with the capabilities of Avaya messaging. Besides sending, receiving, and printing a fax over the telephone, a user can also forward a fax, annotate a fax with a voice message, or send and broadcast a fax to multiple telephone users. These features allow a user to handle a fax message just as they would a voice message.

The following information can be provided to FAX Messaging users who have Message Manager:

- Message Manager Quick Reference Guide, available on the *DEFINITY ECS Documentation Library* CD
- Message Manager online help, available by selecting Contents from the Message Manager Help menu

The following information can be provided to FAX Messaging users who do not have Message Manager

- Messaging Solutions Quick Reference Guide, available on the *DEFINITY ECS Documentation Library* CD
- Online help available from the telephone user interface by pressing   or   at any time

## Automated Attendant

An automated attendant is an interactive telephone answering system. It answers incoming calls with a prerecorded announcement and routes each one based on the caller's response to menu options and prompts.

The system administrator sets up an automated attendant so that callers hear a menu of options. Callers indicate the desired menu option by pressing the corresponding touch-tone key. The automated attendant executes the selected option. Callers from rotary or dial-pulse button telephones are, typically, told that they can hold or call another number to speak with a live attendant.

An automated attendant menu system, or *menu tree*, can be designed to contain subordinate layers of menus or bulletin boards. These sub-menus, or *nested menus*, play additional options, including a choice leading to another nested menu.

The voiced menu options that callers hear are actually personal greetings that the administrator records for the automated attendant's extension. As with any personal greeting, the content of the message can be changed. The Multiple Personal Greetings function provides different menus and options for different types of callers.

If your messaging system has multiple language sets available, the menu options route callers to a sub-menu voiced entirely in another language. The Multiple Personal Greetings function can record menus in various languages.

For more information on setting up and maintaining automated attendants, see the AUDIX System Administration documentation on the *DEFINITY ECS Documentation Library* CD.

## Bulletin Board

A bulletin board is an electronic messaging system. Callers dial the bulletin board's telephone number and the system answers and plays a recorded message. The major difference between a bulletin board and an automated attendant is that a bulletin board does not have an option to route callers to a live attendant.

## CornerStone Software

The CornerStone software has been modified, and now supports co-resident announcements.

# 5 Call Center

The Avaya IP600 Call Center applications are designed to efficiently connect each caller to the appropriate representative. Before the call is routed, information is captured about the caller and integrated with existing databases (see [Chapter 7, Computer Telephony Integration](#)). The combined data is used to match the caller to an agent. Additional features politely inform callers waiting in queue (a holding place for incoming calls) of the length of time it will take to process the call. Detailed call statistics are constantly available to the agents and supervisors.

Calls into the Avaya IP600 Call Center are queued up and routed based on information the system continually acquires. Each caller can be presented with a variety of options for leaving a voice message, a fax, or monitoring the status of the call. Using CONVERSANT voice response software, the system responds appropriately to spoken information.

The following are switch features:

- Automatic Call Distribution, which manages call traffic and workflow
- Basic Call Management System (BCMS), an optional product which provides call management reporting for smaller Call Center operations
- Attendant Vectoring
- Call Center Basic, Call Center Deluxe, and Call Center Elite, which enable you to set up a Call Center (switch)
- CentreVu Virtual Routing
- CentreVu Advocate

The following are PC applications:

- BCMS Vu, which enhances the capabilities of the Basic Call Management System
- CentreVu Computer Telephony (CT)
- CentreVu Call Management System (CMS).  
CMS add-ons enhance CMS, and include:
  - ~ CentreVu Supervisor
  - ~ CentreVu Explorer II
  - ~ CentreVu Visual Vectors

CentreVu Compact Call Center Solutions packages are available in Basic and Enhanced versions.

Avaya IP600 provides an applications platform that consists of several elements. When these elements are integrated to meet business requirements, advanced call distribution and management capabilities deliver the performance and growth necessary for your business success.

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

## Automatic Call Distribution (ACD)

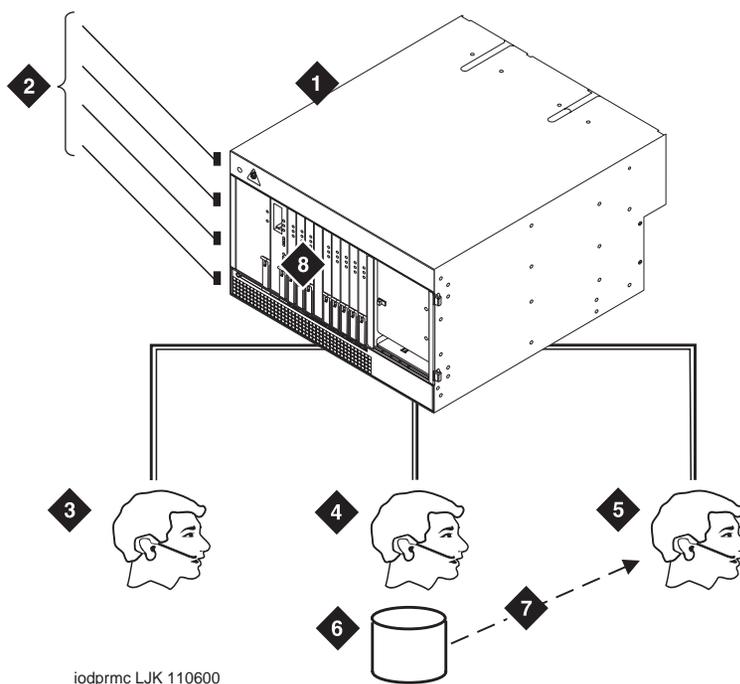
If your company has departments (such as sales, billing, or customer service) that handle large volumes of incoming calls, you can benefit by using Avaya IP600's powerful ACD capabilities. ACD is the basic building block for Call Center applications.

ACD offers a method of distributing incoming calls efficiently and equitably among available employees or agents. ACD also offers a number of ways to connect an agent to a call. For example, with most idle agent distribution, an incoming call is routed to the agent who has been available for the longest time, resulting in more balanced agent workload.

Agents in an ACD environment are assigned to a hunt group, a group of agents handling the same types of calls. Avaya IP600 supports up to 99 different hunt groups. Each hunt group has associated trunks, stations, recordings, and queues. You can assign many ACD features on a per-hunt group basis to meet the different needs of diverse agent groups. You can link a telephone number to an ACD hunt group by associating a published number (often an 800 number) with the hunt group's extension number.

In the [Figure 3 on page 45](#) example of a travel agency, Hunt Group A receives calls only when agents are available since it has no queue. Calls to Hunt Group B can be queued while agents are unavailable and then redirected to Hunt Group C if not answered within an administrable time. Calls to Hunt Group C are redirected to AUDIX if they are not answered within an administrable time.

Figure 3. A Basic Example of Automatic Call Distribution



- |                            |                                |
|----------------------------|--------------------------------|
| 1 Avaya IP600              | 5 Group C: General Information |
| 2 Incoming Trunks          | 6 Queues                       |
| 3 Group A: Business Travel | 7 Call Coverage to Group C     |
| 4 Group B: Personal Travel | 8 AUDIX                        |

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

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

Two features provide for redirection of ACD hunt group calls:

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

ACD agents can use any DEFINITY telephone. The CALLMASTER digital telephone is particularly recommended to meet the needs of ACD agents. A number of special ACD agent features can be assigned to agents' telephones to enable them to perform their jobs effectively. In addition, special features are available to assist supervisors in observing and monitoring the performance of these agents.

Additional features provide even more options when using ACD:

- Queue-Status lamps or displays (on telephones with a digital display) show call status for calls waiting in an ACD queue. Queue-Status also displays oldest call waiting time.
- Dialed Number Identification Service allows agents to identify (via display telephones) the purpose of each incoming call and appropriately greet the caller.
- Automatic Available hunt group allows the CONVERSANT Voice Information System or other "nonhuman" agent positions to be staffed automatically and made available.
- Each agent can be logged in to as many as four hunt groups at once.
- Malicious Call Trace allows you to designate stations that can trace emergency or threatening calls. When an agent receives a malicious call, the agent presses the Malicious Call Trace button. The system gathers trace information and connects a voice recorder to the call. All equipment used to complete the call is held up (the call cannot be disconnected) until the feature is deactivated.
- Redirection on No Answer allows an unanswered, ringing call to be redirected to an ACD queue or to a Vector Directory Number after an administered interval. The agent position will also be taken out of service.
- Station Hunting allows calls to be routed first to the called extension, then according to a linear, circular, or modified circular sequence of extensions. The circular sequences work to distribute calls equitably, ensuring that there are no overworked "first" extensions in a hunt group.

## Basic Call Management System

The Basic Call Management System (BCMS), an integrated, internal capability of Avaya IP600, is a cost-effective solution for small start-up Call Centers or for existing companies with minimum system-measuring/reporting requirements. BCMS helps you fine-tune your Call Center's operation by providing reports with the data necessary to measure Call Center agent performance.

This feature offers call-management control and reporting at a low cost for Call Centers of up to 50 agents. BCMS is ideal for companies that need call management features.

BCMS collects and processes Avaya IP600 ACD call data (up to 7 days) within the system; an adjunct processor is not required to produce call-management reports.

BCMS provides various measurements for monitoring the operations of an ACD application. BCMS software organizes ACD calls and Call Center measurements into different reports that supply useful information for managing ACD facilities and personnel. The reports can be displayed on the system administration terminal in real time.

The following reports can be generated:

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

## Attendant Vectoring

This is a Avaya IP600 feature for calls where the caller is seeking an attendant. It provides a lower-priced solution for customers who want to vector attendant calls that go to an attendant without purchasing the full vectoring software. This alternative provides some powerful capabilities, such as announcement in queue, time-of-day routing, and routing with coverage. Some of the vector steps include:

- Announcement
- Busy
- Disconnect after announcement
- Go to step/vector:
  - ~ Time-of-day
  - ~ Unconditionally
  - ~ Queue-fail
- Queue-to:
  - ~ Attendant group
  - ~ Attendant
  - ~ Hunt-group
- “Route-to number with coverage y/n”
- Wait-time hearing silence/ring back/music
- Stop

## Call Center Basic

The Call Center Basic package, a Avaya IP600 feature, enhances your Call Center by providing the following features:

- Automatic Call Distribution (ACD)
- Auto Available Split
- Most Idle Access (MIA) Across Splits/Skills Option
- MIA Treatment for After Call Work (ACW)
- Multiple Call Handling on Request
- Forced Multiple Call Handling
- Move Agent/Change Skills while Staffed
- Multiple Announcement Boards
- Redirect on no Answer (RONA)
- Service Observing by Class of Restriction

- Service Observing Remote
- Timed After Call Work/Agent Pause Between Calls
- VuStats (including the Service Level and Login IDs enhancements)

**Note:** 12-Agent Call Center Basic is included with Avaya IP600 software.

## Call Center Deluxe

The Call Center Deluxe package, a Avaya IP600 feature, enhances the basic package by including sophisticated Call Center capabilities such as advanced routing, vectoring, and expected wait-time announcements. The Call Center Deluxe package includes the capabilities of the basic package, plus the following features:

- Call Work Codes (CWC)
- Call Vectoring
- Call Prompting (Administrable Interdigit Timeout and Administrable Converse Data-Passing Rate)
- Redirect on No Answer to Vector Directory Number
- Support Network-Provided Digits (Caller Information Forwarding)
- Service Observing on Vector Directory Numbers
- Vector Directory Number-of-Origin Announcement
- Vector Directory Number Return Destination
- Vector Administration (Route-to with/without Coverage and Multiple Audio/Music Sources)
- Vector-Initiated Service Observing
- Vectoring Advanced Routing
- Automatic Number Identification/Information Indicator (ANI/II) Digits Routing
- ASA Routing
- Best-Service Routing Single Site
- Estimated Wait Time Routing (EWT) Routing
- Vector Directory Number Calls Routing
- Wildcard Matching

## Call Center Elite

The Call Center Elite package, a Avaya IP600 feature, enhances your Call Center by including all the capabilities of the deluxe package in addition to the following features:

- Expert Agent Selection
- Reason Codes for Login, Logout, and ACW

## CentreVu Virtual Routing

CentreVu Virtual Routing, an Avaya IP600 feature, helps you provide the best possible service to your customers while using all your Call Center resources wisely and cost-effectively. CentreVu Virtual Routing allows multiple locations to work together as a single virtual Call Center. Its smart routing capabilities monitor and anticipate changing conditions across your virtual Call Center network to find the best place to deliver each call, every time.

CentreVu Virtual Routing can help you:

- Save on network costs
- Optimize existing resources
- Balance agent workloads
- Ensure consistent and reliable customer call handling and service
- Equalize enterprise-wide call volume across sites or across multiple splits/skills at a single site

CentreVu Virtual Routing provides Best Service Routing — the ability to automatically deliver each call to the best place based on a combination of criteria. You can use Best Service Routing with CentreVu Advocate to make your multisite routing even more precise and effective. Once Best Service Routing delivers a call to the “right” Call Center or split/skill, CentreVu Advocate determines the best agent to handle the call based on the caller’s needs and the caller’s value to your business.

In addition to Best Service Routing, CentreVu Virtual Routing offers Enhanced Look-Ahead Interflow (LAI) multisite routing. LAI can help improve customer service and satisfaction by speeding the distribution of calls among locations or skills/splits with low call volumes and long hold times.

CentreVu Virtual Routing also supports enhanced information forwarding to provide valuable details along with each routed call. The information “attached” to each call may include:

- Vector Directory Number (VDN)
- Caller-supplied collected digits
- Dialed Number Identification Service (DNIS)
- Accumulated time waiting

CentreVu Virtual Routing also passes along a Universal Call ID (UCID), a unique identification “tag” that is attached to each call and remains with the call as it is routed throughout your network. By passing Universal Call ID, CentreVu Virtual Routing enables lifetime tracking of calls routed among Call Centers, Avaya IP600 systems, or adjuncts such as CONVERSANT for interactive voice response.

## CentreVu Advocate

You can leverage your Call Center as a strategic business asset with Avaya's innovative CentreVu Advocate software solution. CentreVu Advocate, a Avaya IP600 feature, eliminates the chaos and randomness associated with call handling and provides directed routing with customer-pleasing results. This breakthrough software offers new methodology for aligning your enterprise objectives with agent and management performance and customer needs.

With CentreVu Advocate, you can drive Call Centrally Center performance according to your business plan. This application features expert routing algorithm software from Bell Labs that lets you implement complex customer service, agent resource, and enterprise planning strategies as a critical formula in Call Center operations. With CentreVu Advocate, you can transform your Call Center into a powerful strategic advantage for your enterprise. CentreVu Advocate works in conjunction with and requires Expert Agent Selection (EAS).

Advocate will provide your Call Center with the most innovative methods and enhanced flexibility in selecting the optimum agent for a call or the best call for an agent. With CentreVu Advocate, you determine which call to select the moment an agent becomes available.

CentreVu Advocate provides the following features:

- Service Objective

This capability enables you to establish a unique service objective for each skill in your Call Center. Service Objective can be used to establish different levels of service for multiple types of calls with various media and priority handling needs. You can match the service levels your customers expect by combining the power of your service-level plan with the power of Service Objective.

- Predicted Wait Time

Predicted Wait Time will enable your Call Center to predict service-affecting events while minimizing the impact on your key Call Center metrics. By balancing the average speed of call answering across skills, this feature provides more uniform customer service levels. By matching the needs of your caller to the skills of your agent, Predicted Wait Time ensures that all calls are given the best possible service. Predicted Wait Time will help your Call Center build stronger customer relationships and will improve your overall Call Center efficiency.

- Least Occupied Agent

This capability distributes calls evenly across all available agents in order to balance the workload among those with few skills and those with several skills. When one or more agents are available, Least Occupied Agent uses agent occupancy rather than position in an idle agent queue to determine which agent to select when a call arrives. Least Occupied Agent can help you maintain your staff by promoting agent fairness and eliminating hot seats.

- Service Level Supervisor with Reserve Agents

Service Level Supervisor gives you the ability to set Estimated Wait Time (EWT) thresholds for skills and to assign agents as reserve, in the event a skill overruns its threshold. Service Level Supervisor will override your agents' normal call handling preference to assist calls from a skill whose threshold has been exceeded. This feature allows your Call Center to rapidly adjust to high traffic conditions with the flexibility of automatically activating predefined Reserve Agents when a skill is in an over-threshold condition. This feature will improve your overall efficiency by eliminating the need for your supervisors to manually intervene when traffic conditions change and by effectively scheduling workloads for agents with multiple skills.

- Percent Allocation

Percent Allocation allows you to designate the percentage of time your agents spend in each skill. Incoming calls are matched to those agents with the "best fit" based on their allocated skill percentage. By scheduling an agent's time among multiple skills, you can better utilize and schedule your agents. Percent Allocation can also improve agent performance and satisfaction by assuring them a certain amount of time in each skill.

## BCMS Vu

BCMS Vu Release 2 is a 32-bit client/server software application that works with the Basic Call Management (BCMS) software. The BCMS Vu client runs on Windows 95/98, Windows NT 4.0 or Windows 2000. (BCMS Vu client does not support Windows 3.1 or later or Windows for Workgroup 3.11 or later.) The BCMS Vu server runs on Windows NT 4.0 or Windows 2000.

Using BCMS Vu, Call Center managers can:

- Capture BCMS historical data and store the data on the PC for up to 1 year (depending on the amount of information being stored)
- Report on the historical data
- Monitor the BCMS real-time data in graphical and tabular form
- Display BCMS real-time data on a wallboard
- Display text messages on a wallboard
- Schedule printing of real-time reports

BCMS Vu comes with pcAnywhere software, enabling Avaya's maintenance engineers to perform remote diagnostics and maintenance for BCMS Vu.

**Note:** The Web interface on Avaya IP600 includes the ability to download call center clients for BCMS Vu.

## CentreVu CT

Computer Telephony Integration (CTI) is the linking of telephone communication systems to personal computers, which can increase productivity and customer satisfaction through the exchange of information between the PC and the telephone. CTI applications integrate data processing, data communications, and voice communications.

### Requirements

System requirements for single machines running BCMS Vu and CentreVu CT are:

- An IBM-compatible Pentium single processor
- A minimum of 64 megabytes (MB) of RAM for the server (Windows NT or 2000) and 32 MB of RAM for the clients (Windows 95 or 98)
- A minimum of 500 MB of hard-disk space (recommended)
- The requirements for disk space on a user's PC depend on the size of the Call Center configuration and on the requirements for storing the historical data.
- 10 Base-T network interface card
- A double-speed (2X) or higher CD-ROM drive
- A second serial port is required for remote maintenance using pcAnywhere if connecting to an external modem. A third serial port is required if you are connected to a wallboard.
- A PC with:
  - ~ A sound board and speakers are required for CD-ROM training
- Microsoft Windows NT 4.0 or Windows 2000 is required for BCMS Vu Server and CentreVu CT and Windows 95/98/NT/2000 is required for BCMS Vu Client.
- BCMS software installed on Avaya IP600
- LAN connectivity between the BCMS Vu Server and Avaya IP600

For more information on LAN connectivity and installing BCMS Vu, see the [\*BCMS Vu Software Release 2.0 Version 2 Installation Guide\*](#).

**Note:** The Web interface on Avaya IP600 includes the ability to download call center clients for BCMS Vu and CentreVu CT.

## CentreVu Call Management System (CMS)

The performance of the CentreVu Call Center is critical to your business success. The CentreVu Call Management System (CMS) supplies the tools needed to use the knowledge of the present as well as the past to improve performance in the future. Call Center supervisors and managers can answer questions about call handling, agent workload, and traffic capacities to create a Call Center that delivers maximum productivity while controlling expenses.

CentreVu CMS offers you one of the most comprehensive and advanced Call Center management systems in the industry. CentreVu CMS has sophisticated control mechanisms and reporting capabilities for effective management of Call Centers of all sizes, including multi-location operations.

CentreVu CMS provides a comprehensive array of real-time and historical reports on virtually every aspect of Call Center operations. Managers can get real-time reports, updated as often as every three seconds, and historical reports that summarize call data into daily, weekly, or monthly totals.

Enhanced features built into the standard software include customization of real-time and historical reports, exception notification, and the ability to design, test, change, and store call vectors in real-time. These features allow your Call Center managers to fine-tune the Call Center on the fly to maintain peak performance levels. You will be able to quickly:

- Analyze trends
- Establish performance benchmarks
- Plan new marketing or customer service campaigns
- Match personnel resources to caller volumes and skill needs
- Identify areas for productivity gains and cost savings
- Identify training needs by agent and application

Optional features include Multiple ACD reports and “what if” forecasting. CentreVu CMS provides the information needed to manage the people, traffic load, and equipment in an ACD environment.

CentreVu CMS operates on a Sun SPARCserver or Ultra enterprise 3500 platform with a high performance reduced-instruction-set computer (RISC) processor in conjunction with the ACD features of CentreVu Call Center. Status information is sent to CentreVu CMS from the Avaya IP600 while ACD activities are in progress. This information includes specific event data on calls by agent, agent group, station, queued calls, trunks, trunk groups, and agent actions. With optional Call Vectoring, vector and Vector Directory Number (VDN) data is also tracked and stored. CentreVu CMS provides the information needed to manage the people, traffic load, and equipment in an ACD environment.

**Note:** Avaya IP600 does not support the CMS High Availability option.

## CMS add-on packages

The following sections describe CMS add-on packages:

- CentreVu Supervisor
- CentreVu Explorer II
- CentreVu Visual Vectors

### CentreVu Supervisor

CentreVu Supervisor is an effective management tool that expands the capabilities of the CentreVu Call Management System (CMS). CentreVu Supervisor gives call center managers access to these capabilities and much more — all from the convenience of a desktop or laptop PC.

Now you can view your Call Center through a user-friendly, graphical user interface (GUI). With CentreVu Supervisor, the powerful capabilities of CentreVu CMS are expanded to provide a variety of administrative tools and reports to maximize your Call Center performance. CentreVu Supervisor enables you to:

- Generate status reports in full customizable color graphical formats that are easy to interpret at a glance
- Perform administration tasks easily using a mouse versus a series of commands
- Run other PC applications while actively monitoring Call Center conditions
- Create thresholds for each individual supervisor or manager
- Connect to a LAN allowing a CentreVu Supervisor user to print reports on any network printer for which the user has permissions
- View reports on the Web, saving time and distribution costs
- Schedule reports, printing and other administrative operations at a later time
- Access multi-site, real-time reporting for optimal Call Center management

CentreVu Supervisor gives Call Centers access to these capabilities from the convenience of desktop PC supported by Windows 95, Windows 98, or Windows NT 4.0.

The recommended PC configuration to support Call Center client applications in a Windows environment is:

- Processor: Pentium 133 MHz or faster
- RAM: 48 megabytes
- Resolution: SVGA with a graphics adapter supporting 16-bit color (64K colors) or higher, with 800 x 600 resolution or higher
- Available free disk space: 30 megabytes or more before installation of CentreVu Supervisor (English)
- Communications: TCP/IP protocol stack

## CentreVu Explorer II

CentreVu Explorer II takes your CentreVu CMS reporting capabilities to the next level by providing a more granular view of agent and call activity throughout your call center operation. CentreVu Explorer II is an optional, server-based application that collects and stores the historical information that is gathered in CentreVu CMS. Using standard web-browser software and CentreVu Explorer II's graphical user interface, you can easily access CentreVu Explorer's unique query and reporting capabilities from virtually any client PC.

CentreVu Explorer II gives your Call Center the following advantages:

- **Cradle-to-Grave Reporting**

All queries result in the return of accurate information produced by your Call Center. With CentreVu Explorer II, you have a complete view of all touch points for a caller, including the number of times the caller was transferred or placed on hold plus each call's total hold and call-handling time for the caller for months and even years after the actual call was received.
- **Continuous Query Engine**

CentreVu Explorer II's query engine enables thousands of query combinations to transform your current Call Center information into strategic knowledge.
- **Reporting Engine**

Common queries can be created and shared with all system users for efficient and consistent reporting.
- **Efficient Automatic Number Identification (ANI) Analysis**

CentreVu Explorer II implements powerful analysis and queries of ANI.
- **Customer Classification**

Using Information Indicator (II) digits, available with ISDN, CentreVu Explorer II allows the analysis of a call's origin, identifying customers who call from pay phones, prisons, hotels, coin, and cellular phones (to mention a few).
- **Abandoned Call Analysis**

CentreVu Explorer II provides details not only for callers who abandon the queue, but also those callers who abandon while placed on hold by the agent. Without expensive custom software, information is rarely available regarding callers who abandon a Call Center.
- **Special Call Treatment Analysis**

CentreVu Explorer II tracks and stores unique call events such as calls marked as malicious, calls having audio problems, or calls that were service observed.

Detailed call information, along with the CentreVu Explorer II software, is stored on a Microsoft Windows NT server with SQL 7.0 connected to the Call Center's local area network (LAN). Call Center personnel simply use their desktop PCs, equipped with standard Web browsers, to access the server and retrieve, sort, and analyze call data stored in the CentreVu Explorer II's local database. CentreVu Explorer II enables you to track how each and every incoming call was handled.

You can use your Windows-based workstations with a Web browser to connect to the LAN and use the CentreVu Explorer II GUI to access the local database and access details such as how many times a call has been put on hold or transferred, and by whom. With CentreVu Explorer II, your Call Center managers can select and analyze a comprehensive array of detailed call criteria, produce a variety of reports, and perform database administration, all from the convenience of their desktop PCs.

CentreVu Explorer II transforms valuable CentreVu Call Center information into powerful knowledge. With CentreVu Explorer II, you can feel confident that you're making informed decisions and evaluating your business armed with all the knowledge available to you.

## CentreVu Visual Vectors

CentreVu Visual Vectors is a client application that communicates with CentreVu CMS through CentreVu Visual Vectors server software residing on the CMS platform. CentreVu Visual Vectors is a Java application that provides a GUI for creating and editing vectors and administering VDN assignments. Icons are provided for vector steps, with the capability to display actual vector contents in text format. Customers can use "drag and drop" operations to construct or edit vectors. Additional information can be associated with the vector steps. For example, comments can be attached with descriptions of announcements or route-to destinations. The vector editor can be used in a standalone mode to create or edit vectors and store them in a local scratchpad on the client for later installation on an ACD.

## CentreVu Compact Call Center Solutions Packages

The CentreVu Compact Call Center solutions are an easy and cost-effective way for businesses to implement small Call Centers. Two packages are available:

- **Basic Package**

The Basic Package offers the following features:

- ~ Support for 6, 12, 25, or 50 agents
- ~ DEFINITY Release 9 and Release 9 Deluxe Call Center software Right-to-Use (RTU) license
- ~ Basic Call Management System RTU license
- ~ BCMS Vu Release 2 single-user license
- ~ CD-ROM-based ACD/Vectoring training
- ~ CD-ROM-based Basic Call Management System administrative training

- **Enhanced Package**

The Enhanced Package offers the following features:

- ~ Support for 6, 12, 25, or 50 agents
- ~ Deluxe Call Center software RTU license
- ~ Basic Call Management System RTU license
- ~ BCMS Vu Release 2 single-user license
- ~ CD-ROM-based ACD/Vectoring training
- ~ CD-ROM-based Basic Call Management System administrative training
- ~ Avaya IP600 Integrated Announcements are activated to enable delayed announcements
- ~ Avaya IP600 Call Classifier circuit pack, which enables call centers to offer callers simplified call prompting capabilities for basic menu selections and routing options without the need for a CONVERSANT system

# 6 Wireless Solutions

Most businesses today struggle to improve customer service and increase profits while controlling staff size and costs. To maintain a balance between service and costs, employees must be more productive, responsive, and mobile in performing their jobs. Wireless solutions offer cost control by:

- Reducing time and resources paging employees
- Not having to interrupt work to find a telephone
- Not having to rush to answer calls
- Not having to be tethered to a desk waiting for an important call

Reliable wireless tools remove the fear of losing customers who cannot reach you at your desk.

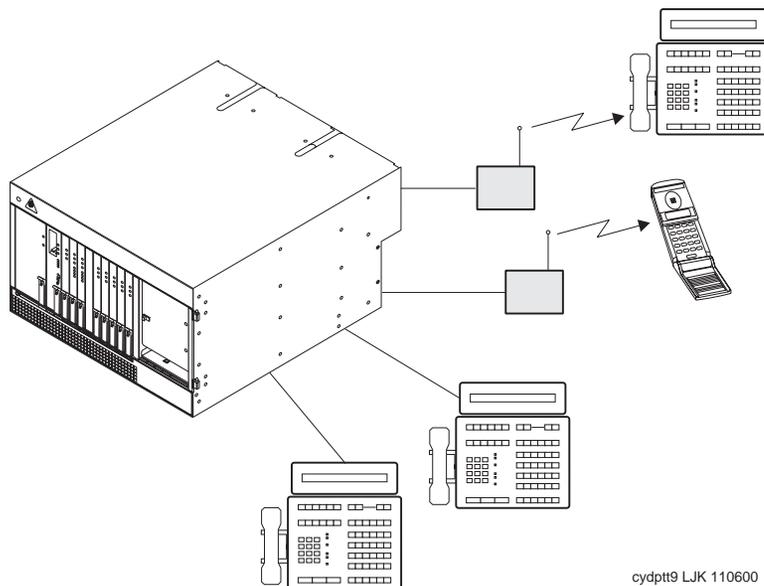
Avaya is the top U.S. provider of wireless solutions for business. Avaya's cordless telephones and speakerphones provide the freedom to place and receive calls while out of the immediate work area. Avaya's Mobility Solutions offer a range of options, from cordless telephones to integrated cellular business systems that greatly enhance the flexibility of wireless services.

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

## Medium-Range mobility

The TransTalk 9000 depicted in [Figure 4 on page 60](#) is a multiline, single- or multizone solution that allows you to roam up to 700 feet (213 meters) from the base station. In most business environments it covers up to 500,000 square feet (45,000 square meters).

Figure 4. TransTalk 9000



cydptt9 LJK 110600

TransTalk 9000 is available in two configurations:

- Complete System: carrier that holds up to six radio modules, MDW 9031 pocketphone, and corresponding charging cradles, radio modules, and holsters
- Stand-alone: a single radio module, wireless telephone, charging cradle, and holster

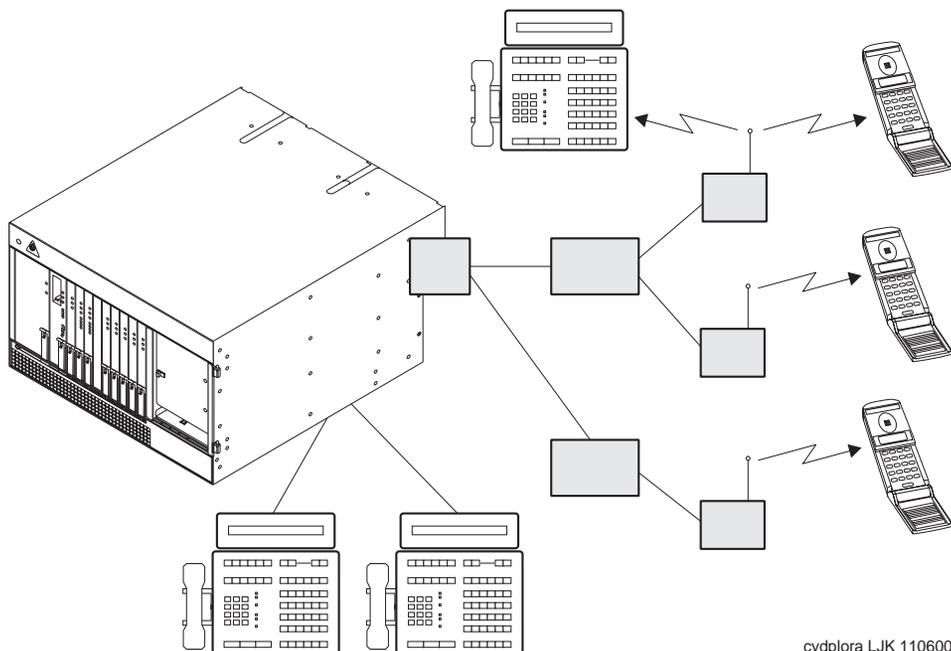
Avaya's wireless telephones offer the following features:

- Crystal-clear voice quality
- Consistent privacy and secure operation
- Intercom feature
- Conference and transfer capabilities
- Programmable feature buttons
- Automatic registration
- Trouble lights
- Extended battery life
- Battery pack and optional battery backup
- Rapid battery charger (2½ hours)
- Dynamic power adjustment
- Mute button
- Mobility-range test capabilities

## Long-Range mobility

Avaya offers two powerful long-range solutions for contact with customers, coworkers, and suppliers from anywhere in the office. In both systems, overlapping zones enable mobility without changing telephones. (See [Figure 5.](#)) The telephone connection is “handed off” from one transmitter to another, as directed by a single radio controller.

**Figure 5. Long-Range mobility**



The DEFINITY Wireless Business System PWT, when integrated with Avaya IP600, offers inherent efficiencies.

This solution is supported by a patented software product unique to the wireless industry. Avaya’s Wireless System-Engineering Expert-Design System analyzes a building or campus space and determines how the wireless system should be configured. By determining the ideal location for base stations within the structure or structures, this software simplifies one of the most difficult aspects of wireless implementation — ensuring maximum efficiency and lower lifecycle costs.

## DEFINITY Wireless Business System PWT

The DEFINITY Wireless Business System PWT relies on Avaya IP600 to manage mobility. It uses Personal Wireless Telecommunications-based technology, which is a leading protocol in the United States. This protocol permits up to 12 simultaneous conversations per base station and defines the radio interface between the portable telephones and the base stations in the system.

The DEFINITY Wireless Business System PWT is fully integrated with Avaya IP600, offering full access to Avaya IP600 features. This configuration has the following maximum capacities:

- 1500 wireless telephones
- 240 base stations
- 7,000 to 40,000 calls per busy hour (depending on Avaya IP600 configuration)
- 6 million square-foot (360,000 square-meter) coverage area

# 7 Computer Telephony Integration

Computer Telephony Integration (CTI) is the linking of telephone communication systems to personal computers, which can increase productivity and customer satisfaction through the exchange of information between the PC and the telephone. CTI applications integrate data processing, data communications, and voice communications.

Avaya IP600 supports the following types of CTI applications.

- Server-based solutions, which require the Avaya CentreVu Telephony server, the co-resident DEFINITY LAN Gateway (DLG) function (which resides on the Avaya IP600 TN795 Processor circuit pack) and connectivity to DLG via the TN795 Processor circuit pack, or optionally via the TN799C C-LAN board.
- Avaya IP Softphones, which enable users to control telephone calls (both incoming and outgoing) directly from a personal computer.
- [www.messenger](#), which provides quick and easy access to your telephone, fax, and text messages through your Web browser.

## Server-based solutions

Avaya IP600 supports third-party CTI applications via ASAI and TCP/IP links. These CTI links are supported on Avaya IP600 via the DEFINITY LAN Gateway functionality, which is co-resident on the Avaya IP600 Processor board (TN795).

The co-resident DEFINITY LAN Gateway supports one ASAI link. The ASAI link is normally routed from the CTI Server (an external Windows NT or Windows 2000 server) running CentreVu to the TN795 Processor circuit pack via TCP/IP.

For security reasons, the link may be routed from the CTI Server to a C-LAN board (TN799) in the Avaya IP600 system. The C-LAN board requires use of one additional slot in the Avaya IP600 cabinet. The maximum message rate over the ASAI link is 100 messages per second full duplex, regardless of whether the link is going directly to the TN795 Processor board or being routed via a TN799 C-LAN board.

## Third-Party applications

All third-party CTI applications are supported by Avaya IP600, except for those that require adjunct routing. The following is a list of some CTI applications that are currently available. Availability varies by country.

- Intuition

Intuition is designed to be a cost-effective software application providing easier entry into CTI for small Call Center customers. Intuition automates the business process by using sophisticated rules-based intelligence. It “listens” for events such as inbound and outbound calls, Dynamic Data Exchange (DDE), hot key and time-based events, then applies the rules you define. For example, you can define an Intuition Rule that runs a script or opens a spreadsheet when you get a call from a stockbroker.

While Intuition is similar to Sixth Sense, Intuition integrates closely with SoftPhone Agent v.5 and includes the following features:

- ~ Script Recorder for creating scripts by recording user keystrokes
- ~ Simulation for telephony events
- ~ Auto-attendant support for scriptless call handling
- ~ Enhanced User Interface

- FastCall Agent 3.0

FastCall Agent is the next generation of Avaya’s CTI middleware product called FastCall. The new release is designed to offer even easier installation and usability. FastCall Agent provides a broad range of CTI functionality without requiring changes to applications or development of custom software programs.

FastCall Agent resides between the telephone system and computer applications – thus the term “middleware.” This approach allows the agent to enable these applications with inbound and outbound CTI capabilities without computer code changes within the application itself. This provides a great degree of flexibility for companies with multiple departments, particularly when each department has a different application.

In addition, changes to the application do not affect FastCall Agent. FastCall Agent can be reconfigured to adapt to a new application quickly and easily. FastCall Agent “screen pops” populate a call center agent’s Windows-based application screen based on the calling number (ANI), called number (DID, DNIS, ACD group, or other telephone system identifier), or the caller’s touch tone input as the incoming call is received. These applications could include databases, help desk packages, sales force automation programs, personal information managers (PIMs), contact managers, word processors, spreadsheets, customized inquiry systems, or a combination of these applications.

# 8 Enterprise Class IP Solutions

The capabilities and applications of Avaya IP600 are extended with the introduction of Enterprise Class IP Solutions (ECLIPS). ECLIPS supports audio/voice over a LAN or WAN, and it ensures that remote workers have access to communication system features from their PCs.

Although voice quality can and will vary based on LAN conditions, Avaya IP600 offers features that enable management of the quality of voice communications. Avaya IP600 supports 3 Quality of Service methods:

- IP standard, Differentiated Services (DiffServ) — sets Type-of-Service (TOS) in IP header of voice packets
- Ethernet standard 802.1 p/q — sets priority level in the layer 2 Ethernet packet header
- UDP port range administration — ports largely dedicated for voice packets

IP trunk bypass to PSTN trunk is also supported with administrable thresholds for latency and packet loss.

This release also introduces hairpin and IP-IP direct connections, two features that make voice communications more efficient. These features increase the efficiency of voice communications by reducing both per port costs and IP bandwidth usage. Hairpin connections reroute the voice channel connecting two IP endpoints, so that the voice goes through the media processor board in IP format, thereby bypassing the DEFINITY TDM bus. IP-IP direct connections route the voice channel connecting two IP endpoints by sending the voice directly through the LAN or WAN between the two endpoints, instead of carrying a mixed connection of IP signaling and TDM bus signaling.

To maximize voice quality using ECLIPS, you must consider both your hardware and network configurations. For example, with IP Softphones, you can send the audio over traditional circuit switch lines, providing high quality voice, or over IP using LAN connections. When making calls over the LAN, voice quality will vary and may be unacceptable if you don't tune your computers and data network.

Avaya IP600 supports a trunk configuration, four types of softphones, and three models of IP telephone, using the TN2302AP, which is an IP-media processor circuit pack. The TN2302AP IP media processor provides H.323 trunk connections and H.323 voice processing for IP telephones. The features that use the TN2302AP also require the TN799C C-LAN circuit pack.

## Trunks

ECLIPS support the TN2302AP IP media processor, which enables H.323 trunk service using IP connectivity between two DEFINITY systems. The H.323 trunk groups can be configured as DEFINITY-specific tie trunks, generic tie trunks, or direct-inward-dial (DID) “public” trunks. In addition, the H.323 trunks support ISDN features such as QSIG and BSR.

Up to 64 Avaya IP600s can be networked through DCS with full feature transparency. More than 1000 Avaya IP600s can be networked with near full feature transparency. Avaya IP600s network seamlessly with DEFINITY systems.

## Avaya IP Softphones

Avaya IP Softphones operate on a PC equipped with Microsoft Windows 95/98/NT and with TCP/IP connectivity to the Avaya IP600. Avaya IP600 used with Avaya IP Softphones offers enhancements to information display, security, and serviceability. For example, the administrator can obtain information about the Avaya IP Softphone connection type and can list registered IP stations.

Release 2 introduces the improved audio quality via i-Clarity and a call-bar option for the visual graphical user interface (phone picture is already available). A lightweight directory access protocol (LDAP) client allows access to LDAP-compliant databases. The Avaya IP Softphone Release 2 is CTI/TAPI compliant.

Multiple call appearances, conference, transfer, hold, mute, redial, and volume control are provided by the Avaya IP Softphone. Access to DEFINITY station features is standard. Multiple audio voice codecs are supported as well as multilanguage.

ECLIPS supports the following four softphone applications: Road Warrior, Telecommuter, CentreVu IP Agent, and Native H.323.

### Road Warrior

Enables use of the full feature set from temporary remote locations anywhere in the world. The road-warrior application consists of two software applications running on a PC that is connected to the Avaya IP600 over an IP network. The single network connection between the PC and Avaya IP600 carries two channels, one for the signaling path and one for the voice path. On the Avaya IP600, the road-warrior application requires the TN799C C-LAN circuit pack for signaling and the TN2302AP IP Media Processor for voice processing.

### Telecommuter

Enables telecommuters to use the full Avaya IP600 feature set from home. It consists of a PC and a telephone with separate connections to Avaya IP600. The PC provides the signaling path and the user interface for call control. A standard telephone provides a high-quality voice path. The telecommuter application requires the TN799C C-LAN circuit pack for signaling. The telecommuter application does not use the TN2302AP IP Media Processor.

## CentreVu IP Agent

Provides a variation of the telecommuter application. CentreVu IP Agent emulates an Avaya set and provides use of the call end capabilities required for Call Center operations from a remote location, such as the agent's home.

## Native H.323

This is an IP-connected softphone running off-the-shelf H.323 software. It operates as a single-line phone with limited features, which are activated by Feature Access Codes.

## Avaya IP Telephones

The 4600-series Avaya IP telephones use the IP technology with Ethernet line interfaces and downloadable firmware. These telephones emulate DCP 6400-series telephones and provide all of the same features except for the group listen speakerphone feature. The first release of the 4600-series Avaya IP telephones uses the dual connection architecture to register and communicate with the Avaya IP600 switch. This series of telephones includes the 4606, 4612, and 4624 models.

## Avaya R300 Remote Office Communicator

Avaya R300 Remote Office Communicator (Avaya R300) is a product that acts like a simple switch at the remote site to connect remote stations and local access trunks. The Avaya R300 unit supports VOIP and DCP, as well as analog line and trunk connections. In addition, each Avaya R300 unit supports 12 remote dial access data channels. An Avaya IP600 switch can support up to 16 Avaya R300 units.



# 9 Telecommuting/Virtual Office

Avaya's research and independent industry studies show that telecommuters are generally 15 to 30 percent more productive. Telecommuters convert travel time into productive work time, are less likely to be distracted by normal office routines, and frequently end up working longer hours with greater output. During severe weather, telecommuters can continue working while others are stuck at home without access to work-related systems and tools.

Special Avaya IP600 system modules are available for telecommuting workers. In addition, many standard Avaya IP600 and voice messaging features are effective for telecommuters.

## Avaya IP600 features for telecommuting

Avaya IP600 includes several features that make telecommuting more convenient. See "[Avaya IP Softphones](#)" on page 66

### Remote Call Coverage/ Call Forwarding Off-Net/Coverage of Calls Redirected Off-Net

Remote Call Coverage and Call Forwarding Off-Net allow calls to be redirected to a remote location. This allows calls that are placed to your office telephone number to be redirected to your home office. If not answered, you can administer the system to monitor calls and retrieve them for additional processing, or leave calls at the remote location. There is a one-second delay before the caller connects to the remote telephone.

### Extended User Administration of Redirected Calls (Telecommuting Access)

Extended User Administration of Redirected Calls (also called Telecommuting Access) allows you to change the active call coverage path or forwarding extension from any on-site or off-site location. This feature allows you to change the path or extension from your home office.

### Personal Station Access

Personal Station Access allows you to transfer telephone station preferences and permissions to any other compatible telephone. Preferences can include the definition of terminal buttons, abbreviated dial lists, and Class of Service and Class of Restrictions permissions. It can be used on-site or off-site using DEFINITY Extender. This feature has several telecommuting applications. For example, several telecommuting employees can share an office on different days of the week.

## Station Security Codes

Station Security Codes protect access to telephone stations and can be changed by the telephone users. This feature allows you to ensure protection of your console features.

All of these features are described in detail in *DEFINITY Enterprise Communications Server Release 9 Administrator's Guide*, (555-230-506), which is on the documentation CD under the following feature names:

- Call Coverage
- Call Forwarding
- Extended User Administration of Redirected Calls
- Personal Station Access
- Station Security Codes

## Pipeline 15

The Ascend Pipeline 15 is an Integrated Services Digital Network-Basic Rate Interface (ISDN-BRI) terminal adapter that provides single user access to remote services, such as corporate headquarters, intranet, or the internet over an ISDN-BRI line. The Pipeline 15 supports high-speed digital connections while simultaneously offering two analog ports for sharing the ISDN-BRI line with analog devices such as a telephone, fax machine, answering machine, and/or modem. By combining separate transmission services over a single line, the Pipeline15 allows users to consolidate billing and achieve superior consolidated performance.

Installing and configuring the PipeLine 15 is easy. The Pipeline 15 connects to an IBM-compatible PC, Macintosh, or UNIX workstation via an RS-232 serial cable and has a powerful graphical user interface that lets users set up and configure their unit in less than 15 minutes.

The Pipeline 15 supports integrated Multilink PPP, Multilink Protocol Plus, and Bandwidth Allocation Control Protocol, which will save users money each year by dynamically adding and subtracting bandwidth based on need. The Pipeline 15 also supports caller line ID devices on its two analog ports and advanced analog calling features such as hold, drop, conference, and transfer.

Additionally, a comprehensive series of Pipeline and SuperPipe access routers are available.

## DEFINITY Extender

DEFINITY Extender is a single-box remote voice and data solution for telecommuters, remote agents, and branch offices using Avaya IP600. DEFINITY Extender helps increase the productivity and performance of remote workers by allowing them to access the features of the Avaya IP600 system and their corporate LAN. With the DEFINITY Extender, remote voice access is just as simple as remote data access for off-premises employees.

The DEFINITY Extender product family provides off-site employees with all of the features of their Avaya IP600 system, no matter where they are located, over analog or ISDN-BRI connections. A switch module located at the Avaya IP600 location and a remote module located at the off-premises location are all you need to provide an off-premises employee with full voice and data communications functionality.

## AUDIX features for telecommuting

The following AUDIX features are useful for telecommuting:

- Multiple Personal Greetings allow subscribers to prepare a pool of up to 9 personal greetings to save time and provide personalized customer service. Separate messages can indicate the subscriber is on the telephone, away from the desk, or on vacation.

**Note:** Multiple Personal Greetings only works in a centralized environment. With the Mode Codes interface, you cannot set up separate internal and external greetings.

- Outcalling automatically dials a prearranged telephone number or pager when messages are received in a user's mailbox. The system tells whoever answers that messages have been received and allows them to log in to the AUDIX system.
- Priority Outcalling provides outcalling notification of priority messages only. This allows the telecommuter to be relatively undisturbed by notifications of messages that do not require immediate attention.
- Call Answering for Nonresident Subscribers provides AUDIX system mailboxes for remote users who do not have a telephone but do have an extension number on Avaya IP600.

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



# 10 System Administration

Avaya IP600 offers a variety of modular tools for managing your system.

Telephone and facility administration features allow you to administer telephones, computers, facilities, and features throughout your system or network. Traffic management features allow you to measure, manage, and report on the voice and data communications traffic throughout your system or network. Maintenance features allow you to view the health of your system and perform maintenance procedures on your own system.

Avaya's broad system management philosophy extends Avaya IP600's power and flexibility into the tools for managing the system. These tools are based on the user-friendly architecture that is the hallmark of Avaya's DEFINITY products.

## DEFINITY Site Administration

Avaya IP600 applications are pre-loaded on the hardware platform. The actual set up of customer translations are administered through DEFINITY Site Administration (DSA), which is integrated into the hardware platform. DSA is a general-purpose system management tool that simplifies basic administration. With this application, users can easily navigate, display, add, modify, and/or remove the Avaya IP600 system and related objects. The standard SAT (system administration terminal) interface is still available for use through SAT emulation.

DSA streamlines common system administration tasks by providing:

- Shortcuts to administration commands
- The ability to schedule tasks to run at a later date
- The ability to print button labels
- The ability to easily create AUDIX subscribers with either a default mailbox or a custom mailbox

DSA provides a Windows 32-bit graphical user interface and runs on Windows 95, Windows 98, and Windows NT 4.0 or later. Designed to support DEFINITY AUDIX and INTUITY AUDIX systems, DSA requires an active Avaya IP600 or AUDIX connection for proper operation.

DSA provides the following functionality:

- Browser

The Browser provides navigation and access to features and services. The user creates hosts and related data objects and accesses the Avaya IP600 and/or AUDIX hosts from the Browser. The Browser is based on a standard tree view and forms the central user interface component in DSA.

- Emulation

DSA's emulation support includes AT&T 4410 and provides the most basic form of system administration.

- Graphically Enhanced DEFINITY Interface (GEDI)

The GEDI feature provides users with a Windows-like interface to:

- ~ Add objects
- ~ Remove objects
- ~ Change objects
- ~ View the status of objects
- ~ Duplicate objects
- ~ Test objects
- ~ Generate tasks that may be scheduled to run at a later date and time

- Scheduler

The Scheduler lets users specify a task to run at a specific date and time. A task is a collection of one or more operations that users specify to run at a predetermined time. Tasks are generated from either the Graphically Enhanced DEFINITY Interface, the Add User Wizard, or Call Accounting Data Export.

- Event Log

The Event Log allows users to view the results of running and completed tasks.

- Job Viewer

The Job Viewer allows users to view the task or job status while it is being executed. The Job Viewer also shows the queue of jobs to be run.

- Button Label Printing

The Button Label Printing feature lets users print button labels for the handsets using a standard Windows laser printer. This feature also provides a graphical print preview. The Button Label Printing feature supports printing multiple labels of the same type.

- Add User Wizard

The Add User Wizard assists in creating station and subscriber details by automatically providing help such as available extensions and ports and allowing users to base the creation on an existing template.

- Call Accounting Data Export

The Call Accounting Data Export feature lets users export information on stations, trunks, agent login identification, Authorization Codes, and trunk circuits from the Avaya IP600 system to share with any third party call accounting program that supports DSA.

- Import/Export Capability

DSA provides easy graphical exporting and importing of agent login, coverage paths, hunt groups, data modules, stations, trunk groups, and VDNs. Users can export data fields to databases such as Microsoft Excel. Users can then change the data, import the data back into DSA, and then resend the data to the Avaya IP600 system. The import/export capability can also assist users in creating corporate directories and custom reports.

- Global Change Capability

The global change capability lets users select and change field values in one or more of the following objects that matches a search filter:

- ~ Agent login ID
- ~ Coverage path
- ~ Data module
- ~ Hunt group
- ~ Station
- ~ Trunk group
- ~ VDN

- Create Station Templates Wizard

The Create Station Templates wizard steps users through instructions on how to create station templates.

- Add Bridged Appearances Wizard

The Add Bridged Appearances wizard steps users through instructions on how to add bridged appearances to telephones.

- Out-of-Service Trunks

The Out-of-Service Trunks feature creates a task that checks periodically for out-of-service trunks. If an out-of-service trunk is found, the users are notified either in the DSA message box or by email.

- Reports

DEFINITY Site Administration provides the following reports:

- ~ Browse Dial Ranges lets users quickly and easily view the complete dial ranges in the Avaya IP600 system.
- ~ Browse Stations lets users quickly view all assigned stations in the Avaya IP600 system.
- ~ Browse Unused Ports lets users view the available ports in the system.
- ~ Find Unused Extension lets users view unused and available extensions.

## Administration

Avaya IP600 includes features that simplify and accelerate the administration process.

### Portless Administration/Administration Without Hardware

The Administration Without Hardware feature offers the capability to administer station forms without specifying a port location. Administered stations will not cause alarms or errors when the station is translated but not yet installed. These station types are referred to as “phantom” stations. Phantom extensions are used for Automatic Call Distribution Dialed-Number Identification Service (ACD-DNIS). This feature allows a phantom extension to be administered on the switch for each call type that needs to be identified to agents. The phantom ACD extension is either “call forwarded” (via an attendant console) to an ACD split or has its coverage path defined to include the ACD split. The name field administered for the phantom extension will identify to the ACD agent which service the caller is attempting to reach, allowing the agent to properly address the caller.

The Administration Without Hardware feature 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 switch.

The Administration Without Hardware feature 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.

The Administration Without Hardware feature can be used on the following telephone types:

- Analog telephones
- Digital Communications Protocol (DCP) telephones
- Hybrid telephones

Avaya IP600’s configurations support telephone types in addition to those listed above. These include:

- Attendant consoles
- Voice/computers (such as DCP telephones with voice and data capabilities)
- Data modules
- Analog queue warning ports
- Announcement circuit packs

## Automatic Station Relocation/Terminal Translation Initialization

Terminal Translation Initialization (TTI) is a feature that works with the Administration Without Hardware feature. TTI is part of the Portless Administration/Administration Without Hardware feature, but can also be a stand-alone feature. TTI associates the terminal translation data with a specific port location through the entry of a special feature-access code, a TTI security code, and an extension number from a telephone that is connected to a wired — but untranslated — jack.

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

TTI reduces 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 to a specific port.

System administrators maintain control over the use of TTI through security codes. TTI and when they use it.

## Scheduling

Avaya IP600's functional scheduling allows you to specify the time a command will be executed or specify that it should be executed on a periodic basis. Only commands that do not require user interaction after being entered on the command line (such as list, display, test) can be scheduled.

Avaya IP600 also supports scheduling of "one shot" requests — commands that are executed only once and are then removed from the scheduling queue automatically by the feature, such as save translation commands.

Functional scheduling enhances administration. For example, scheduling of save translations is particularly important when large numbers of translation changes are made during the day and ensures they will be saved to tape at the specified time. The "one shot" report is particularly useful for scheduling large print jobs at night that are normally run only once.

## Basic reporting

Avaya IP600 has built-in capabilities for generating reports. 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. Avaya IP600 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 switches. These enhancements allow your system administrator to restrict data reporting to only the desired amount of switch parameters.

Avaya IP600 also includes the following reports:

- The 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.
- The 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.
- The 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.

## Performance measurements

A number of performance measurements are available on Avaya IP600. These measurements are available in the form of switch-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:

- Socket; DSP; packet loss reports for managing Voice over IP
- Call Coverage reports
- Coverage Points

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

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

- The Traffic Summary report offers additional measurements that help configure the switch, determine the switch's capacity for growth, and report unauthorized switch-access attempts.

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

- Attendant Position report
- Security Violations report
- Tandem Traffic report

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
- Automatic Route Selection Pattern Measurements
- Trunk Group Detailed Measurements

The following measurements and reports are needed for engineering and load balancing a large switch. These measurements include:

- Blockage Study report
- Port Network and Link Usage

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

## ECS Reports Generator

The ECS Reports Generator is an easy-to-use, graphical reporting tool that does the following:

- Maintains a location database of all the systems managed (in addition to Avaya IP600, it supports DEFINITY Enterprise Communication Systems)
- Provides automated connections via predefined scripts to the various systems
- Captures all predefined reports immediately, or schedules off-peak downloading to your personal computer
- Creates faxable order forms and keeps a record of all purchases for all systems in the network
- Provides cut-through administration capability with a 513 terminal emulator
- Provides flexible sorting and formatting options for report display and export to other applications
- Provides an easy-to-navigate interface, with simple setup procedures

The scheduler can be used for off-peak, automatic polling of systems for daily reports required for monitoring your Avaya IP600 environment. It can also be set up to invoke special scripts or personal computer applications.

The ECS Reports Generator produces all standard reports, plus the following:

- The Unused Extension Report shows all unused extensions.
- The Configuration Pictorial graphically depicts your system, with cabinet, carrier, and slot representation. It maps the station data to the configuration data so you can easily determine where stations are assigned for a port on a circuit pack. You can easily see which ports are free on which slots and what the port names are.
- The Configuration Summary provides a total system inventory with totals of circuit packs in use and the total number of free ports. It also recommends ways to consolidate and conserve resources.
- The Station Reports allow you to sort station data in a variety of columns.
- The Phone Directory allows you to create and maintain a directory list for general distribution. You can define some extensions as unlisted, and they will not be printed in the directory.
- The Out of Service Trunks report notifies you during off-peak hours of any trunks that are not functioning.

All of these reports can export data formatted for use by other database management applications.

## Call Charge information

Avaya IP600 provides two ways to know the approximate charge for outgoing calls:

- Advice of Charge — For ISDN trunks

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

- Periodic Pulse Metering — For non-ISDN trunks

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

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

**Note:** This is unavailable in some countries. Please check with your Account Executive or local distributor for availability in your country.

# Call Detail Recording

Call Detail Recording (CDR) helps you manage call costs by letting you monitor and analyze call patterns and usage in your system.

## Call Detail Recording features

Avaya IP600 Release 9 CDR includes the following capabilities:

- Distinguish voice from data on trunk calls
- Determine if a data call used a conversion resource, such as a modem pool
- Choose whether to record the vector directory number in the “Dialed Number” field of the CDR record, or record either the split or the agent extension in the same field
- Allow CDR records to be generated for internal calls (calls to and from a set of extensions, including data endpoints) so administered (a maximum of 500 extensions in large configurations)
- With Call Privacy, allow up to seven digits of the dialed number to be blanked from the CDR record
- Provide CDR 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

## Variable format records

Avaya IP600 provides many different selectable formats. This offers a flexible means of incorporating new fields in the call detail record as new switch features and new CDR devices become available. The variable format allows you to define a record in terms of its content (from a set of available data elements), the position of its fields, and the spacing between the fields. This method can be used to construct the 15-, 18-, and 24-word standard formats and custom formats.

If calls come in while the CDR link is down and the buffer is filled to maximum, Avaya IP600 gives you the following administrable call-record handling options:

- Block the calls with reorder
- Allow the calls to overwrite records
- Route the calls to an attendant with the option to proceed as a Non-Call Detail Recording call

## Call Detail Recording devices

There is no RS-232 interface provided by Avaya IP600 for LAN. Your CDR Output Records are stored in D:\Lucent Data\CDR in files Cas.in and Cdr.out.

### Call Accounting Systems

Several options are available to you for call accounting, depending on what type of system administration tools you are using.

### DEFINITY Network Telemangement (DNT)

DEFINITY Network Telemangement (DNT) is a multi-user telemangement system. See “[Call Accounting Systems](#)” in [Chapter 3, Adjuncts](#) for more information.

### Call Accounting System for Windows (CFW)

The Call Accounting System for Windows allows you to generate comprehensive and accurate accounting reports using the Microsoft Windows environment. See “[Call Accounting Systems](#)” in [Chapter 3, Adjuncts](#), for more information.

### Call Accounting System XP

The Call Accounting System XP has functionality equivalent to DEFINITY Network Telemangement, but only offers call accounting. Call Accounting System XP is a LAN/WAN-based application that supports customers with multi-user requirements.

### Call Accounting System Terminal

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

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

### INTUITY Call Accounting System

If you are using any of the INTUITY voice messaging products, the INTUITY Call Accounting System is probably the most effective call accounting solution.

## Security

Besides the toll-fraud detection options available with the Call Accounting Systems described in the previous section, Avaya IP600 includes many other security features, some of which are an integral part of the system design.

## Call Restrictions

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

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

Avaya IP600 provides powerful voice and data capabilities and connections to a variety of voice and data networks. Avaya IP600 builds on Avaya's established networking strengths to offer you network-management features, network interfaces, a variety of private-network configurations, and end-to-end Integrated Software Defined Network (ISDN) capabilities. Avaya's leadership in developing and supporting open international networking standards is also apparent in Avaya IP600's compatibility with the QSIG global standards.

**Note:** Some applications or products are unavailable in some countries. Please check with your local distributor for further information.

## Uniform Dial Plan

Uniform Dial Plan provides a common 4- or 5-digit dial plan that can be shared among a group of private-network switches. Interswitch and intraswitch dialing both require 4- or 5-digit dialing. This feature is used with either:

- An electronic tandem network (ETN)
- A main/satellite/tributary configuration and Distributed Communications Systems (DCS).

In addition, it can provide uniform 4- or 5-digit dialing between 2 or more private-switching systems without ETN, main, satellite, and tributary switches, or DCS.

With Uniform Dial Plan (UDP), a unique 4- or 5-digit number is assigned to each station in the network. A unique number (private-network location code plus extension) can be used at any location in the ETN to access that station. Avaya IP600 enhances the standard uniform dial plan with the unrestricted 5-digit uniform dial plan, which allows up to five digits to be parsed for call routing.

## Distributed Communication System — Integrated SDN and Non-Integrated SDN

For a multilocation company that requires several systems, DCS may be the answer. DCS is an arrangement of private-network switches, referred to as nodes. The maximum number of nodes that can be in a DCS complex varies from 20 to 63, depending on the particular configuration of switches. DCS nodes can be physically located in the same building, spread across a campus, or scattered across the country or around the world. Digital trunks interconnect the switches that serve the DCS complex. The links connecting a DCS may also be provided via a Virtual Private Network (VPN).

The functions and features of DCS are made possible by the use of an advanced interprocessor data link connecting each switch, allowing call-processing information to be passed from one switch to another. The data link supplies selected feature transparency and efficient utilization of shared facilities.

Feature transparency means that features work the same from a user's perspective, whether the telephones involved are assigned to the same switch or to different switches. Users in a DCS can dial each other with four or five digits as if they were all on the same switch.

Here are some examples of feature transparency in a DCS:

- Leave Word Calling (LWC) allows you to press a button on your telephone and leave a standard "call me" message with your name and phone number. When your Avaya IP600 is linked with other switches in a DCS, you can call any employee in the DCS complex and press the LWC button to automatically leave a standard message.
- Calling-Party Name Display — If your telephone is equipped with a digital display, information about the person calling you is displayed before you pick up the receiver. With DCS you know who is calling and whether that person is in a nearby building or across the country.
- Centralized Messaging services for an entire DCS complex (subnetwork) may be coordinated by one system, depending on the traffic volumes and versions of the main and remote switches. This means that switches with smaller messaging requirements do not share a voice messaging system with another switch.

Avaya IP600's DCS features DCS over Integrated Software Defined Network-Primary Rate Interface (ISDN-PRI) with path replacement for optimizing trunks. Thus, when you transfer out of your AUDIX voice messaging system, for example, Avaya IP600 sets up a new path that optimizes system resources.

## Distributed Communications System and ISDN

DCS nodes are connected by digital trunks (for example, using DS1 or ISDN-PRI facilities). Avaya IP600 can send DCS messages over ISDN-PRI D channels. As a result, you are not limited to private or leased facilities between your various locations. You can also use public-network services. (See [Figure 6](#).)

The SDN supports every DCS transparency except the following:

- DCS attendant control of trunk group access
- DCS attendant direct trunk group selection
- DCS busy verification of terminals

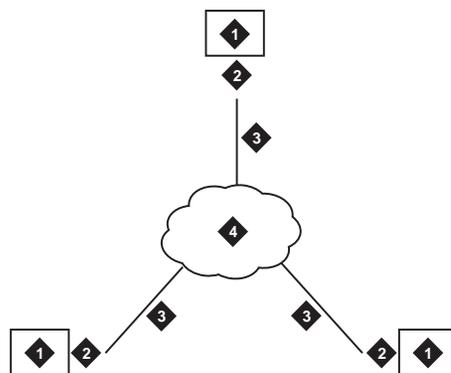
All other capabilities and limitations associated with the DCS still apply.

AUDIX systems networked via DCS can also be supported over ISDN-PRI. (See [Chapter 4, Intuity AUDIX Messaging](#), for more information.)

## Avaya IP600 DCS networks

If your company has two or more sites with Avaya IP600, you can network them using the DCS over ISDN-PRI feature (DCS+). This requires a system to use ISDN-PRI signaling. The network connections can be either ISDN-PRI or DS1 private-network dedicated facilities. Figure 6 shows a network using ISDN-PRI signaling.

**Figure 6. A Network Using DCS with ISDN-PRI**



- |                          |   |
|--------------------------|---|
| 1 Avaya IP600 switch     | 3 Transmission via ISDN-PRI or Private Network T1/E1 Facilities |
| 2 Signaling via ISDN-PRI | 4 Public or Private Network                                     |

## QSIG global networking

Avaya IP600 is a pioneer in providing compatibility with the QSIG global networking protocol. This means that you can connect Avaya IP600 with other switches throughout the world. Avaya developed the QSIG Global Networking feature to comply with the QSIG standards developed by the European Computer Manufacturer's Association and the International Standardization Organization. It supports the ISDN-PRI connection from switch to switch as long as both switches support the same protocol.

Avaya's implementation of QSIG features the Name Identification supplementary service and the Call Forwarding and Call Transfer features. QSIG enables the system to move calls from their original paths to new paths that cost less or use resources more efficiently. New paths can be set up after a call has been transferred or after a call has been forwarded using the Diversion with Rerouting feature. Avaya IP600's implementation of QSIG also supports the ISO QSIG private network diversion supplementary service, as described in the QSIG standard.

## World-Class Routing

Avaya IP600 is a world-class system that meets the needs of global customers. One capability essential in meeting these needs is the ability for users to flexibly dial any location in the world, regardless of the dial plan used at that location. To fulfill this requirement, Avaya IP600 provides World-Class Routing.

World-Class Routing is a powerful enhancement to Avaya IP600'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 (ARS) 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 (AAR) 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 ARS and AAR as separate features, but through generalized administration applicable to both features, provides both with 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 Avaya IP600 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. Avaya IP600's support for International Direct Distance Dialed calls eliminates any restriction on the grouping and format of digits on ARS 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 ARS, into AAR 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

Avaya IP600 has a variety of features that enable you to manage your network resources effectively. Here are just a few examples of Avaya IP600 features that can be used to manage your network:

- Time of Day Routing
- Automatic Route Selection (ARS)
- Automatic Alternate Routing (AAR)
- Additional Network Feature Path Replacement
- Subnetwork Trunking
- Generalized Route Selection
- Facility Restriction Level
- Bearer Capacity Class
- Remote Network Access
- Public Network Call Priority
- Authorization Codes

### 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 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

ARS routes public network calls on the most desirable (usually the most economical) trunking facilities available on your Avaya IP600 when the call destinations are accessible through your public network.

Avaya IP600 supports up to 254 routing patterns. Each routing pattern consists of up to 6 routing preferences (types of facilities) set up in the order 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 system checks the routing preferences in that pattern in the order they were listed, and the first available facility is used to place the call. If a facility is not available, the call can be queued until a facility becomes available.

## Automatic Alternate Routing

AAR 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 ARS, you set up various patterns for routing calls — in this case, with the private network. Depending on your Avaya IP600's configuration, you can have up to 254 routing patterns. Each pattern includes a primary preference — the most preferred and direct route — and 5 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.

## Generalized Route Selection

Generalized Route Selection gives you the capability to not only select the optimal call routing based on the dialed number, but also to select the appropriate facility based on the type of call. Generalized Route Selection enhances ARS and AAR 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. Avaya IP600 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. Avaya IP600 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, Avaya IP600 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 system administration, Avaya IP600's dial plan can be independent of the type of call being dialed. This flexibility makes life easier for your system 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 system management tools. Avaya IP600 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 (equal to or higher).

Avaya IP600 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.

## Authorization codes

Authorization codes are used on particular calls to temporarily raise a telephone's Facility Restriction Level. This feature is useful for those who make calls from telephones other than their own or from outside the network. If a call you dial is blocked because the telephone's Facility Restriction Level is too low, you can enter your authorization code. If the Facility Restriction Level associated with the authorization code is equal to or higher than the Facility Restriction Level of the trunk facilities required to place the call, the call is then completed. Up to 5000 different authorization codes will be in effect for your system at any one time. Using Avaya IP600's system management tools, you can assign authorization codes and change their associated Facility Restriction Level and network access permissions.

## Network interfaces and equipment

Avaya IP600 supports a variety of interfaces to voice and data networks. Trunks supply links between Avaya IP600, the public network, and other switches. DS1 interfaces offer high-speed digital connectivity between switches.

### Trunk group circuits

Trunks provide the communications links between Avaya IP600 and other switches, 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 Avaya IP600 via port circuit packs. Avaya IP600's trunk group circuit types include the following:

#### Local exchange trunks

Local exchange trunks connect Avaya IP600 to a central office. The following are some of the types available:

- Central office trunks, which connect Avaya IP600 to the local central office for incoming and outgoing calls
- Foreign exchange trunks, which connect Avaya IP600 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, 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 (DID) trunks, which connect Avaya IP600 to the local central office for incoming calls dialed directly to stations without attendant assistance
- Digital Service 1 (DS1) trunks, which can be used to provide ISDN Primary Rate Interface local exchange trunk services. DS1 by itself can be used to provide local exchange trunk services

#### Tie trunks

Tie trunks carry communications between Avaya IP600 and other switches 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 with the switch. 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** Miscellaneous trunks perform functions that do not fit neatly into any of those already described:
- Release-link trunks are used between switch locations to provide Centralized Attendant Service.
  - Remote-access trunks provide off-premises users with access to Avaya IP600's features and networking.

## Digital interfaces

Avaya supports both T1 and E1 facilities. As industry standards around the world, T1 and E1 facilities provide the latest alternative to analog trunking.

**E1 interface** Avaya IP600 supports E1 connections. T1/E1 access and conversion allows simultaneous connection to both T1 (1.544 Mbps) and E1 (2.048 Mbps) facilities (using separate circuit packs).

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

Used to connect switches to the public network or to other switches in a private network, DS1 also delivers high-speed, end-to-end digital connectivity. Voice and data calls are completed at transmission speeds of up to 64 Kbps.

Avaya IP600 offers several options in supporting the DS1 interface. The options include support for voice-grade DS1, alternate voice/data, and Digital Multiplexed Interface. The voice-grade DS1 interface is a T1 D4 channel-bank-compatible interface.

## ISDN

Avaya IP600 provides complete Integrated Services Digital Network (ISDN) support. ISDN eliminates the need for multiple, separate access arrangements for voice, data, facsimile, and video services and networks. Using inexpensive twisted copper, 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 7-layer Open Systems Interconnect Reference Model of the International Standards Organization.

Avaya IP600 supports the ISDN-PRI, which is used for connecting premises equipment such as switches to the network and acts as a powerful interface between intelligent equipment such as switches and computers. However, trunk-side Basic Rate Interface (BRI) is supported in countries that support the Euro-ISDN (ETSI) standards.

## Centralized Attendant Service

Avaya IP600 owners who have more than one switch location can benefit greatly by using the Centralized Attendant Service (CAS) feature. CAS reduces the number of required attendants, and, in most cases, all those attendants can be located at one of the switch locations, called “main.” Switches at the other locations, called “branches,” redirect their calls to the CAS main. Thus, a company can have a centralized attendant group at the headquarters office and can handle calls from there for the branch offices.

All locations in a CAS arrangement have a listed directory number. Calls to a branch listed directory number terminate at the main location, even if the branch location has an attendant. These listed directory number calls are routed to the centralized attendant group over trunk circuits called release-link trunks or over QSIG trunks. These release-link trunks are used only for centralized attendant calls and signaling.

After a call is processed by the centralized attendant, it can be extended back to the branch location. The release-link trunk is then dropped and made available for other calls to the centralized attendant.

If an Avaya IP600 is a node within a DCS and CAS is provided, a centralized attendant can do the following:

- Control access to specific trunks at other nodes
- Directly access specific trunks at another location
- Place test calls to telephones and trunk groups at other nodes
- Receive a visual warning that all trunks in a remote trunk group are busy or that the number of busy trunks in a remote group has reached a specified level

This feature ensures that all calls directed to an attendant at your company are handled efficiently.

## Main/Satellite/Tributary

If you have modest network requirements, a main/satellite/tributary configuration is an attractive possibility for private networking. In this configuration, one Avaya IP600 location is the main, and remote switches are satellites or tributaries. Attendant positions and public network facilities are usually concentrated at the main.

All calls to or from a satellite pass through the switch at the main. The system appears to be a single switch with one listed directory number. A uniform dial plan provides a common 4-digit or 5-digit dial plan for a main/satellite configuration.

A tributary is similar to a satellite, but it has one or more attendant positions and its own listed directory number. Calls to its listed directory number go directly to the tributary.

The switches in a main/satellite/tributary network are connected by tie trunks. Trunks and switching facilities can be added as requirements grow.

An important Avaya IP600 networking feature is Main/Satellite Extended Trunk Access. Extended Trunk Access allows dialed digits that are undefined at a satellite or tributary switch to be routed over a trunk group to a main switch for interpretation. This flexibility means changes to the network numbering plan do not have to be propagated to all switches. Extended Trunk Access improves your control and reduces administration costs by making trunk networks considerably easier to maintain.

## Electronic Tandem Network

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

An ETN consists of tandem switches, inter-tandem tie trunks that interconnect them, access or bypass trunks from tandem switches to main switches, and the software and equipment to support call routing over the trunking facilities. Different ETN 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 ETN locations.

An ETN can be configured hierarchically. An ETN can connect individual switches; it can also connect other private networks (such as Main/Satellite/Tributary networks) together.

Within an ETN, 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 ETN, a user dials a feature access code for the Automatic Alternate Routing feature plus the 7-digit number, for a total of eight digits.

# 12 SNMP Native Agent Software

Avaya IP600 includes a SNMP (Simple Network Management Protocol) Native Agent. Native Agent provides a SNMP interface to the system's alarm and error tables, select performance measurements, and select configuration data. Native Agent also supports SNMP traps for Avaya IP600 alarms and restarts, INTUITY alarms, and Windows NT events.

DEFINITY Network Management (DNM), Avaya's SNMP-based DEFINITY fault and performance management product, also supports Avaya IP600. DNM runs on NT and UNIX workstations, and collects fault, performance, and configuration data from switches via the DEFINITY Proxy Agent. DNM receives Avaya IP600 data by sending SNMP requests to Avaya IP600's SNMP Native Agent.

Users can integrate DNM with a Network Management System (NMS), either HP OpenView or Tivoli NetView. This allows users to manage their Avaya IP600s and data networks from a central location. The NMS catches traps that Native Agents send, colors the Avaya IP600 icons according to the trap's severity, and records the traps in the NMS trap log. A network manager can use the NMS to copy and move the icons created for different Avaya IP600 devices to specific OpenView or NetView maps of their choosing. The network manager can then look at the Avaya IP600 data that the Native Agent provides by using the NMS's MIB browser.

## Agent Administration

All SNMP Agent Administration can be performed from the command line and Web server. An administrator specifies the community string that Agent uses for authentication. However, Agent must be restarted for community string administration changes to take effect. In addition, an administrator can specify the MIB access permissions and whether traps can be received for up to 50 different network managers (IP addresses). Again, Agent must be restarted for network manager administration changes to take effect.

## Avaya IP600 Data

The Avaya IP600 SNMP Native Agent provides an MIB interface to all the configuration and fault data that the DEFINITY Proxy Agent provides for DEFINITY ECS devices (except, of course, for data that does not apply to Avaya IP600). The supported data is as follows:

### Via the SNMP MIB

Agent provides Avaya IP600 version information for the active switch processing element (including memory resident software version, update identifier, and update state). It also supports retrieval of Avaya IP600 alarms, errors, and restarts.

## Via the SNMP

Agent allows retrieval of Avaya IP600 status data, system time, trunk group information, board data, DS1 board data, ATM board data, port data, station data, and data modules. It provides retrieval of information regarding trunk outage data, lightly-used trunks, long and short trunk holding times, and trunk group performance measurements. In addition, it provides tables of the Avaya IP600's external devices, of the trunks in a trunk group, tables allowing access to the contents of the Avaya IP600's bulletin board, and tables of the Avaya IP600's signaling groups.

## SNMP Traps

The Avaya IP600 agent generates SNMP traps for Avaya IP600 alarms and restarts. As per requests by DEFINITY Proxy Agent customers, it also generates traps for resolved alarms. The same traps are generated for all alarms. SNMP traps are sent for each new Avaya IP600 alarm, when Avaya IP600 alarms are resolved, and for Avaya IP600 restarts.

The agent will generate SNMP traps for INTUITY AUDIX alarms that are sent to the Global Alarm Monitor (GAM), and send SNMP traps for NT events that are sent to the GAM.

For INTUITY AUDIX alarm traps, the SNMP native agent also implements the portions of the CornerStone MIB that apply to alarms. For Avaya IP600 data, Avaya IP600 uses a new MIB, which is a modified subset of the DEFINITY Proxy Agent MIB.

## Avaya IP600 Co-Resident Modules

The SNMP agent uses the Avaya IP600 co-resident modules as follows:

### GAM

The SNMP agent will not send alarms to the GAM for INADS alarm reporting, but will receive alarm notifications from the GAM. The SNMP agent generates TCP/IP alarm notifications (traps) based on alarm notifications received from the GAM.

### WatchDog

The SNMP agent is registered with the Avaya IP600 WatchDog process. This allows it to start automatically when the operating system is booted. An administrator will also be able to start and stop the native agent manually. The SNMP native agent will not subscribe to the Avaya IP600 WatchDog "heartbeat" (handshake) service.

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## License Server

The SNMP agent will use the license server.

## DSA

SNMP Native Agent does not require DSA. Administration of SNMP Native Agent is via GAS commands.

## Logins and the LAC

The LAC is modified to provide a special interface for the SNMP. This interface allows the SNMP agent to access Avaya IP600 at a craft level without knowing the password for this login. The interface will be restricted to local machine instances of SNMP only.



# A Avaya IP600 Features

This appendix provides a list of the features of Avaya IP600 arranged in the following categories:

- Automatic Routing Features
- Basic Features
- Call Center Features
- Private Networking Features
- Trunk Group Features

This appendix lists all Avaya IP600 capabilities available anywhere. *Some of the listed features are optional.* Please check with your local Avaya representative for further information about system features and what is available in your country.

The *DEFINITY ECS Release 9 Administrator's Guide* (555-233-506) 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, AUDIX voice messaging system, and Call Management System. See your local distributor for more information on each of these features.

## Automatic routing features

Avaya IP600 provides a variety of automatic routing features for public and private networks. Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) are the foundation for these automatic routing features. They route calls based on the preferred (normally the least expensive) route available at the time the call is placed. Generally, AAR routes calls over a private network and ARS routes calls using the public network numbering plan. However, both AAR and ARS support public and private networks. You can use the other features listed in this section when you use AAR and ARS.

- Automatic Alternate Routing (AAR)
- Automatic Route Selection (ARS)
- AAR/ARS Overlap Sending
- AAR/ARS Partitioning
- Alternate Facility Restriction Levels
- Facility Restriction Levels and Traveling Class Marks
- Generalized Route Selection
- Subnet Trunking
- Time of Day Routing

## Basic features

The following features are supported with Avaya IP600:

- Abbreviated Dialing
- Administered Connections
- Administrable Language Displays
- Administration Change Notification
- Administration Without Hardware
- Alphanumeric Dialing
- Alternate Operations Support System Alarm Number
- Answer Detection
- Attendant Auto-Manual Splitting
- Attendant Backup Alerting
- Attendant Call Waiting
- Attendant Calling of Inward Restricted Stations
- Attendant Console
- Attendant Control of Trunk Group Access
- Attendant Crisis Alert
- Attendant Direct Extension Selection With Busy Lamp Field
- Attendant Direct Trunk Group Selection
- Attendant Display
- Attendant Intrusion (Call Offer)
- Attendant Override of Diversion Features
- Attendant Priority Queue
- Attendant Recall
- Attendant Release Loop Operation
- Attendant Serial Calling
- Attendant Split Swap
- Audible Message Waiting
- Audio Information Exchange Interface
- Authorization Codes
- Auto Start and Don't Split
- Automatic Callback

- Automatic Call Timer
- Automatic Circuit Assurance
- Automatic Exclusion
- Automatic Incoming Call Display
- Automatic Route Selection/Automatic Alternate Routing Shortcut Dialing
- Automatic Transmission Measurement System
- Block Collect Call
- Bridged Call Appearance — Multi-Appearance Telephone
- Bridged Call Appearance — Single-Line Telephone
- Bulletin Board
- Busy Verification of Terminals and Trunks
- Call Charge Information
- Call Coverage
- Call Detail Recording
- Call Forwarding
- Call Park
- Call Pickup
- Call Waiting Termination
- Class of Restriction
- Class of Service
- Code Calling Access
- Conference — Attendant
- Conference — Terminal
- Consult
- Controlled Toll Restriction
- Coverage Callback
- Coverage Incoming Call Identification
- Crisis Alert to a Digital Station
- Customer-Provided Equipment Alarm
- Data Call Setup
- Data Hot Line
- Data Privacy
- Data Restriction
- Default Dialing

- Demand Print
- Dial Access to Attendant
- Dial Plan
- Dialed Number Identification Service
- Distinctive Ringing
- Dual DCP I-Channels
- Easy Beyond Today
- Emergency Access to the Attendant
- Enhanced Abbreviated Dialing
- Enhanced Voice Terminal Display
- Extended User Administration of Redirected Calls
- External Device Alarming
- Facility Busy Indication
- Facility Test Calls
- Fiber Link Administration
- Go to Cover
- Group Listen
- Group Paging
- Hold
- Hold — Automatic
- Hunt Groups
- Individual Attendant Access
- Integrated Directory
- Integrated Services Digital Network — Basic Rate Interface (ISDN-BRI)
- Intercept Treatment
- Intercom — Automatic
- Intercom — Dial
- Internal Automatic Answer
- Last Number Redial
- Leave Word Calling
- Line Lockout
- Listed Directory Number
- Loudspeaker Paging Access

- Manual Message Waiting
- Manual Originating Line Service
- Manual signaling
- Misoperation Handling
- Modem Pooling
- Multi-Appearance Preselection and Preference
- Music-on-Hold Access
- Night Service
- Numeric Terminal Display
- PC/PBX Connection
- Personal Station Access
- Personalized Ringing
- Power Failure Transfer (Emergency Transfer)
- Priority Calling
- Privacy — Attendant Lockout
- Privacy — Manual Exclusion
- Public Network Call Priority
- Pull Transfer
- Recall signaling
- Recorded Announcements
- Recent Change History
- Recorded Announcement
- Recorded Telephone Dictation Access
- Remote Access
- Restriction — Controlled
- Ringback Queuing
- Ringer Cutoff
- Ringing — Abbreviated and Delayed
- Security Violation Notification
- Send All Calls
- Station Hunting
- Station Security Codes
- Station Used As Virtual Extension
- Station User Administration

- Telephone Self Administration
- Temporary Bridged Appearance
- Tenant Partitioning
- Terminal Translation Initialization
- Terminating Extension Group
- Timed Reminder and Attendant Timers
- Transfer
- Transfer — Outgoing Trunk to Outgoing Trunk
- Translation Copy Protection
- Trunk Flash
- Trunk Group Busy/Warning Indicators to Attendant
- Trunk Identification By Attendant
- Trunk-to-Trunk Transfer
- Visually Impaired Attendant Service
- Voice Message Retrieval
- Voice Terminal Alerting Options
- Voice Terminal Display
- Whisper Page
- World Class Tone Detection
- World Class Tone Generation

## Call Center features

Avaya IP600 offers the following features designed to help you set up and maintain a modern Call Center:

- Abandoned Call Search
- Add/Remove Skills
- Agent Call Handling
- Auto-Available Split
- Automatic Call Distribution
- Basic Call Management System
- BCMS-VU (additional cost)
- Best Services Routing (Queue to Best)
- Call Prompting

- Call Vectoring
- Calling Party/Billing Number
- CentreVu Advocate
- CentreVu CT Server (additional cost)
- CentreVu Virtual Routing
- Direct Agent Announcement
- Duplicate Agent
- Expert Agent Selection
- Flexible Billing
- Holiday Vectoring
- Inbound Call Management
- Intraflow and Interflow
- Enhanced Look-Ahead Interflow
- Malicious Call Trace
- Multimedia Call Handling
- Multiple Call Handling
- Queue Status Indications
- Reason Codes
- Redirection on No Answer
- Remote Agent Logout
- Service Observing
- Universal Call ID
- VDN in a Coverage Path
- VDN of Origin Announcement
- Voice Response Integration
- VuStats

## Private networking features

The great expandability of Avaya IP600 makes it a logical choice for setting up private networks. Consequently, the system includes many private networking features:

- Centralized Attendant Service
- Distributed Communications System
- DCS Alphanumeric Display for Terminals
- DCS Attendant Control of Trunk Group Access
- DCS Attendant Display
- DCS Automatic Callback
- DCS Automatic Circuit Assurance
- DCS Busy Verification of Terminals and Trunks
- DCS Call Coverage
- DCS Call Forwarding
- DCS Call Waiting
- DCS Distinctive Ringing
- DCS Leave Word Calling
- DCS Multiappearance Conference/ Transfer
- DCS Over ISDN-PRI D-channel
- DCS Trunk Group Busy/Warning Indication
- DCS With Reroute
- Enhanced DCS
- Extended Trunk Access
- Extension Number Portability
- Inter-PBX Attendant Calls
- Node Number Routing
- Private Network Access
- QSIG
- QSIG Call Completion
- QSIG Call Forwarding (Diversion)
- QSIG Call Independent Signaling Connections
- QSIG Call Transfer
- QSIG DCS Interworking - Called Number ID
- QSIG Message Waiting Indication (MWI)

- QSIG Name and Number Identification
- QSIG Path Replacement With Replacement
- QSIG Value-Called Number ID
- Transit Counter
- Uniform Dial Plan
- User to User Information over Public Network

## Trunk group features

Avaya IP600 offers an array of features for managing trunk groups efficiently:

- ATM-CES Trunks
- ATM Trunks
- Brazil — R2 MFC Backwards Signal
- Call-by-Call Service Selection
- Caller ID on Analog Trunks
- CAMA - E911 Trunks
- DS1 Trunk Service (T1 and E1)
- Digital Multiplexed Interface
- Facility and Non-Facility Associated Signaling
- IP Trunks
- ISDN — BRI and PRI
- Wideband Switching



# B Telephone Features

## Handling Incoming Calls

- **Station Used as a Virtual Extension** — Allows a customer to assign multiple, individual, virtual extensions to one physical phone. The physical phone must be analog and on the local switch. The administrator can set each virtual extension with a unique ring pattern to identify the extension for which the incoming call is intended. For example, an administrator could assign three virtual extensions, each with a unique ring pattern, to a single telephone shared by three roommates in a college dormitory. This feature affects incoming calls only; all outgoing calls are associated with the physical extension.
- **Station Hunt Before Coverage** — This feature changes the interaction that occurs between station hunting and call coverage. Station Hunt before Coverage causes a call going to a busy station to go through a station hunting process before going to coverage. If all the stations in the Hunt group are busy, the call will go to the coverage path.
- **Automatic Hold** — Allows Attendants and multi-function telephone users to alternate easily between two or more calls. For example, with automatic hold, selection of a second call automatically puts the active call (if any) on hold and makes the second call active. This feature can be activated on a system-wide basis only. When automatic hold is not activated, the selection of the second call drops the first call.
- **Hold** — Allows you to disconnect from a call temporarily, use your telephone for other call purposes, and then return to the original call.
- **Long Hold Recall** — Visual and audible warnings are sent to the telephone where a call has been on hold past a specified period of time. Both visual and audible warnings are used if the telephone is on-hook. If the telephone is off-hook, a “priority ring” is used. This is an optional feature at the system level.
- **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. Multi-appearance telephone users press the Transfer button, dial the desired extension number, and press the Transfer button again. This is an optional feature at the system level.
- **Pull Transfer** — Allows *either* the party who was originally called *or* the party to whom the held call will be transferred to complete the transfer. This is a convenient way to connect a party with someone better qualified to handle the call. Attendant assistance is not required and the call does not have to be redialed. It interfaces with satellite workstations via TGU/TGE trunks and is always available for calls that use TGU/TGE trunks.

- **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.
- **Transfer (Outgoing Trunk to Outgoing Trunk)** — Allows a user or attendant to initiate two or more outgoing trunk calls and then transfer the trunks together. The transfer operation removes the original user from the connection and conferences the outgoing trunks. Alternatively, the controlling party can establish a conference call with the outgoing trunks and then drop out of the conference, leaving only the outgoing trunks on the conference. This is an optional enhancement to Trunk-to-Trunk Transfer and requires careful administration and use. DCS Trunk Turnaround (not available with Category B) may be a safer alternative to this feature.
- **Transfer Upon Hang-Up** — Provides you with the ability to transfer a call by hanging up instead of having to press the Transfer button a second time. You would press the Transfer button, dial the number the call is being transferred to and then hang up. This is an optional feature at the system level. You will still be able to transfer a call by pressing the Transfer button a second time.
- **Transfer Recall** — Returns the unanswered transfer calls back to the person who transferred the call. Transfer Recall uses a priority alerting signal, and the display on the telephone shows "rt", which indicates a returned call from a failed transfer operation.
- **Abort Transfer** — Allows a user to abort a transfer attempt by pressing a non-idle line appearance. The call being transferred would be taken off a transfer-type hold and be put on a traditional hold. The transfer will also be aborted when you hang up (going on-hook), unless Transfer Upon Hang-Up is activated on the switch. This is an optional feature at the system level.

## Conference

The Conference button allows multiappearance telephone users to make up to six-party conference calls without attendant assistance. This feature also allows single-line telephone users to make up to three-party conference calls without attendant assistance.

- **Abort Conference on Hangup** — When you punch the conference button and for any reason you hang up before you complete the conference, you will cancel the conference. The original call that was put on soft-hold will now be on hard-hold.

## Call Park

Allows you to put a call on hold and then retrieve a call from any other telephone on the system. This is helpful when you are on a call and need to go to another location for information. It also allows you to answer a call from any telephone after being paged by a telephone user or an attendant.

## Misoperation Handling

**Note:** This feature is required only in France and Italy, but it can be used at any location where the feature has been turned on.

Defines how calls are handled when a misoperation occurs. A misoperation is when calls are left on hold when the controlling station goes on hook.

For example, a misoperation can occur under either of the following conditions:

- If you hang up prior to completing a feature operation (in some cases, hanging up completes the operation, as in call transfer). If, for example, you place a call on hold, begin to transfer the call, dial an invalid extension number, and then hang up, that's a misoperation.
- When the system enters night service while attendant consoles have calls on hold.

The system administrator can alter the standard Misoperation Handling to ensure that an external caller is not left on hold indefinitely, or dropped by the system after a misoperation with no way to reach someone for help.

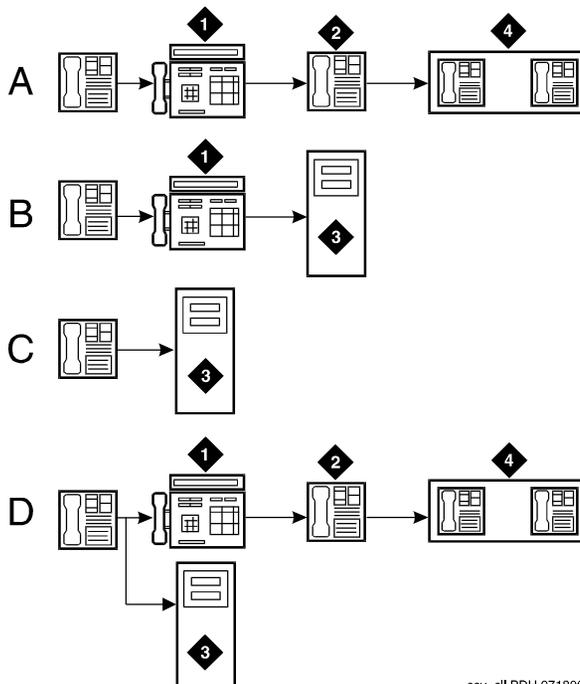
- **Manual Message Waiting** — Allows multi-appearance telephone users to light the status lamp associated with the manual Message Waiting button at another multi-appearance telephone. They do this by simply pressing a button on their own telephone. This feature can be administered only to pairs of telephones such as 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.
- **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. You simply press an Internal Automatic Answer feature button, and calls are automatically answered when the telephone is idle. Internal and Distributed Communications System (DCS) calls (not available with Category B) can be answered using Automatic Answer, but only attendants can use Automatic Answer to answer external calls directed to the attendant.
- **Recall Signaling** — Recall Signaling allows the user of an analog station to place a call on hold, use the telephone for other call purposes, and then return to the original call.
- **Local Call Timer Automatic Start/Stop** — Automatically starts the local timer of a 6400-series telephone when a call is received. The timer is stopped automatically when a call is ended. When a call is placed on hold the timer continues to run, but is not displayed. When the call comes off hold, the total elapsed call-time displays.
- **Administrable Timeout on Call Timer** — Enhances the Call Timer feature on the 6400-series telephones. The Call Timer feature measures the duration of a call, starting a timer when the call is answered and stopping the timer when the call is dropped. Previously, the Call Timer feature displayed the duration of the call for only 5 seconds after the call was dropped. With Release 9, the Administrable Timeout on Call Timer feature introduces a new system-wide field that allows the user to specify how long to display the duration of the call.

## Call Redirection

- Call Forwarding — Call Forwarding provides four functions:
  - ~ Call Forwarding All Calls — Allows calls to be forwarded to an internal extension, external (off-net) number, an attendant, or an attendant group.
  - ~ Call Forwarding Override — Allows the user at the forwarded-to extension to override Call Forwarding and either initiate a call or transfer a call back to the forwarded-from extension.
  - ~ Call Forward Busy/Don't Answer — Allows calls to be forwarded when the called extension is busy or when the call is not answered after an administrable interval. If the extension is busy, the call forwards immediately. If the extension is not busy, the incoming call rings the called extension, then forwards only if it remains unanswered longer than the administered interval.
  - ~ Call Forwarding Off Net — Allows calls forwarded off net to be tracked for busy or no-answer conditions. The system brings the call back for further call-coverage processing if specified conditions are met. This feature is particularly useful for Telecommuters, who can have their on-site office calls forwarded to their home offices.
- Call Coverage — Call Coverage provides automatic redirection of calls that meet specified criteria to alternate answering positions in a Call Coverage path. A coverage path can include any of the following: a telephone, an attendant group, a Uniform Call Distribution (UCD) hunt group, a Direct Department Calling (DDC) hunt group, an Automatic Call Distribution (ACD) hunt group, a voice messaging system, or a Coverage Answer Group (CAG) established to answer redirected calls.
  - ~ In addition to redirecting a call to a local answering position, you can administer Call Coverage to:
    - ~ Redirect calls based on time-of-day
    - ~ Redirect calls to a remote location
    - ~ Allow users to change back and forth between two lead-coverage paths from either an on- or off-site location
    - ~ Coverage Callback — Allows a covering user to leave a message for the called party to call back the person who called.
    - ~ Coverage Incoming Call Identification — Allows multi-appearance telephones users without a display in a Coverage Answer Group to identify an incoming call to that group.
    - ~ Go to Cover — Allows users who call another internal extension to send the call directly to coverage.
    - ~ Send All Calls — Allows users to temporarily direct all incoming calls to coverage regardless of the assigned call-coverage redirection criteria. Covering users can temporarily remove their telephones from the coverage path. The feature is activated and deactivated via a button or access code.
    - ~ Consult — Allows a covering user, after answering a call received 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.

~ Reset Shift Call — If a call number is busy and doesn't have coverage or the called number and the coverage are both busy, you have an opportunity to replace the last digit that was entered. This allows you to call another extension without having to hang up and redial. Reset Shift Call is a feature that is active for station to station (internal) calls and for Private Network calls. The Private Network trunks must signal busy using out-of-band signaling.

**Figure 7. Typical Avaya IP600 Call Coverage Options**



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

## Night Service

There are five Night Service features:

- Hunt Group Night Service allows an attendant or a split supervisor to assign a hunt group or split to Night Service mode. All calls for the hunt group then are redirected to the hunt group's designated Night Service extension. When a user activates Hunt Group Night Service, the associated button lamp lights.
- Night Console Service directs all calls for primary and daytime attendant consoles to a night console. When a user activates Night Console Service, the Night Service button for each attendant lights and all attendant-seeking calls (and calls waiting) in the queue are directed to the night console. To activate and deactivate this feature, the attendant typically presses the Night button on the principal attendant console or designated console.
- Night Station Service directs incoming calls for the attendant to designated extensions. Attendants can activate Night Station Service by pressing the Night button on the principle console if there is not an active night console. If the night station is busy, calls (including emergency attendant calls) receive a busy tone. They do not queue for the attendant.
- 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 the calls. The incoming call activates a gong, bell, or chime and a voice-terminal user dials an access code to answer the call.
- Trunk Group Night Service allows an attendant or a designated telephone user to individually assign a trunk group or all trunk groups to the night service mode. Specific trunk groups individually assigned to the service are in Individual Trunk Night Service Mode. Calls coming into these trunk groups are redirected to designated night service extensions. Incoming calls on other trunk groups are processed normally.

## Enhanced Night Service

Avaya IP600 informs a Voice Mail System (VMS) that it is in Night Service, allowing the VMS to perform different actions and call handling for out-of-hours operation. For example, the VMS may be administered to provide recorded announcements after hours. The enhancement is made to the Mode Code Voice Mail Interface.

### Bridged Call Appearance — Multi-Appearance Telephone

Allows calls made to or from a primary telephone user's extension number 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 multi-lamp button on another telephone. This feature is most often used 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 on all telephones administered with this feature. The call can be answered by anyone having a telephone with this feature and handled as if the primary extension user was answering it. The maximum number of bridged appearances is 64.

- **Bridged Call Appearance (Single-Line Telephone)** — Allows single-line telephones users to have a bridged appearance on a multi-appearance telephone.
- **Temporary Bridged Appearance** — Allows multi-appearance telephone users in a terminating extension group or personal central office line group to bridge onto an existing group call. If a call has been answered using the Call Pickup feature, the originally called 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.
- **Privacy (Manual Exclusion)** — Allows multi-appearance telephone users to keep other users with appearances of the same extension number from bridging onto an existing call. Exclusion is activated by pressing the Exclusion button on a per-call basis.
- **Privacy (Auto Exclusion)** — When the Class of Service is set for the Automatic Exclusion option, the feature is activated when you take your telephone off-hook. The feature can be deactivated when you push the Exclusion button before dialing a call or during a call. An excluded call that is on hold can be taken off hold by any telephone that has a bridged appearance of the telephone that put the call on hold.

### Call Pickup

Along with Directed Call Pickup, allows you to answer calls for other telephones within your specified call pickup group. Directed Call Pickup allows you to pick up any call on the DEFINITY ECS system. With this feature, you do not have to leave your telephone to answer a call for a nearby telephone. You simply dial an access code or press a Call Pickup button.

- **Group Call Pickup** — Allows you to dial a Feature Access Code (FAC) and a Pickup Group Number to answer a call from a different group. For example, Marketing would be able to pickup calls in the Sales group when the Sales group is unavailable. This feature is ideal for offices that are not divided by partitions and generally have the departments on the same floor.
- **Terminating Extension Group** — Allows an incoming call to ring (either audible or silent alerting) as many as four telephones at the same time. Any user in the group can answer the call. Any telephone can be administered as a group member. Only a multi-appearance telephone can be assigned a feature button with an associated status lamp, however. 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.

## Station Hunting

Routes calls made to a busy extension to another extension. To use Station Hunting, you create a station hunting chain that governs the order in which a call routes from one extension to the next when the called extension is busy. Each extension in the chain links *to* only one subsequent extension. An extension may be linked *from* any number of extensions, however.

- **Station Hunt Before Coverage** — This feature changes the interaction that occurs between station hunting and call coverage. Station Hunt before Coverage causes a call going to a busy station to go through a station hunting process before going to coverage. If all the stations in the Hunt group are busy, the call will go to the coverage path.
- **Circular Station Hunting** — This new hunt group type is an alternative to the “ddc” or “hot-seat” algorithm in a hunt group. DEFINITY ECS keeps track of the last extension in the hunt group that received a call. When another incoming call arrives, it is sent to the next idle extension, bypassing the extension that had received the previous call. The first extension in the hunt group will no longer be the busiest telephone while the others in the group are sitting idle.

## Disconnecting Unanswered Calls

Disconnects unanswered outgoing calls after a predetermined amount of time. When any of the following timers expire during an outgoing local, toll, or international call attempt, the switch disconnects the call and applies busy tone, which may or may not be followed by howler tone:

- Pre-dialing and interdigit timer
- Outgoing seizure acknowledge timer
- Answer supervision timer
- 60-, 90-, and 120-second no-answer disconnect timers, based on ARS call type
- 120-second timer used for calls without a call type, such as calls to trunk access codes.

# Speed/Convenience Calling

## Abbreviated Dialing

Provides lists of stored numbers you can use to:

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

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

- **Abbreviated Dialing Labeling** — of Abbreviated Dialing (AD) Buttons on Softkeys allows users of 8400-, 6400-, and 4600-series display telephone sets to administer labels for the AD buttons that appear on their softkeys. These personalized labels appear on the menu display.
- **Abbreviated Dialing On-Hook Programming** — On-Hook Programming allows users of 8400-, 6400-, and 4600-series telephone sets with enabled speakers to access the programming mode without going off-hook during available call appearances. Signaling changes from DTMF to the S-channel, allowing the use of a longer (60 seconds) time-out period. Signaling will remain DTMF and the current time-out period of 10 seconds will still apply to non-display telephone sets.
- **Enhanced Abbreviated Dialing** — Supplements Abbreviated Dialing by providing one enhanced number per system. Enhanced number lists can contain any number or dial access code. System Administrators designate privileges for group number lists, system number lists and enhanced number lists. With privileged lists, users can access otherwise-restricted numbers (e.g., Stations without long-distance access can be programmed to access specified long-distance numbers).

## Active Dialing

6400- and 4600-series telephone sets have a dialing option where the set will send S-channel button codes when the user presses a number on the dial pad when on-hook.

## Telephone Self-Administration

Allows you to program feature buttons on 6400- and 4600-series telephones yourself.

## 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.

When a user activates Automatic Callback, the system monitors the called telephone. When the called telephone becomes available to receive a call, the system originates the Automatic Callback call. The originating party receives priority ringing. The calling party then lifts the handset and the called party receives the same ringing provided on the original call.

## 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 signal when called back.

## Last Number Dialed

Allows you 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 you press the Last Number Dialed button or dial the Last Number dialed feature access code, the system places the call again.

## Remote Access

Permits authorized callers from remote locations to access the system via the public network and then use its features and services. There are a variety of ways of accessing the feature. After gaining access, you hear a system dial tone, and, for system security, may be required to dial a barrier code.

## 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.

## Emergency Access to the Attendant

Provides for emergency calls to be placed to an Attendant. These calls can be placed automatically by the system or can be dialed by system users. Emergency access calls can receive priority handling by the Attendant.

- Crisis Alert uses both audible and visual alerting to notify designated extensions when an emergency call is made. Audible alerting sounds like an ambulance siren. Visual alerting flashes the CRSS-ALRT button lamp and the display of the caller's name and extension. Crisis Alert's display of the origin of the emergency call enables the attendant or other user to direct emergency-service response to the caller.

If an emergency call is made while another crisis alert is still active, the incoming call will be placed in the queue. If the system is administered so that all users must respond, then every user must respond to every call, in which case the calls are not necessarily queued in the order in which they were made. If the system is administered so that only one user must respond, the first crisis alert remains active at the phone where it was acknowledged. Subsequent calls are queued to the next available station in the order in which they were made.

Crisis Alert can also send notification of an emergency call to a digital pager. In this case, it sends a message of 7 to 22 digits to the pager and displays a crisis alert code, an extension and room number, and a main number (if one is entered). The person paged thus knows the origin of the emergency call and can direct emergency-service response to the appropriate location. To use Crisis Alert with a digital pager, the system is administered so that at least one digital set has a CRSS-ALRT button and the Alert Pager field is set to y. Any station with a CRSS-ALRT button and a pager receives the correct alert.

## 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.

## Trunk Flash

Trunk Flash allows a feature or function button on a multifunction telephone or attendant console to be assigned as a Flash button. Pressing this button while connected to a trunk (which must have been administered to allow Trunk Flash) causes the system to send a flash signal out over the connected trunk.

Trunk Flash enables multifunction telephones to access central office customized services that are provided by the Central Office to which DEFINITY ECS is connected. These services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the DEFINITY System station on an active trunk call. The Trunk Flash feature can help to reduce the number of trunk lines connected to the DEFINITY system. "Digit 1 as Flash" as used in Italy and the United Kingdom will not serve as the flash button in this application.

## Special Dial Tone

Provides the ability to play a Special Dial Tone whenever an analog set is not able to receive calls. When such conditions as Call Forward All Calls, Call Forward Busy/NA, Send All Calls or Do Not Disturb are activated on a telephone set, a Special Dial Tone lets you know that you cannot receive any calls.

## Call Charge Information

DEFINITY ECS provides two ways to know the approximate charge for outgoing calls:

- Advice of Charge — For ISDN trunks

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

- Periodic Pulse Metering — For non-ISDN trunks

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

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

## Telephone Displays

- Telephone Display — Provides multi-appearance telephone users with updated call and message information. This information is displayed on a display-equipped telephone. The information displayed depends upon the display mode selected by the user. Information that allows personalized call answering is available on many calls.

Users may select any of the following as the display message language: English (default), French, Italian, or Spanish. In addition, messages can be administered on the system in a fifth language. The language for display messages is selected by each user.

- ICLID on Analog Trunk — In the US, the user's telephone displays calling party information. Name and calling number are available from the US central offices. This feature may be used in countries that comply with either US. The display of name and number will work with all DEFINITY digital telephones (DCP and BRI) equipped with a 40-character or a 32-character alphanumeric display.
- Enhanced Telephone Display — The Enhanced Telephone Display feature allows you to choose the character set that you want to see in DEFINITY ECS softkeys and display telephones. In addition to the standard Roman character set, you can choose either the Katakana or characters used for most European languages.

- **Administrable Language Displays** — Allows the messages that appear on telephone display units to be shown in the language spoken by the user. These messages are available in English (the default), French, Italian, Spanish, or one other user-defined language. The language for display messages is selected by each user. The feature requires 40-character display telephones.
- **Directory** — Allows users with display-equipped telephones to access the system database, use the touch-tone buttons to enter a name, and retrieve an extension number from the system directory. The directory contains the names and extensions assigned to all telephones on the system.

## Group Communication

- **Conference (Telephone)** — Allows multi-appearance telephone users to set up six-party conference calls without attendant assistance. Single-line telephone users can set up three-party conference calls without attendant assistance.
- **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 multi-appearance telephone users to easily call others within an administered group. The calling user lifts the handset, presses the Dial Intercom button, and dials the one- or two-digit code assigned to the desired party. The called user's telephone 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.
- **Manual Signaling** — Allows one user to signal another user. The receiving user hears a two-second ring. The signal is sent each time the button is pressed by the signaling user. The meaning of the signal is prearranged between the sender and the receiver. Manual Signaling is denied if the receiving telephone is already ringing from an incoming call.
- **Group Listen** — Simultaneously activates your speakerphone in listen only mode and your handset or headset in listen and speak mode. This allows you to serve as spokesperson for a group. You can participate in a conversation while everyone else in the room is listening to what is said. This feature works only on certain types of telephones. It is not supported on IP hardphones.
- **Group Paging** — Allows a user to make an announcement to a group of people via their speakerphones. The speakerphones are automatically turned on when the user begins the announcement. The recipients can listen to the message via the handset if they wish, but they cannot speak to the user in return. A group page member will not receive the page if the member is active on a call appearance, has a call ringing, is off-hook, has "send-all calls" active, or has "do not disturb" active.

- **Whisper Page** — Allows an assistant or colleague to bridge onto your telephone conversation and give you a message without being heard by the other party or parties you are talking to. Whisper Page works only on certain types of telephones.
- **Loudspeaker Paging Access** — Provides attendants and telephone users dial access to voice paging equipment. As many as nine paging zones can be provided by the system and one zone can be provided that activates all zones at the same time. (A zone is the location of the loudspeakers — for example, conference rooms, warehouses, or storerooms.) A user can activate this feature by dialing the trunk access code of the desired paging zone, or the access codes can be entered into Abbreviated Dialing Lists. Once you have activated this feature, you can simply speak into the handset to make the announcement.

Deluxe Loudspeaker Paging Access (called Deluxe Paging) provides attendants and telephone users with integrated access to voice-paging equipment and Call Park capabilities. When you activate Deluxe Paging, the call is automatically parked. The parked call returns to the parking user with distinctive alerting when the time-out interval expires.

- **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.

## Special Ringing

- **Distinctive Ringing** — Rings or activates alerting on your telephone in such a way that you are aware of the type of incoming call before answering it. This feature operates in a Distributed Communication System (DCS) environment the same as it does within a single system.

By default, internal calls are identified by a 1-burst ringing pattern, external calls by a 2-burst ringing pattern, and priority calls by a 3-burst ringing pattern. You can administer these patterns, however.

- **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.
- **Priority Calling** — Allows you 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. You activate priority calling by Dialing a Priority Calling access code or pressing a feature button, followed by the extension number. You can use Priority Calling only if your telephone has been administered with the required class of service.
- **Ringing Options** — Provides multi-appearance telephone users with different ringing patterns. This feature primarily affects audible ringing for calls directed to telephones that are off hook, or calls directed to idle and active CALLMASTER telephones.

- **Ringling — Abbreviated and Delayed** — Allows you to manually or automatically assign one of four ring types to each call appearance on a telephone. Whatever treatment you assign to a call appearance is automatically assigned to each of its bridged call appearances.
- **Ringer Cutoff** — Allows the user of a multi-appearance telephone to turn audible ringing signals on and off. Visual alerting is not affected by this feature. When this feature is enabled, only Priority (three-burst) ring, Redirect Notification, Intercom ring, and Manual Signaling ring at the telephone. Internal and external calls do not ring.
- **Multiappearance Preselection and Preference** — Provides options for placing or answering calls on selected call appearances. Ringing Appearance *Preference* automatically connects you to the incoming ringing call when the user picks up the handset. *Idle Appearance Preference* automatically connects you to an idle appearance. *Preselection* allows the user to manually select an appearance. Preselection is used, for example, when you want to reconnect with a held call or activate a feature. Preselection can be used with a feature button. For example, if you press an Abbreviated Dialing button, the call appearance is automatically selected and, if you pick up the handset within five seconds, the call is automatically placed. The Preselection option overrides both of the other preference options.

## Messaging

- **Leave Word Calling** — Allows internal system users to leave a short preprogrammed 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 DEFINITY ECS, the Message Lamp on the called telephone automatically lights. Leave Word Calling messages can be retrieved using a telephone display, Voice Message Retrieval, or AUDIX. Messages may be retrieved in English, French, Italian, Spanish, or a user-defined language.
- **Audible Message Waiting** — Places a stutter at the beginning of the dial tone when a telephone user picks up the telephone. The stutter dial tone indicates that the user has a message waiting. This feature is particularly useful for visually impaired people who may not be able to see a message light. It is often used with telephones that have no Message Waiting Lights. Audible Message Waiting may not be available in countries that restrict the characteristics of dial tones provided to users.
- **Voice Messaging and Call Coverage** — Often an AUDIX system is set up as the last point on a call-coverage path, as shown in Figure 7 on page 113. 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.

- **Voice Message Retrieval** — Allows telephone users, remote access users, and attendants to retrieve Leave Word Calling and Call Coverage voice messages. It can be used to retrieve a user's own messages or messages for another user. However, a different user's messages can be retrieved only by a user at a telephone or attendant console in the coverage path, by an administered system-wide message retriever, or by a remote-access user when the extension and associated security code are known. The system restricts unauthorized users from retrieving messages.
- **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 telephone.

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

- **Demand Print** — Allows you to print your undelivered messages without calling the Message Center.
- **Voice Messaging Systems** — The Avaya voice messaging solutions include:
  - ~ DEFINITY AUDIX
  - ~ INTUITY AUDIX
  - ~ INTUITY Lodging
  - ~ Octel 100

Voice Response solutions include:

- ~ CONVERSANT Voice Information System V7 (Not available with Category B)

## Telecommuting

- **Remote Call Coverage/Call Forwarding Off-Net** — Remote Call Coverage and Call Forwarding Off-Net allow calls to be redirected to a remote location. This allows you to have calls placed to your on-site office redirected to your home office. You can administer the system to either monitor calls and bring them back for additional processing if not answered or to leave calls at the remote (off-net) location.
- **Coverage of Calls Redirected Off-Net** — Coverage of Calls Redirected Off-Net (CCRON) allows calls that have been redirected to locations outside of the switch to return to the switch for further processing. For example, an employee that telecommutes can have two coverage paths. One coverage path is used when the employee is in the office and the other coverage path is used when the employee is working from home. The coverage path used from home would have a call to the employee's work phone cover to his or her home phone. If the employee does not answer the call or is busy on another call, the call is redirected back to the switch for further processing, such as coverage to voice mail.
- **Extended User Administration of Redirected Calls (Telecommuting Access)** — Extended User Administration of Redirected Calls (also called Telecommuting Access) allows you to change the lead call coverage path or forwarding extension from any on-site or off-site location. Thus you can change the path or extension from your home office, for example.
- **Off-Premises Station** — A trunk-data module connects off-premises private-line trunk facilities and DEFINITY ECS. The trunk-data module converts between the RS-232C and the DCP, and can connect to DDD modems as the DCP member of a modem pool.
- **Personal Station Access** — Allows you to transfer your telephone station preferences and permissions to any other compatible telephone. This includes the definition of telephone buttons, abbreviated dial lists, and Class of Service and Class of Restrictions permissions. It can be used on-site or off-site (with DEFINITY Extender). Personal Station Access can also be used to prevent unauthorized calls from your phone when you are away from your desk. You would disassociate your desk telephone from your extension number when you are away from your desk and re-associate your desk telephone number with your extension number when you return. Personal Station Access has several telecommuting applications. For example, several telecommuting employees can share the same office on different days of the week. The employees can easily and remotely make the shared telephone "theirs" for the day. Remote use requires DEFINITY Extender.
- **Station Self Display** — Station Self Display shows the extension number of the telephone set when a user either dials the Feature Access Code while off-hook or depresses the "Inspect" button when on-hook. The dialed number will be displayed once the user starts to dial. This feature is helpful to people who move from one desk to another while they are working. This feature is also used by maintenance personnel to ensure that an extension number is correctly administered.



# C Attendant (Operator) Features

This section discusses the features of the following attendant console:

- [Attendant Console](#)
- [DEFINITY PC Console](#)
- [Attendant with DCS](#)

## Attendant (Operator)

A person at a console who provides personalized service for incoming callers and voice-services users by performing switching and signaling operations.

## Attendant Console

A digital call-handling station with push-button control used not only to answer and place calls, but also to manage and monitor some system operations.

The Attendant Display shows call-related information that helps the attendant to operate the console. Also shows personal service and message information. Information is shown on the alphanumeric display on the attendant console. Attendants may select one of several available display message languages: English, French, Italian, or Spanish. In addition, your company may define one additional language for use by users and attendants on their display.

## DEFINITY PC Console

Avaya DEFINITY PC Console allows your Attendants to handle incoming calls efficiently by personal computer. Using the familiar Microsoft Windows graphical interface, the Attendants can easily keep 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 make 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.

The PC Console is easily customized, so even if attendants from different shifts share the same computer, they can each preserve their preferences in the call processing environment. The PC Console is available in English, Parisian French, Latin American Spanish, German, Dutch, Italian and Portuguese. 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 online help to Spanish.

# Attendant Features

## Call Handling

- **Listed Directory Number** — Allows outside callers to access your attendant group in two ways, depending on the type of trunk used for the incoming call. You can allow attendant group access via incoming direct inward dial trunks, or you can allow attendant group access via incoming (Central Office) and foreign exchange trunks.
- **Call Waiting** — Allows an Attendant to let a single-line telephone user who is on the phone know that a call is waiting. The Attendant is then free to answer other calls. The Attendant hears a call waiting ringback tone and the busy telephone user hears a call waiting tone. This tone is heard only by the called telephone user.
- **Calling of Inward Restricted Stations** — A telephone with a Class of Restriction that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls. This feature allows you to override this restriction.
- **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 twelve different categories of incoming attendant calls, including emergency calls, which are given the highest priority.
- **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.
- **Backup Alerting** — Notifies backup Attendants that the primary Attendant cannot pick up a call. It provides both audible and visual alerting to backup stations when the attendant queue reaches its queue warning level. When the queue drops below the queue warning level, alerting stops. Audible alerting also occurs when the attendant console is in night mode, regardless of the Attendant queue size.
- **Timed Reminder and Attendant Timers** — Automatically alerts the Attendant after an administered time interval for the following types of calls: extended calls to be answered or waiting to be connected to a busy single-line telephone, one-party calls placed on hold on the console, and transferred calls that have not been answered after transfer. Timed Reminder informs the Attendant that a call requires additional attention. After the Attendant reconnects to the call, the user can either choose to try another extension number, hang up, or continue to wait. DEFINITY ECS supports a variety of administrable attendant timers for use in a variety of situations.
- **Privacy (Attendant Lockout)** — Prevents an Attendant from re-entering 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.
- **Intrusion (Call Offer)** — Allows an Attendant to enter an existing call to inform the person being called about a message or another call.

- **Release Loop Operation** — Allows the Attendant to hold a call at the console if the call cannot immediately go through to the person being called. A timed reminder begins once the call is on hold. If the call is not answered within the allotted time, the call returns to the queue for the Attendant. Timed reminders attempt to return the call to the Attendant who previously handled it. Only when the original Attendant is unavailable are calls returned to the queue.
- **Attendant Vectoring** — Vectoring provides a highly flexible approach for managing incoming calls to an attendant. For example, with current night service operation, calls redirected from the attendant console to a night station can ring only at that station and will not follow any coverage path. With Attendant Vectoring, night service calls will follow the coverage path of the night station. The coverage path could go to another station and eventually to a voice mail system. The caller can then leave a message that can be retrieved and acted upon.
- **Attendant Split Swap** — Allows the attendant to alternate between active and split calls. This operation may be useful if the attendant needs to transfer a call but first must talk independently with each party before completing the transfer.
- **Serial Calling** — Enables an Attendant to transfer trunk calls that return to the same Attendant after the called party hangs up. The returned call can then transfer to another station within the switch. This feature is useful if trunks are scarce and Direct Inward Dialing services are unavailable. An outside caller may have to redial often to get through because trunks are so busy. Once callers get through to an Attendant they can use the same line into the switch for multiple calls. The Attendant's display shows if an incoming call is a serial call.
- **Conference** — Allows an Attendant to set up a conference call for as many as six conferees, including the Attendant. Conferences from inside and outside the system can be added to the conference call.

## Making Calls

- **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.
- **Auto-Manual Splitting** — Allows an Attendant to announce a call or consult privately with the called party without being heard by the calling party on the call. It splits the calling party away so the Attendant can confidentially determine if the called party can accept the call.
- **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.

## Accessing the Attendant

- **Recall** — users to recall the Attendant when they are on a two-party call or on an Attendant Conference call held on the console. Single-line users press the Recall button or flash the switchhook to recall the Attendant. Multi-appearance users press the Conference or Transfer button to recall the Attendant and remain on the connection when either button is used.
- **Emergency Access to the Attendant** — Provides for emergency calls to be placed to an Attendant. These calls can be placed automatically by the system or can be dialed by system users. Emergency access calls will generally receive priority handling by the Attendant.
- **Dial Access to Attendant** — llows you to reach an Attendant by dialing an access code. The Attendant can then extend the call to a trunk or to another telephone.
- **Individual Attendant Access** — Allows you to call a specific attendant console. Each attendant console can be assigned an individual extension number.

## Monitoring Calls

- **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.
- **Crisis Alert** — Visibly and audibly alerts Attendants when an emergency call is placed. The feature indicates from where an emergency call is made, which allows the Attendant to direct emergency-service response to the caller. Though often used in the hospitality industry, it can be set up to work with any standard attendant console.

Audible alerting sounds like an ambulance siren. Visual alerting consists of flashing of the crisis-alert button lamp and display of the caller name and extension. When crisis alerting is active, the console is placed in position-busy mode so that no other incoming calls interfere with the emergency call. The console can still originate calls. The Attendant must press the position-busy button to unbusy the console and the crisis-alert button to deactivate audible and visual alerting.

- **Trunk Group Access** — Allows an Attendant to control trunk groups and prevents telephone users from directly accessing a controlled trunk group. This allows the Attendant to monitor the use of these trunk groups. By watching the lamps associated with the trunk groups, the Attendant can determine if the number of busy trunks in a specific trunk group has reached a preset warning level and if all trunks in a specific trunk group are busy. The Attendant can then handle other calls to these trunk groups accordingly.
- **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.

- **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.
- **Room Status (Hospitality Industry)** — Allows an Attendant to see whether a room is vacant or occupied and the housekeeping status of each room. This feature is only available when you have Enhanced Hospitality enabled for your system. This feature combines the property management capabilities of Check-In/Check-Out and Housekeeping Status but does not require that you have a Property Management System.
- **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.
- **Visually Impaired Attendant Service (VIAS)** — Provides voice feedback to a visually impaired attendant in either Italian or British English. Each voice phrase is a sequence of one or more single-voiced messages. This feature defines six new attendant buttons to aid visually impaired attendants:
  - ~ **Visually Impaired Service Activation/Deactivation button:** activates or deactivates the feature. All ringers previously disabled (for example, recall and incoming calls) become reenabled.
  - ~ **Console Status button:** voices whether the console is in Position Available or Position Busy state, whether the console is a night console, what the status of the attendant queue is, and what the status of system alarms is.
  - ~ **Display Status button:** voices what is shown on the console display. VIAS support is not available for all display features (for example, class-of-restriction information, personal names, and some call purposes).
  - ~ **Last Operation button:** voices the last operation performed.
  - ~ **Last Voiced Message button:** repeats the last voiced message.
  - ~ **Direct Trunk Group Selection Status button:** voices the status of an attendant-monitored trunk group.

The visually impaired attendant may use the Inspect mode to locate each button and determine the feature assigned to each without actually executing the feature.

## Centralized Attendant Service

Enables Attendant services in a private network to be concentrated at a central location. Each branch in a Centralized Attendant Service has its own listed directory number or other type of access from the public network. Incoming calls to the branch, as well as calls made by users directly to the Attendants, are routed to the centralized Attendants over Release Link Trunks.

## Attendant with DCS

- **Control of Trunk Group Access** — Allows an Attendant at any node in the Distributed Communication System (DCS) to take control of any outgoing trunk group at an adjacent node. This is helpful when an Attendant wants to prevent telephone users from calling out on a specific trunk group for any number of reasons, such as reserving a trunk group for incoming calls or for a very important outgoing call.
- **Direct Trunk Group Selection** — Allows the Attendant direct access to an idle outgoing trunk in a local or remote trunk group by pressing the button assigned to that 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. Direct Trunk Group Selection is intended to expedite the handling of an outgoing call by the Attendant.
- **Display** — Shows call-related information that helps the Attendant to operate the console. Also shows personal service and message information. Information is shown on the alphanumeric display on the Attendant console. Attendants may select one of several available display message languages: English, French, Italian, or Spanish. In addition, your company may define one additional language for use by users and Attendants on their display.
- **Inter-PBX Attendant Calls** — Allows Attendants for multiple branches to be concentrated at a main location. Incoming trunk calls to the branch, as well as Attendant-seeking voice-terminal calls, route over tie trunks to the main location.
- **DCS With Reroute** — Allows a DCS call to be rerouted over a different path if the switch finds a better quality and lower cost route. Earlier versions of this feature allowed for rerouting the call only after a transfer. With Release 9 enhancements, however, this feature now allows for rerouting during a call. DCS With Reroute is similar to the rerouting capabilities used with QSIG.
- **Italian DCS Protocol** — Enhanced DCS adds features to the existing DCS capabilities and requires the use of Italian TGU/TGE tie trunks. Additional features include:
  - ~ Exchanging information to provide Class of Restriction (COR) checking between switches in the EDCS network
  - ~ Providing call-progress information for the Attendant
  - ~ Allowing Attendant intrusion between a main and a satellite PBX
  - ~ Allowing a main PBX to provide DID/CO intercept treatment rather than the satellite PBX

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