Overview for

Avaya MultiVantage™ Software

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Notice

Every effort was made to ensure that the information in this document was complete and accurate at the time of printing. However, information is subject to change.

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 is not 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.

Avaya Fraud Intervention

If you suspect that you are being victimized by toll fraud and you need technical assistance or support, in the United States and Canada, call the Technical Service Center's Toll Fraud Intervention Hotline at 1-800-643-2353.

How to Get Help

For additional support telephone numbers, go to the Avaya Web site:

http://www.avaya.com/support/

If you are:

- Within the United States, click *Escalation Lists*, which includes escalation phone numbers within the USA.
- Outside the United States, click *Escalation Lists* then click *Global Escalation List*, which includes phone numbers for the regional Centers of Excellence.

Providing Telecommunications Security

Telecommunications security (of voice, data, and/or video communications) is the prevention of any type of intrusion to (that is, either unauthorized or malicious access to or use of) your company's telecommunications equipment by some party.

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 is not 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:

- Utilization (of capabilities special to the accessed equipment)
- 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).

Responsibility for Your Company's Telecommunications Security

The final responsibility for securing both this system and its networked equipment rests with you - Avaya's customer system administrator, your telecommunications peers, and your managers. Base the fulfillment of your responsibility on acquired knowledge and resources from a variety of sources including but not limited to:

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

To prevent intrusions to your telecommunications equipment, you and your peers should carefully program and configure:

- Your Avaya-provided telecommunications systems and their interfaces
- Your Avaya-provided software applications, as well as their underlying hardware/software platforms and interfaces
- Any other equipment networked to your Avaya products.

Voice Over Internet Protocol (VoIP)

If the equipment supports Voice over Internet Protocol (VoIP) facilities, you may experience certain compromises in performance, reliability and security, even when the equipment performs as warranted. These compromises may become more acute if you fail to follow Avaya's recommendations for configuration, operation and use of the equipment. YOU ACKNOWLEDGE THAT YOU ARE AWARE OF THESE RISKS AND THAT YOU HAVE DETERMINED THEY ARE ACCEPTABLE FOR YOUR APPLICATION OF THE EQUIPMENT. YOU ALSO ACKNOWLEDGE THAT, UNLESS EXPRESSLY PROVIDED IN ANOTHER AGREEMENT, YOU ARE SOLELY RESPONSIBLE FOR (1) ENSURING THAT YOUR NETWORKS AND SYSTEMS ARE ADEQUATELY SECURED AGAINST UNAUTHORIZED INTRUSION AND (2) BACKING UP YOUR DATA AND FILES.

Standards Compliance

Avaya Inc. is not responsible for any radio or television interference caused by unauthorized modifications of this equipment or the substitution or attachment of connecting cables and equipment other than those specified by Avaya Inc. The correction of interference caused by such unauthorized modifications, substitution or attachment will be the responsibility of the user. Pursuant to Part 15 of the Federal Communications Commission (FCC) Rules, the user is cautioned that changes or modifications not expressly approved by Avaya Inc. could void the user's authority to operate this equipment.

The equipment described in this manual complies with standards of the following organizations and laws, as applicable:

- Australian Communications Agency (ACA)
- American National Standards Institute (ANSI)
- Canadian Standards Association (CSA)
- Committee for European Electrotechnical Standardization (CENELEC) – European Norms (EN's)

- Digital Private Network Signaling System (DPNSS)
- European Computer Manufacturers Association (ECMA)
- European Telecommunications Standards Institute (ETSI)
- FCC Rules Parts 15 and 68
- International Electrotechnical Commission (IEC)
- International Special Committee on Radio Interference (CISPR)
- International Telecommunications Union - Telephony (ITU-T)
- ISDN PBX Network Specification (IPNS)
- National ISDN-1
- National ISDN-2
- Underwriters Laboratories (UL)

Product Safety Standards

This product complies with and conforms to the following international Product Safety standards as applicable:

Safety of Information Technology Equipment, IEC 60950, 3rd Edition including all relevant national deviations as listed in Compliance with IEC for Electrical Equipment (IECEE) CB-96A.

Safety of Laser products, equipment classification and requirements:
- IEC 60825-1, 1.1 Edition
- Safety of Information Technology Equipment, CAN/CSA-C22.2 No. 60950-00 / UL 60950, 3rd Edition
- Safety Requirements for Customer Equipment, ACA Technical Standard (TS) 001 - 1997
- One or more of the following Mexican national standards, as applicable: NOM 001 SCFI 1993, NOM SCFI 016 1993, NOM 019 SCFI 1998

Electromagnetic Compatibility (EMC) Standards

This product complies with and conforms to the following international EMC standards and all relevant national deviations:

Limits and Methods of Measurement of Radio Interference of Information Technology Equipment, CISPR 22:1997 and EN55022:1998.

Information Technology Equipment – Immunity Characteristics – Limits and Methods of Measurement, CISPR 24:1997 and EN55024:1998, including:
- Electrostatic Discharge (ESD) IEC 61000-4-2
- Radiated Immunity IEC 61000-4-3
- Electrical Fast Transient IEC 61000-4-4
- Lightning Effects IEC 61000-4-5
- Conducted Immunity IEC 61000-4-6
- Mains Frequency Magnetic Field IEC 61000-4-8
- Voltage Dips and Variations IEC 61000-4-11
- Powerline Harmonics IEC 61000-3-2
- Voltage Fluctuations and Flicker IEC 61000-3-3

Federal Communications Commission Statement

Part 15:

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

Part 68: Answer-Supervision Signaling. Allowing this equipment to be operated in a manner that does not provide proper answer-supervision signaling is in violation of Part 68 rules. This equipment returns answer-supervision signals to the public switched network when:

- answered by the called station,
- answered by the attendant, or
- routed to a recorded announcement that can be administered by the customer premises equipment (CPE) user.

This equipment returns answer-supervision signals on all direct inward dialed (DID) calls forwarded back to the public switched telephone network. Permissible exceptions are:

- A call is unanswered.
- A busy tone is received.
- A reorder tone is received.

Avaya attests that this registered equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

This equipment complies with Part 68 of the FCC Rules. On the rear of this equipment is a label that contains, among other information, the FCC registration number and ringer equivalence number (REN) for this equipment. If requested, this information must be provided to the telephone company.

The REN is used to determine the quantity of devices which may be connected to the telephone line. Excessive RENs on the telephone line may result in devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed 5.0. To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company.

REN is not required for some types of analog or digital facilities.

Means of Connection

Connection of this equipment to the telephone network is shown in the following table.

Manufacturer's Port Identifier	FIC Code	SOC/REN/ A.S. Code	Network Jacks
Off/On premises station	OL13C	9.0F	RJ2GX, RJ21X, RJ11C
DID trunk	02RV2-T	0.0B	RJ2GX, RJ21X
CO trunk	02GS2	0.3A	RJ21X
CO trunk	02LS2	0.3A	RJ21X
Tie trunk	TL31M	9.0F	RJ2GX
Basic Rate Interface	02IS5	6.0F, 6.0Y	RJ49C
1.544 digital interface	04DU9-BN, 1KN, 1SN	6.0F	RJ48C, RJ48M
120A2 channel service unit	04DU9-DN	6.0Y	RJ48C

If the terminal equipment (for example, the MultiVantage™ Solution equipment) causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If trouble is experienced with this equipment, for repair or warranty information, please contact the Technical Service Center at 1-800-242-2121 or contact your local Avaya representative. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

It is recommended that repairs be performed by Avaya certified technicians.

The equipment cannot be used on public coin phone service provided by the telephone company. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

This equipment, if it uses a telephone receiver, is hearing aid compatible.

Canadian Department of Communications (DOC) Interference Information

This Class A digital apparatus complies with Canadian ICES-003.
Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

This digital apparatus does not exceed Class A limits for radio noise emission set out in the radio interference regulation of the Canadian Department of Communications.

Le Présent Appareil Nomérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils manicures de la class A préscrites dans le reglement sur le brouillage radioélectrique édicté par le ministére des Communications du Canada.

This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

DECLARATIONS OF CONFORMITY

United States FCC Part 68 Supplier's Declaration of Conformity (SDoC)

Avaya Inc. in the United States of America hereby certifies that the equipment described in this document and bearing a TIA TSB-168 label identification number complies with the FCC's Rules and Regulations 47 CFR Part 68, and the Administrative Council on Terminal Attachments (ACTA) adopted technical criteria.
Avaya further asserts that Avaya handset-equipped terminal equipment described in this document complies with Paragraph 68.316 of the FCC Rules and Regulations defining Hearing Aid Compatibility and is deemed compatible with hearing aids.

Copies of SDoCs signed by the Responsible Party in the U. S. can be obtained by contacting your local sales representative and are available on the following Web site:

http://support.avaya.com/elmodocs2/DoC/SDoC/index.jhtml/

All MultiVantage™ system products are compliant with FCC Part 68, but many have been registered with the FCC before the SDoC process was available. A list of all Avaya registered products may be found at:

http://www.part68.org/

by conducting a search using "Avaya" as manufacturer.

European Union Declarations of Conformity

Avaya Inc. declares that the equipment specified in this document bearing the "CE" (*Conformité Europeénne*) mark conforms to the European Union Radio and Telecommunications Terminal Equipment Directive (1999/5/EC), including the Electromagnetic Compatibility Directive (89/336/EEC) and Low Voltage Directive (73/23/EEC). This equipment has been certified to meet CTR3 Basic Rate Interface (BRI) and CTR4 Primary Rate Interface (PRI) and subsets thereof in CTR12 and CTR13, as applicable.

Copies of these Declarations of Conformity (DoCs) signed by the Vice President of MultiVantage™ Solutions research and development, Avaya Inc., can be obtained by contacting your local sales representative and are available on the following Web site:
http://support.avaya.com/elmodocs2/DoC/IDoC/index.jhtml/

Japan

This is a Class A product based on the standard of the Voluntary Control Council for Interference by Information Technology Equipment (VCCI). If this equipment is used in a domestic environment, radio disturbance may occur, in which case, the user may be required to take corrective actions.

この装置は、情報処理装置等電波障害自主規制協議会（ＶＣＣＩ）の基準に基づくクラスＡ情報技術装置です。この装置を家庭環境で使用すると電波妨害を引き起こすことがあります。この場合には使用者が適切な対策を講ずるよう要求されることがあります。

Network Connections

Digital Connections - The equipment described in this document can be connected to the network digital interfaces throughout the European Union.

Analogue Connections - The equipment described in this document can be connected to the network analogue interfaces throughout the following member states:

Belgium	Germany	Luxembourg
Netherlands	Spain	United Kingdom

LASER Product

The equipment described in this document may contain Class 1 LASER Device(s) if single-mode fiber-optic cable is connected to a remote expansion port network (EPN). The LASER devices operate within the following parameters:

- Maximum power output –5 dBm to -8 dBm
- Center Wavelength 1310 nm to 1360 nm
- CLASS 1 LASER PRODUCT IEC 60825-1: 1998

Use of controls or adjustments or performance of procedures other than those specified herein may result in hazardous radiation exposure. Contact your Avaya representative for more laser product information.

To order copies of this and other documents:

Call: Avaya Publications Center
Voice 1.800.457.1235 or 1.410.568.3680
FAX 1.800.457.1764 or 1.410.891.0207

Write: Globalware Solutions
200 Ward Hill Avenue
Haverhill, MA 01835 USA
Attention: Avaya Account Management

E-mail: totalware@gwsmail.com

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

Contents

About This Book

What Is the Purpose of This Book?

This book provides general information about the features and capabilities of Avaya MultiVantage™ software (referred to as MultiVantage™ or the system). It also discusses practical and creative applications for the MultiVantage™ platform.

This document covers all of the features related to MultiVantage™ software. For details about changes for the current release, refer to the *Highlights of Avaya MultiVantage™ Software*.

Who Should Read This Book?

This book is written for those who are considering the purchase of a MultiVantage™ system, and for Avaya representatives and distributors who need high-level information about the system and how it can be used.

What Is in This Book?

This book outlines all MultiVantage™ features and capabilities that are available world-wide.

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

Trademarks and Service Marks

This book contains references to the following Avaya trademarked products:

- AUDIX®
- Callmaster®
- CallVisor®
- CenterVu™
- CONVERSANT®
- DEFINITY®
- GuestWorks®
- INTUITY™
- MULTIQUEST®
- MultiVantage™
- OneVision™
- ProLogix™
- Quorum™

The following are trademarks or registered trademarks of other companies:

- MicroSoft® is a registered trademark of Microsoft Corporation
- Windows™ is a trademark of the Microsoft Corporation
- Vari-A-Bill™ is a trademark of AT&T

How Can I Make Comments About This Book?

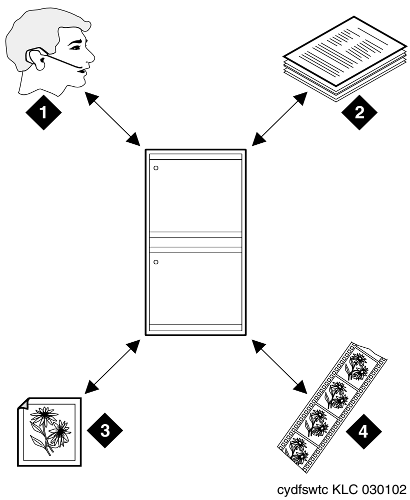
Avaya welcomes your feedback. Your comments are of great value and help improve our documentation.

Fax your comments to 303-538-1741, and mention this document's name and number, *Overview for Avaya MultiVantage™ Software, 555-233-767, Issue 3*.

1 — Overview of Avaya MultiVantage

Avaya MultiVantage software organizes and routes voice, data, image and video transmissions. It can connect to private and public telephone networks, Ethernet LANs, ATM networks, and the Internet.

Avaya MultiVantage software seeks to solve business challenges by powering voice communications and integrating with value-added applications. MultiVantage is an open, scalable, highly reliable and secure telephony application. It provides user and system management functionality, intelligent call routing, application integration and extensibility, and Enterprise Communications networking.



1	Voice	3	Image
2	Data	4	Multimedia

Figure 1. MultiVantage software system

MultiVantage Basic Offering and Advanced Offering

Avaya's MultiVantage software is available as a Basic offering (called "Category B") and as an Advanced offering (called "Category A"). This book describes all of the Advanced MultiVantage features. Some of these features are not available with the Basic offering — which includes DEFINITY BCS and GuestWorks. For a list of features not supported in the Basic offering, see your Avaya representative.

Optional Software

In addition to the Basic or Advanced MultiVantage offerings, various optional packages can enhance the capabilities of the system. Some of the capabilities described in this document require optional software. See your Account Representative for more information.

Capacities

For a table of Avaya MultiVantage software capacities, see the Avaya Web site at http://www.avaya.com/support.

2 — Application Programming Interfaces (API)

Application Programming Intefaces (APIs) allow the interface between the applications and MultiVantage software.

Avaya Computer Telephony (formally named CentreVu Computer Telephony) is server software that integrates the premium call control features of MultiVantage system with customer information in customer's databases. It is a local area network (LAN)-based CTI solution consisting of server software that runs in a client/server configuration. Avaya Computer Telephony delivers the computer telephony integration (CTI) architecture and platform that supports contact center application requirements, along with new emerging applications programming interfaces (APIs).

Adjunct Switch Application Interface (ASAI)

See "Adjunct Switch Application Interface (ASAI)" on page 15.

ASAI Routing

Supplies ASAI routing information to the MultiVantage system.

DAPI

DEFINITY Application Programming Inteface (DAPI) is an object-oriented Application Programming Interface (API) for accessing control and data paths within the MultiVantage software. It enables applications to easily monitor events, inject stimuli, and access switch data using a high-speed LAN connection. It allows development of enhanced debugging tools for Services Engineers, field support, and software developers. DAPI is for internal use only, meaning that any applications created using DAPI are limited to the use of Avaya personnel.

JTAPI

Java Telephony Application Programming Interface (JTAPI) is an open API supported by Avaya Computer Telephony that enables integration to MultiVantage ASAI. It is an object-oriented programming interfaces favored for the development of multimedia solutions. JTAPI applications are supported on any clients that supports a JAVA Virtual machine (this includes Windows, UnixWare, and Solaris platforms), or a Java-compatible Web browser.

TAPI

Telephony Application Programming Interface (TAPI).

TSAPI

Telephony Services Application Programming Interface (TSAPI) is an open API supported by Avaya Computer Telephony that allows integration to MultiVantage ASAI. TSAPI interface supports a wide breadth of application hardware and operating systems, including Windows 95, Windows 98, Windows NT, and UnixWare for clients.

3 — Attendant Features

Accessing the Attendant

Dial Access to Attendant

Allows 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.

Recall

Allows users to recall the Attendant when they are on a two-party call or on an Attendant Conference call held on the console. Single-line users press the Recall button or flash the switchhook to recall the Attendant. Multi-appearance users press the Conference or Transfer button to recall the Attendant and remain on the connection when either button is used.

Attendant Backup

The Attendant Backup feature allows you to access most attendant console features from one or more specially-administered backup telephones. This allows you to answer calls more promptly, thus providing better service to your guests and prospective clients.

When the attendant console is busy, you can answer overflow calls from the backup telephones by pressing a button or dialing a feature access code. You can then process the calls as if you are at the attendant console. The recommended backup telephones are the Avaya Models 6408, 6416, or 6424.

Attendant Room Status

Allows an attendant to see whether a room is vacant or occupied and what the housekeeping status of each room is. This feature is available only when you have Enhanced Hospitality enabled for your system. This feature combines the property management capabilities of Check-In/Check-Out and Housekeeping Status, but does not require that you have a Property Management System.

Attendant with the Distributed Communications System (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.

Automated Attendant

Automated attendant allows the calling party to enter the number of any extension on the system. The call is then routed to the extension. This allows you to reduce cost by reducing the need for live attendants.

Call Handling

Attendant Lockout — Privacy

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.

Attendant Split Swap

Allows the attendant to alternate between active and split calls. This operation may be useful if the attendant needs to transfer a call but first must talk independently with each party before completing the transfer.

Attendant Vectoring

Attendant 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.

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.

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.

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.

Intrusion (Call Offer)

Allows an Attendant to enter an existing call to inform the person being called about a message or another call.

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.

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.

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.

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.

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.

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. MultiVantage supports a variety of administrable attendant timers for use in a variety of situations.

Centralized Attendant Service (CAS)

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.

Making Calls

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.

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.

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.

Monitoring Calls

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.

Crisis Alerts To an Attendant Console

Crisis Alert uses both audible and visual alerting to notify attendant consoles 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 (or room). 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. Though often used in the hospitality industry, it can be set up to work with any standard attendant console.

When crisis alerting is active, the console is placed in position-busy mode so that other incoming calls can not interfere with the emergency call notification. The console can still originate calls to allow notification of other personnel. Once a crisis alert call has arrived at a console, the console user must press the position-busy button to unbusy the console, and press the crisis-alert button to deactivate audible and visual alerting.

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.

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 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.

Trunk Group Busy/Warning Indicators to Attendant

Provides the Attendant with a visual indication that the number of busy trunks in a group has reached an administered level. A visual indication is also provided when all trunks in a group are busy. This feature is particularly helpful to show the Attendant that the Attendant Control of Trunk Group Access feature needs to be invoked.

Trunk Identification By Attendant

Allows an Attendant or display-equipped telephone user to identify a specific trunk being used on a call. This capability is provided by assigning a Trunk ID button to the attendant console or telephone. This feature is particularly helpful for identifying a faulty trunk. That trunk can then be removed from service and the problem quickly corrected.

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.

4 — Call Center

The Avaya Call Center provides a fully integrated telecommunications platform that supports a powerful assortment of features, capabilities, and applications designed to meet all of your customers' call center needs.

Computer Telephony Integration (CTI)

Computer Telephony Integration (CTI) enables MultiVantage to be controlled by external applications, and allows integration of customer databases of information with call control features.

Avaya Computer Telephony (formally named CentreVu Computer Telephony) is server software that integrates the premium call control features of MultiVantage system with customer information in customer's databases. It is a local area network (LAN)-based CTI solution consisting of server software that runs in a client/server configuration. Avaya Computer Telephony delivers the computer telephony integration (CTI) architecture and platform that supports contact center application requirements, along with new emerging applications programming interfaces (APIs).

Adjunct Switch Application Interface (ASAI)

Adjunct Switch Application Interface (ASAI) allows adjunct applications to access a collection of MultiVantage features and services. Integration with adjuncts occurs through APIs. ASAI is part of Avaya Computer Telephony.

CallVisor ASAI

Links MultiVantage and adjunct applications. The interface allows adjunct applications to access MultiVantage features and supply routing information to the system. CallVisor ASAI improves ACD agents' call handling efficiency by allowing an adjunct to monitor, initiate, control, and terminate calls on the switch. CallVisor ASAI may be used for Inbound Call Management (ICM), Outbound Call Management (OCM), and office automation/messaging applications. It uses two transport types: ISDN-BRI transport (CallVisor ASAI-BRI) and LAN Gateway Transmission Control Protocol/Internet Protocol transport (MultiVantage LAN Gateway TCP/IP). CallVisor ASAI messages and procedures are based on the ITU-T Q.932 international standard for supplementary services.

Co-Resident DEFINITY LAN Gateway

Co-Resident DEFINITY LAN Gateway is an application that serves as an ISDN router of ASAI messages through a TCP "tunnel" over an Ethernet LAN.

Direct Agent Announcement

Direct Agent Announcement (DAA) enhances Direct Agent Calling capabilities for CallVisor Adjunct-Switch Application Interface (ASAI) and Expert Agent Selection (EAS). It plays an announcement to Direct Agent callers waiting in a queue.

Flexible Billing

Allows MultiVantage or an adjunct to communicate with the public network using ISDN PRI messages to change the billing rate for an incoming 900-type call. Rate-change requests to specify a new billing rate can be made anytime after a call is answered and before it disconnects.

Flexible Billing is available in the U.S. for use with AT&T MultiQuest 900 Vari-A-Bill™ Service. Flexible billing requires a CallVisor Adjunct-Switch Application Interface and other application software.

Pending Work Mode Change

Allows ASAI applications to change the current work mode of an agent while that agent is busy on a call. The change is a pending change that will take effect as soon as all the current calls are cleared.

Trunk Group Identification

Provides ASAI applications with the capability to obtain Trunk Group information even when the Calling Party Number (CPN) is known. ASAI will provide the Trunk Group information in the Event Reports for both inbound and outbound calls. If the ANI is known, the Event Reports will contain the Trunk Group information and the CPN.

User-to-User Information (UUI) Propagation During Manual Transfer/Conference Operations

This feature enables UUI specifically used by ASAI to be propagated to the new call during a manual transfer or conference operation. Previously, ASAI UUI could not be sent in a SETUP message when the call was transferred to another MultiVantage, so the ASAI UUI was never passed to an application monitoring calls on the MultiVantage receiving the transfer. This feature only applies to manual transfer and conference operations. If the transfer or conference operation is controlled by a software application (for example, controlling calls or agents over an ASAI link), the application can insert the desired ASAI UUI into the new call.

Automatic Call Distribution (ACD)

Automatic Call Distribution (ACD) is the basic building block for Call Center applications. ACD offers you a method for distributing incoming calls efficiently and equitably among available agents. With ACD, incoming calls can be directed to the first idle or most idle agent within a group of agents.

Agents in an ACD environment are assigned to a hunt group, a group of agents handling the same types of calls. A hunt group is also known as a split or skill with EAS.

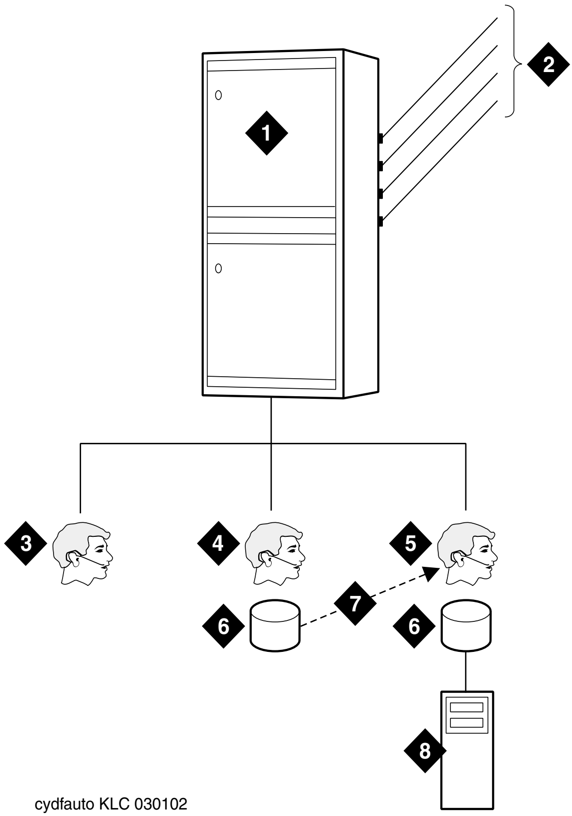
A hunt group is especially useful when you expect a high number of calls to a particular phone number. A hunt group might consist of people trained to handle calls on specific topics. For example, the group might be:

- A benefits department within your company
- A service department for products you sell
- A travel reservations service
- A pool of attendants

In addition, a hunt group might consist of a group of shared telecommunications facilities. For example, the group might be:

- A modem pool
- A group of data-line circuit ports
- A group of data modules

In the Figure 2 on page 19 example, 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 redirected to Hunt Group C if not answered within an administrable time. Calls to Hunt Group C are redirected to voice mail if not answered within an administrable time.



1	MultiVantage	5	Group C: General Information
2	Incoming Lines	6	Queues
3	Group A: Business Travel	7	Call Coverage to Group C
4	Group B: Personal Travel	8	Voice Mail

Figure 2. A Basic Example of Automatic Call Distribution

Abandoned Call Search

Allows a Central Office that does not provide timely disconnect supervision to identify abandoned calls. An abandoned call is one in which the calling party hangs up before the call is answered. Abandoned Call Search is suitable only for older Central Offices that do not provide timely disconnect supervision.

Adjunct Routing

Adjunct Routing is a vector step that when executed sends a route request over the specified link to the connected adjunct asking where to route the call being processed. The adjunct is then to respond with a route-select message specifying the destination either internal or outside number where the call is to be routed. Adjunct Routing is used in conjunction with ASAI.

Auto-Available Split (AAS)

Allows members of an ACD split to be in Auto-In work mode continuously. An agent in Auto-In work mode becomes available for another ACD call immediately after disconnecting from an ACD call. You can use AAS to bring ACD-split members back into Auto-In work mode after a system restart. Although not restricted to such, this feature is intended to be used for splits containing only recorders or voice-response units.

Call Center Release Control

Determines which features are “active” on your switch. Call Center Release Control will control whether certain new Call Center software features are available to you.

Circular Station Hunt Group

This hunt group type is an alternative to the “ddc” or “hot-seat” algorithm in a hunt group. MultiVantage 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.

Avaya Basic Call Management System (BCMS)

The Basic Call Management System helps you fine tune your Call Center operation by providing reports with the data necessary to measure your Call Center agents' performances.

This feature offers call management control and reporting at a low cost for Call Centers of up to 2000 agents. The BCMS collects and processes ACD call data (up to seven days) within the MultiVantage system; an adjunct processor is not required to produce Call Management reports.

The following are the types of reports that 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 report

VuStats

VuStats presents Basic Call Management System (BCMS) statistics on telephone displays. Agents, supervisors, Call Center managers, and other users can press a button and view statistics for agents, splits or skills, VDNs, and trunk groups. These statistics can help agents monitor their own performance or respond appropriately to the caller's request. Features include:

- VuStats Login IDs
- VuStats Service Level

Avaya Call Management System (CMS)

The Avaya Call Management System collects call traffic data, formats management reports, and provides an administration interface for Automatic Call Distribution (ACD) on your MultiVantage. It helps you manage the people, traffic load, and equipment in an ACD environment by answering such questions as:

- How many calls are we handling?
- How many callers abandon their calls before talking with an agent?
- Are all agents handling a fair share of the calling load?
- Are our lines busy often enough to warrant adding additional ones?
- How has traffic changed in a given ACD hunt group over the past year?

CMS Measurement of ATM

Provides the capability to externally measure ATM trunks on CMS. The CMS messages and reports are modified to support the expanded equipment location.

Dual Links to CMS

Provides an additional TCP/IP link to a separate CMS for full, duplicated CMS data collection functionality and high availability CMS configuration. The same data is sent to both servers and the administration can be done from either server. The ACD data will be delivered over different network routes to prevent any data loss from such conditions as ACD link failures, CMS hardware or software failures, CMS maintenance or CMS upgrades.

Site Statistics for Remote Port Networks

Forwards location IDs to CMS to provide Call Center site-specific reports.

Call Prompting

Allows the system to collect information from the calling party and direct the calls via Call Vectoring. The caller is verbally prompted by the system and enters information in response to the prompts. This information is then used to redirect the call or handle the call in some other way (taking a message, for example). This feature is mostly used to enhance the efficient handling of calls in the Automatic Call Distribution application.

Call Center messaging

Call Center messaging gives the calling party the option of leaving a message or waiting in queue for an agent. This may be used for an online order entry system or to further automate an incoming-Call Center operation.

Data Collection

Data collection allows the calling party to enter data that can then be used by a host computer application to assist in call handling. For example, this data may be the calling party's account number, which could be used to support an inquiry/response application.

Data In/Voice Answer (DIVA)

Data In/Voice Answer (DIVA) allows the calling party to hear selected announcements based on the digits that he or she enters. This may be used for applications such as an audio bulletin board.

Call Vectoring

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

Vector commands may direct calls to on-premises or off-premises destinations, to any skill or hunt group, or to a specific call treatment such as an announcement, forced disconnect, forced busy, or music.

With combinations of different vector commands, incoming callers can be treated differently depending on the time or day of the call, the Expected Wait Time (EWT), the importance of the call, or other criteria. Each vector can have up to 32 commands. MultiVantage also allows vectors to be linked via the “goto vector” command.

Advanced Vector Routing

Advanced Vector Routing is a collection of features that enhance MultiVantage Vector routing capabilities.

Average Speed of Answer (ASA) Routing

Average Speed of Answer (ASA) routing is an enhancement to call vectoring that provides a flexible method for routing calls or queuing calls based on their average speed of answer for a VDN or a split/skill.

Best Service Routing (BSR)

Best Service Routing (BSR) distributes the call to the best local or remote split/skill among the resources to be considered, based on Expected Wait Time (EWT) or available agent characteristics.

Best Service Routing (BSR) Polling Over IP Without B Channel

Best Service Routing (BSR) Polling Over IP Without B Channel provides the ability to do BSR polling between multiple site over H.323 IP trunks without requiring an ISDN PRI B-channel. This also eliminates the associated IP Media Processor hardware. QSIG Temporary Signaling Connections are used by the BSR polling software to eliminate the need for the IP Media Processor board, thereby making BSR an even more cost effective multi-site solution.

Expected Wait Time Routing

Expected wait time makes call center routing decisions based on waiting time for calls in queue, using a patented algorithm that continuously estimates call waiting times. Announcements of expected wait time can make customers' time in queue more comfortable.

Holiday Vectoring

With Holiday Vectoring, a flexible approach for managing incoming calls on special dates is available. Holiday Vectoring allows for branching and routing of calls based on information about special schedules. The special schedules are recorded in tables, each of which can hold up to 15 special dates or ranges of dates. Holiday Vectoring makes it possible for up to 10 tables to be treated differently in vector processing.

Vector Directory Numbers (VDN)

Calls access MultiVantage vectors using Vector Directory Numbers (VDN). A VDN is a “soft” extension number that is not assigned to a physical equipment location. A Vector Directory Number has several properties that are administered by the System Manager.

A Vector Directory Number can be accessed in almost any way that an extension can be accessed.

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

Class of Restriction (COR) for VDN

Class of Restriction is checked for transfer to the VDN. It can also be used to block the AUX Trunk announcement from some Agents. Observing can also be set to allow or restrict to that VDN.

Display VDN for Route-to DAC

Display VDN for route-to DAC provides a VDN option to have the display to the answering agent show the “caller to VDN” format. The option for the “caller to VDN” display is required for ACD applications where a call needs to be routed to a specific agent and have the call go to coverage if the agent doesn't answer or is logged out.

VDN in a Coverage Path

VDN in a Coverage Path enhances Call Coverage and Call Vectoring to allow you to assign Vector Directory Numbers as the last point in coverage paths. Calls that go to coverage can be processed by vectoring/prompting to extend Call Coverage treatments.

VDN of Origin Announcement

VDN of Origin Announcement provides agents with a short message about a caller's city of origin or requested service based on the VDN used to process the call. VOA messages help agents respond appropriately to callers. For example, if you have two 800 numbers, one for placing orders and one for technical support, you can administer two VDNs to route calls to the same set of agents. When an incoming call is routed to a VDN with a VOA assigned (for example, "new order" or "tech help"), the VDN routes the call to a vector, which can place the call in an agent queue. When an agent answers the call, he or she hears the VOA message and can respond appropriately to the caller's request. This feature is particularly useful for visually impaired agents or agents that don't have display sets.

VDN Return Destination

VDN Return Destination is an optional feature that re-routes a call that has been processed through a vector, to the administered Return Destination. This step occurs once all parties, except the originator, have dropped. The Return Destination must be a VDN extension.

Call Work Codes (CWC)

Call Work Codes (CWC) allows ACD agents to enter up to digits for an ACD call to record the occurrence of a customer defined event such as a social security numbers or phone numbers. The agent enters the Call Work Code by operating the CWC feature button and using the dial pad during an ACD (inbound) call without interrupting the conversation, or in the After Call Work (ACW) mode following the call. The digits are displayed on a display equipped voice terminal while being entered.

Avaya Business Advocate

Avaya Business Advocate is the collection of features that provide new flexibility in the way a call is selected for an agent in a call surplus situation, and in the way an agent is selected for a call. Instead of the traditional "First-In, First-Out" approach, The caller's needs, potential business value, and their desire to wait are looked at, and then the system decides what agents should be matched to the callers.

Advocate-Related Enhancements

Auto Reserve Agents

Allows the system to use the Percent Allocation Distribution feature for agent skills.

Call Selection Override per Skill

Call Selection override is determined by skill. Call Center Supervisors can override the normal call handling activity on particular skills only, or for the entire Call Center.

Dynamic Percentage Adjustment

Allows the system to compare actual levels of service with service targets. The system can then adjust the service target so that the overall use of the skill is more efficient.

Dynamic Queue Position

Allows the system to put calls from multiple vector directory numbers (VDNs) into a skill queue based on the ratio of ASA for the VDNs being equal to the ratio of service objectives for the VDNs. This feature ensures balanced call handling across VDNs.

Dynamic Threshold Adjustment

Allows the system to compare actual levels of service with service targets and adjust overload thresholds. This feature makes the use of overload agents more efficient.

Least Occupied Agent (LOA)

Distributes the calls evenly across all available agents to balance the workload among the agents with fewer skills and agents with several skills. LOA solves the problem of agents who were bombarded with calls after logging into a skill at the start of a shift, while the agents who are already logged-in have maintained their current incoming call level.

Logged-In Advocate Agent Counting

Counts agents toward the Advocate agent limit if Service Objective, Percent Allocation, or a Reserved Skill is assigned to the agent’s login ID, or if one of the agent’s skills is assigned Least Occupied Agent or Service Level Supervisor.

Percent Allocation Distribution

Allows the system to distribute calls to auto reserve agents by comparing a reserve agent’s work time in a skill with the target allocation for the skill.

Reserve Agent Time in Queue Activation

This feature activates a reserve agent if a skill’s expected wait time (EWT) exceeds a pre-determined threshold or if the call’s time in the queue exceeds the administered Service Level Supervisor threshold. Reserve agents are then dropped off a skill only when both of the following conditions are met:

- The EWT for the skill drops below both administered thresholds.
- The head call’s time in queue no longer exceeds the Service Level Supervisor threshold.

Dialed Number Identification Service (DNIS)

Displays, for a called party or answering position, the service or product associated with an incoming call. You administer what the system displays.

Direct Agent Calling

Direct agent calling lets the customer's callers go directly to the same agent whenever they call, automatically, for prompt, personalized service. These direct-to-the-agent calls are also included in their call center measurement statistics.

Duplicate Agent Login ID Administration

Duplicate agent login ID administration simplifies administration of similar agent login ID forms.

Expert Agent Selection (EAS)

Enables certain Expert Agent Selection skill types to be assigned to a call type or a Vector Directory Number. Routing calls via vectoring then allows the system administration to direct calls to agents who have the particular agent skills required to complete the customers’ inquiries successfully.

Add/Remove Skills

Allows an agent using Expert Agent Selection to add or remove skills. A skill is a numeric identifier that refers to an agent's specific ability. For example, an agent who speaks English and Spanish could be assigned a language -speaking skill with an identifier of 20. The agent then adds skill 20 to his or her set of working skills. If a customer needs a Spanish-speaking agent, the system routes the call to an agent with that skill. Each agent can have up to four active skills, and each skill is assigned a priority level.

Call Distribution Based on Skill

Calls that require certain agent skills (such as "speaks Spanish" or "knowledgeable about Product X") can be matched to an agent who matches the required skill. You can assign one of up to 999 skill numbers to each need or group of needs. The skills are administered and associated for each of the following:

- Vector Directory Numbers
- Agent Login IDs
- Callers

This refined skill definition capability allows you to organize call handling based on customer, product, and language, for example.

Queue to Best ISDN Support

Queue to Best information is passed transparently over several public networks and QSIG private networks using the envelopes that are part of the QSIG Manufacturer-Specific Information (MSI) and the ISDN platform enhancement.

Avaya Virtual Routing

Avaya Virtual Routing (formerly known as Look-Ahead Interflow or LAI) balances the load of ACD calls across multiple locations. Virtual Routing helps customers balance call loads among their locations by analyzing demand and directing each call to the location best able to handle it — for example, based on call volume, waiting time in queue, or the time of day.

With Avaya Virtual Routing, you can optionally route a call to a backup location based on your system's ability to handle the call within parameters defined in a vector. In turn, the backup system can accept or deny the call also based on defined parameters.

Avaya Virtual Routing allows interflowing only the call(s) at or near the head of the queue to provide First In/First Out (FIFO) or FIFO-like call distribution and significantly reduce call and trunk processing for Avaya Virtual Routing.

Enhanced Information Forwarding

Enhanced information forwarding allows Call Center related information to be passed transparently over some public networks and non-QSIG or QSIG private networks using codeset 0 shared user-to-user information (UUI) (for non-QSIG) or QSIG Manufacturer-Specific Information (MSI). For more information on UUI, see ''User-to-User Information (UUI) Over the Public Network''.

Multiple Call Handling (Forced)

Allows agents to receive an ACD call while other types of calls are alerting, active, or on hold.

Queue Status Indications

Allows you to assign Queue-Status Indicators for ACD calls based on the number of calls queued and time in queue. You can assign these indications to lamps on agent, supervisor, or attendant terminals or consoles to help monitor queue activity. In addition, you can define auxiliary queue warning lamps to track queue status. On display telephones, you can display the number of calls queued and time in queue of a split's oldest call.

Multiple Split Queuing

Multiple split queuing lets customers direct a call to several splits at the same time, so that the first available agent can take the call. It can help customers handle the busiest periods with greater ease and provide faster service to their callers.

Priority Queuing

Priority queuing allows special callers to be assigned "priority status" and routed ahead of other callers. Clients can pamper their best customers with the fastest attention possible.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering Auxiliary (AUX) work mode or for logging out of the system. Reason codes give Call Center managers detailed information about how agents spend their time. You can use this data to develop more precise staffing forecasting models or use it with schedule-adherence packages to ensure that agents are performing scheduled activities at the scheduled time. You must have Expert Agent Selection (EAS) enabled to use reason codes.

Redirection on No Answer

Redirects a ringing ACD split or skill call or Direct Agent Call after an administered number of rings. This prevents an unanswered call from ringing indefinitely. The call can redirect either to the split or skill to be answered by another agent or to a Vector Directory Number (VDN) for alternative call handling. Direct Agent Calls route to the agent's coverage path, or to a VDN if no coverage path is administered. You must have ACD enabled to use this feature.

Miscellaneous

Caller Information Forwarding (CINFO)

The Avaya Call Center also supports AT&T Caller Information Forwarding (CINFO) service, allowing customers to collect customer-provided data forwarded through the network. This information can be used to route calls or provide visual displays on agent voice terminals, or be passed along to computer telephony integration (CTI) applications.

Multiple Music/Audio Sources

Multiple music/audio sources lets customers deliver music or customized announcements to callers while they are in queue, helping to make the waiting time more productive or entertaining. Customers can provide information about their products, services, other call center applications, offer public service information, or play music.

Network Call Redirection (NCR)

Today, call center customers are looking for many ways to reduce their costs. One of these ways is to employ Public Switched Telephone Network (PSTN) Virtual Private Networks (VPNs) to eliminate as much private network cost as possible. These cost reductions are particularly valuable in enterprises or multi-site call-center environments and especially to Enterprise call centers where network costs are typically high.

Network Call Redirection (NCR) offers a call redirection method between sites on a public network or a PSTN Virtual Private Network, to help reduce trunking costs. NCR may only be activated for incoming ISDN trunk calls where the associated trunk group has been enabled by the public network service provider to use Network Call Transfer or Network Call Deflection features.

PC Application Software Translation Exchange (PASTE)

Allows users to view call center data on display phones, displaying what each terminal button is, and what the feature access codes for the switch are. PASTE is used in conjunction with Avaya IP Agent.

Remote Logout of Agent

Remote logout of agents allows a select set of users to log out an agent using a feature access code.

Service Observing

Allows a specified user, such as a supervisor, to observe or monitor another user’s calls. A vector directory number call can also be observed. Observers can observe in listen-only or listen-and-talk mode. You set up Service Observing to observe a particular extension, not all calls to all extensions at a terminal.

> **NOTE:**
> Service Observing may be subject to federal, state, or local laws, rules, or regulations or require the consent of one or both of the call parties. Familiarize yourself and comply with all applicable laws, rules, and regulations before using this feature.

Service Observing by COR

Restricts certain users from using the Service Observing feature.

Service Observing of VDNs

Service observing of VDNs (also known as VDN Observing on Agent Answer) allows a supervisor to start observing a call to the VDN when the call is delivered to the agent station. The observer will not hear the call during vector processing (announcements, music, etc.).

Service Observing Remote

This option will allow observing from non-feature button equipped stations. An observer will be able to monitor a VDN or a physical extension remotely using an “observe FAC” procedure through the Remote Access feature and/or Call Vectoring/Call Prompting features (via VDNs).

Vector Initiated Service Observing

Vector Initiated Service Observing, also called VDN Observing on Agent Answer, allows users to start observing of a call to the VDN when the call is delivered to the agent or station. This saves time for the observer after observing of the VDN has been activated since the observer does not have to wait listening for each subsequent call to go through vector processing and for the agent to answer.

User-to-User Information (UUI) Over the Public Network

Provides the mechanism to pass information across several key public networks, including information that is originated or destined for one of several applications on MultiVantage.

Voice Response Integration (VRI)

Integrates Call Vectoring with the capabilities of voice response units such as the Avaya CONVERSANT Voice Information System. You can also integrate a voice response unit with ACD. All this provides a variety of advantages. For example, while a call is queued, a caller can listen to product information via an audiotext application or can complete an interactive voice-response transaction. It may be possible to resolve the caller's questions while the call is queued, which helps reduce queuing time for other callers during peak times.

5 — Collaboration

MultiVantage contains a variety of features aimed to provide individuals easy ways to collaborate with groups of peers, customers, and partners such as executives, sales people, and professional specialists. These key work groups require a high level of effective interaction.

Avaya Unified Communication Center

The Avaya Unified Communication Center powers productivity and responsiveness through seamless access to critical communications tools, business applications, content, and transactions throughout the business day. It simplifies the way users connect, collaborate, share information, and manage their time — wherever they are located or whatever device they use.

Whether through a Web browser, a wireless device, or a speech command, Avaya Unified Communication Center allows users to seamlessly access rich calling and conferencing capabilities, including MultiVantage features, for easy collaborative working sessions.

Conferencing

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 is now on hard-hold.

Conference/Transfer Display Prompts

Conference/Transfer Display prompts are based on the user's COR. The display prompts are based on the user's COR independently of the Select Line Appearance Conferencing and No Dial Tone Conferencing feature. The display messages vary depending on the activation of the two features, but the choice of displaying the additional information or not is dependent on the station user's COR.

Conference — Three Party

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

Conference — Six Party

The Conference button allows multi-appearance telephone users to make up to six-party conference calls without attendant assistance.

Conference/Transfer Toggle/Swap

The Conference/Transfer Toggle/Swap feature allows users to toggle between two parties in the middle of setting up a conference prior to connecting all parties together, or to consult with both parties prior to transferring a call. The display also toggles between the two parties.

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 Telephones.

Hold/Unhold Conference

Allows user to use the HOLD button to bring the held party back to the conversation. This is an alternative to using the line appearance button. Hold/Unhold only applies if there is only one line on hold and no other line appearances are active. An error message is displayed if the “unhold” feature is attempted when not allowed. This feature is not available for BRI stations or attendant consoles.

Meet-me Conference

The Meet-me Conference feature allows a person to set up a dial-in conference of up to six parties. The Meet-me Conference feature uses Call Vectoring to process the setup of the conference call. Meet-me Conference can be optionally assigned to require an access code. If an access code is assigned, and if the vector is programmed to expect an access code, each user dialing in to the conference call must enter the correct access code to be added to the call. The Meet-me Conference extension can be dialed by any internal or remote access users, and by external parties if the extension number is part of the customer’s DID block.

No Dial Tone Conferencing

This feature can eliminate user confusion over receiving dial tone when trying to conference two existing calls. It skips the automatic line selection if there is already a party on hold or an alerting line appearance. Help messages help guide the user. This feature is assigned on a system wide basis.

Select Line Appearance Conferencing

If you are in a conversation on line “b”, and another line is on hold or an incoming call is alerting on line “a”, then pressing the CONF button bridges the calls together. Using the Select Line Appearance feature on MultiVantage, the user has the option of pressing a line appearance button to complete a conference instead of pressing CONF a second time.

This feature only applies if the line is placed in soft hold by pressing the CONF button. This feature never applies if the soft hold was due to pressing a TRANSFER button.

Selective Conference Party Display, Drop, and Mute

The Selective Conference Party Display, Drop, and Mute feature allows any user on a digital station with display or on an attendant console to use the display to identify all of the other parties on a two-party or conference call. The user would press a feature button while on the call that puts the station or console into conference display mode. The user then can scroll through the display of each party currently on the call by repeatedly pressing the feature button. The display would show the party's number and name (when available).

The user could then do either of the following:

- The user can selectively drop the party currently shown on the display with a single button push. This can be useful during conference calls when adding a party that does not answer and the call goes to voice mail.
- The user can selectively mute the party currently shown on the display with a single button push. This puts the selected party in “listen-only” mode. This can be useful during conference calls when a party puts the conference call on hold and everyone on the call is forced to listen to music-on-hold. The user can mute that party so the conference call can continue without interruption. The muted party can then rejoin the call by pressing the # key on their telephone.

CAUTION:

Station users must be careful when scrolling through the displays when using the Selective Conference Party Display feature. The station hyperactivity feature will take the station out of service if the user repeatedly scrolls through the displays at high enough rates. This will cause the station to be reset and the user will be dropped from the call.

Multimedia Calling

Multimedia calls are initiated with voice and video only. Once a call is established, one of the parties may initiate an associated data conference to include all of the parties on the call who are capable of supporting data. The data conference is controlled by an adjunct device called an Expansion Services Module (ESM).

Multimedia Applications Server Interface (ASI)

The Multimedia Applications Server Interface provides a link between the MultiVantage and one or more Multimedia Communications eXchange nodes. A Multimedia Communications eXchange is a stand-alone multimedia call processor produced by Avaya. This new link to MultiVantage enhances the capabilities of each Multimedia Communications eXchange system by enabling it to share some of the MultiVantage features. In particular, the interface provides the following advantages:

- Call Detail Recording (CDR)— The capture of call detail records so you can analyze the call patterns and usage of multimedia calls just as MultiVantage administrators analyze normal calls.
- Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) — The intelligent selection of the most cost-effective routing for calls, based on available resources and your carrier preference. The system may select public trunks via DEFINITY Multimedia eXchange (MMCX)
- Voice Mail Integration — You can access your EMBEDDED AUDIX or Intuity AUDIX voice messaging system from a Multimedia Communication eXchange (MMCX).

Multimedia Call Early Answer on Vectors and Stations

Early Answer is a feature applied to multimedia calls in conjunction with conversion to voice. Early Answer:

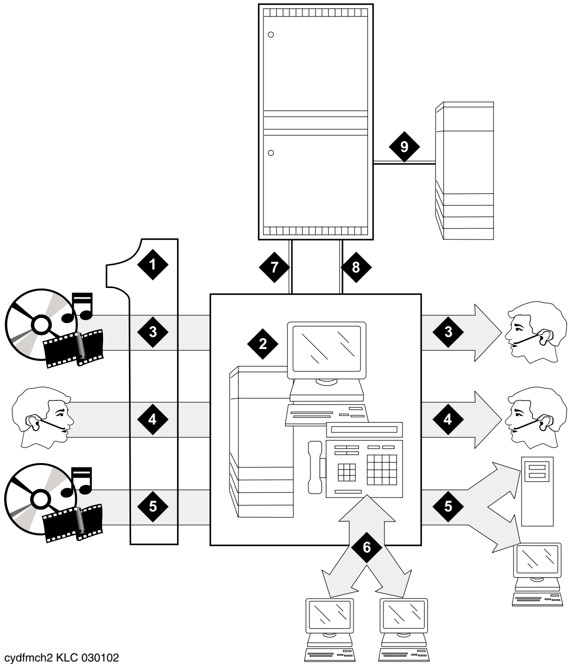
- Answers the data call
- Establishes the multimedia protocol prior to completion of a converted call
- Ensures that a voice path to/from the originator is available when the (voice) call is answered

For an incoming call, Early Answer answers the dynamic service-link calls when the destination endpoint answers, unless Early Answer is specified during routing or termination processing.

NOTE:
The “destination voice endpoint” might be an outgoing voice trunk if the destination voice station is forwarded or covered off-premises.

Multimedia Call Handling (MMCH)

Multimedia Call Handling (MMCH) enables you to control voice, video, and data transmissions using your telephone set. The feature buttons on a multi-function MultiVantage telephone enable you to conduct video conferences, and forward, cover, hold, or park multimedia calls much as you would a standard voice call. You can also share PC applications so that you and colleagues can collaborate while working from remote sites. See Figure 3 on page 41.



1	One number access	5	Call redirection
2	Multimedia call complex	6	Multimedia conferencing
3	Multimedia to voice conversion	7	BRI data connection
4	Standard voice call handling	8	DCP voice connection
		9	ESM data collaboration

Figure 3. MultiVantage Multimedia Call Handling

Multimedia Call Redirection to MM Endpoint

A dual port multimedia station may be a destination of call redirection features such as call coverage, forwarding, and station hunting. The station can receive and accept full multimedia calls or data calls converted to multimedia.

Multimedia Data Conferencing (T.120) via ESM

The data conference is controlled by an adjunct device called an Expansion Services Module (ESM). The Expansion Services Module is used to terminate T.120 protocols [including Generalized Conference Call (GCC), a protocol standard for data conference control] and provide data conference control and data distribution. The MultiMedia Interface circuit pack, TN787, is used to rate adapt T.120 data to/from the ESM.

Multimedia Hold, Conference, Transfer, and Drop

Station users have the ability to activate hold, conference, transfer, or drop on multimedia calls. Multimedia endpoints and voice-only stations may participate in the same conference.

Multimedia Queuing with Voice Announcement

When multimedia callers queue for an available member of a hunt group they are able to hear an audio announcement.

Paging and Intercom

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.

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.

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 — Automatic Answer

Automatic Answer Intercom Calls (Auto Answer ICOM) allows a user to answer an intercom call within the intercom group without pressing the intercom button. Auto Answer ICOM works with digital, BRI, and hybrid phones with built-in speaker, headphones, or adjunct speakerphone.

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.

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.

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.

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.

6 — Communication Device Support

2420 DCP Telephones

The 2420 is a new digital telephone with an optional feature expansion module and downloadable call appearance/feature buttons information. The 2420 DCP phone does not need paper labels. The button information appears on a screen on the phone. The firmware for the 2420 phone can be changed remotely.

The 2420 telephone uses icons to indicate the status of call appearances, bridged call appearances, and features.

4600-Series IP Telephones

The 4600 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. This series of telephones includes the 4602, 4606, 4612, 4620, 4624, and 4630 models.

The 4630 IP Screenphone uses a large color, touch-sensitive screen to operate the telephone functions. The 4620 IP telephone has downloadable call appearance/feature buttons information that eliminates the need for paper labels.

6200-Series Analog Telephones

The 6210, 6211, 6218, 6219, 6220, and 6221 two-wire, analog telephones are designed to take advantage of the many features offered by MultiVantage. They offer the following features.

- Message light
- Flash and redial buttons
- Hold button and hold light
- Handset volume control
- Data jack (for connecting a modem or similar device)

Personalized ringing, speakerphone button and light, and programmable dialing buttons (6220 only)

6400-Series DCP Telephones

The two-wire, DCP 6400 digital telephones are similar to the 8400 telephones and feature global styling and a pullout instruction card. The 6400 telephones also include the following additional features:

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

6400 Tip/Ring Interface Module

This module provides a two-wire analog interface for the 6400 DCP telephones. This allows the operation of an analog adjunct to be independent of the digital telephone's extension for the use of FAX machines or modems without compromising the user's voice extensions.

8400-Series Telephones

The 8400 digital telephones are versatile two-wire/four-wire Digital Communications Protocol (DCP) telephones. They automatically detect whether they are plugged into a two-wire or four-wire digital line circuit card.

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.

Avaya IP Agent

This is a PC-based IP application that allows agents to use their PCs as phones. In addition to the traditional functionality of a standard MultiVantage phone (transfer, hold, conference, and so forth), IP Agent offers directory services, screen pops, call history, and agent mode history.

Avaya IP Softphones

Avaya IP Softphones extend the level of MultiVantage 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.

Avaya SoftConsole

The Avaya SoftConsole is a Windows-based GUI application that can replace the physical 302B “hard” console. It allows attendants to perform call answering and routing through a PC interface via IP.

DEFINITY AnyWhere

DEFINITY AnyWhere gives you remote access to the powerful voice and data capabilities of your MultiVantage system.

MultiVantage provides powerful voice features and data collaboration capabilities in your office. With DEFINITY AnyWhere, you can have the same functionality when you are working at your virtual office, traveling, or in your hotel room.

DEFINITY AnyWhere is a software application that installs and runs on a Microsoft Windows NT server within your intranet. It provides “single number” accessibility by redirecting calls to any remote phone number. With DEFINITY AnyWhere, your customers and colleagues will appreciate that you are accessible at one number and never out of reach.

EC500 Extension to Cellular

The Avaya EC500 Extension to Cellular provides the expansion of mobile services, including one-number availability, increased user capacities, flexibility across facilities and hardware, more control over unauthorized usage, enhanced enable/disable capability, increased serviceability, and support of IP trunk facilities.

For more information, see ‘‘EC500 Extension to Cellular’’ on page 71.

Avaya MultiVantage PC Console

The Avaya MultiVantage 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.

7 — Hospitality

Alphanumeric Dialing

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

Attendant Room Status

See "Attendant Room Status" on page 6.

Automatic Selection of Direct Inward Dialing (DID) Numbers

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

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

Automatic Wakeup

Allows attendants, front desk users, and guests to request that one or two wake-up calls be placed automatically to a certain extension number at a later time. When a wake-up call is placed and answered, the system can provide a recorded announcement (which can be a speech synthesis announcement), music, or simply silence. With the Integrated Announcement feature, multiple announcements enable international guests to use wake-up announcements in a variety of languages. See also Daily Wakeup, Dual Wakeup, and VIP Wakeup.

Check-In/Check-Out

Allows front desk personnel to check guests into a hotel and, when the guests leave, check them out. There are two ways this is done: through the PMS terminal or through the attendant console (or backup telephone). Check-in and check-out from the attendant console should be used only if there is no PMS, or if the link to the PMS is down. If the PMS is installed and working, check guests using the PMS.

For guest check-in or check-out from the console, there are two buttons on the attendant console (or backup telephone): one labeled “Check In” and the other labeled “Check Out.” The check-in procedure performs two functions: it deactivates the restriction on the telephone in the room allowing outward calls, and it changes the status of the room to occupied.

Custom Selection of VIP DID Numbers

This feature builds on the Automatic Selection of DID Numbers feature. It allows hotel personnel to control what DID number is assigned to a hotel room at check-in. That is, the system asks the user to specify the desired DID number when a guest is checked in. The number comes from a new pool of DID numbers that are separate from those used by the Automatic Selection feature. The system never automatically assigns numbers from this new pool. Numbers from this new pool are used only when explicitly specified by the user.

Daily Wakeup

Allows a guest or front desk personnel to schedule a single wakeup request for a daily wakeup call. For example, if a guest needs to receive a wakeup call at 5:30 a.m. for the duration of his or her stay, one request can be placed on the system instead of placing a separate request for each day.

Dial-by-Name

The Dial-by-Name feature allows callers to the system to access guest rooms simply by dialing the name of the guest they are trying to contact. This feature uses recorded announcements and the Call Vectoring feature to set up an automatic attendant procedure. This automatic attendant procedure gives callers the ability to enter a guest’s name. When a single or unique match is found, the call is redirected to the guest’s telephone.

Do Not Disturb

Allows guests, Attendants, and authorized front desk users to request that no calls, other than priority calls, be connected to a particular extension until a specified time.

Dual Wakeup

Allows guests to have two separate wakeup calls. The Dual Wakeup feature is an enhancement to the standard Automatic Wakeup feature used in hospitality environments. With the standard wakeup feature, guests or front desk personnel can create one wakeup call per extension. The Dual Wakeup feature allows guests and front desk personnel to create either one or two wakeup calls. The Dual Wakeup feature for guests is valid only when the system is not equipped with a speech synthesizer circuit pack.

Housekeeping Status

Records the status for up to six housekeeping codes and reports them to the property management system. These status codes are usually entered by the housekeeping staff from the guest room or from a designated telephone, but they can also be updated by the front office personnel using the attendant console or a backup telephone. Six status codes can be used from guest rooms, and four status codes can be used from telephones that do not have the client room Class of Service.

Names Registration

Automatically sends a guest’s name and room extension from the Property Management System (PMS) to the switch at check-in, and automatically removes this information at check-out. The information may be displayed on any attendant console or display-equipped telephone at various hotel locations (for example, Room Service or Security).

Property Management System (PMS) Digit to Insert/Delete

Many customer configurations base the room telephone extension on the room number by adding an extra leading digit. The PMS Insert/Delete Digit feature allows users to delete the leading digit of the extension in messages. The feature is useful for a hotel that has multiple extensions sharing an extra leading digit in front of the room number. The leading digit is automatically inserted when the message goes to the switch.

The PMS interface supports 3-, 4-, or 5-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, you should not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Property Management System (PMS) Interface

The Property Management System allows a customer to control features used in both a hospital-type and a hotel/motel-type environment. The communications link allows the Property Management System to interrogate the switch and allows information to be passed between the switch and the Property Management System. The switch exchanges guest status information (room number, call coverage path, and other data) with the Property Management System.

There are two ways that the guest data can be encoded:

- Using a combination of Binary Coded Decimal (BCD) encoding and the ASCII character set
- Using only the ASCII character set

Single-Digit Dialing and Mixed Station Numbering

Allows hotel staff and guests easy access to internal hotel/motel services and provides the capability to associate room numbers with guest room telephones. The feature provides the following dial plan types: single-digit dialing, prefixed extensions, and mixed numbering.

Suite Check-In

Suite Check-In allows more than one station to be “checked-in” at one time. This is useful for a guest room that may have multiple extensions. This feature allows all extensions to be “checked-in” at the same time. Suite Check-In via the Hunt-to feature will also “check-out” all the extensions in the entire suite at the same time.

VIP Wakeup

Allows front desk personnel to provide personalized wakeup calls to important guests. When a wakeup call has been scheduled for an important guest, a wakeup reminder call is placed to the front desk personnel, who in turn call the guest personally to provide the wakeup call.

Wake-Up Activation via Confirmation Tones

If a speech synthesizer circuit pack is not installed, guests can still enter their own wake-up calls (two if the Dual Wakeup feature is active). The guests do not receive voice prompts as they would using the speech synthesizer circuit pack; instead, guests will receive call progress tones (recall dial tone and confirmation tone) to set up their wake-up calls.

Xiox Call Accounting

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

8 — Localization

Administrable Language Displays

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

Administrable Loss Plan

The Administrable Loss Plan provides the ability to administer signal loss and gain for telephone calls. This capability is necessary because the amount of loss allowed on voice calls can vary by country. With the Administrable Loss Plan feature, switch endpoints are classified into 17 endpoint types, and the loss plan can be administered for trunks, stations, and personal CO lines. Loss values are in the range of 15 dB loss to 3 dB gain. Preset defaults are available and are based on country type.

Bellcore Calling Name ID

Allows the system to accept calling name information from a local exchange carrier (LEC) network that supports the Bellcore calling name specification. The system can send calling name information in the format if Bellcore Calling Name ID is administered. The following Caller ID protocols are supported:

- Bellcore (default) - US protocol (Bellcore transmission protocol with 212 modem protocol)
- V23-Bell - Bahrain protocol (Bellcore transmission protocol with V.23 modem protocol).

Block Collect Call

Blocks collect calls on class-of-restriction basis. This feature is available for any switch that uses the Brazil country code. If enabled for a station, all trunk calls that terminate to the station will send back a double answer to the CO. This double answer tells the CO that this particular station cannot accept collect calls. The CO then tears down the call if it is a collect call.

Busy Tone Disconnect

In some regions of the world the CO sends a busy tone for the disconnect message. With Busy Tone Disconnect, the switch disconnects analog loop-start Central Office trunks when a busy tone is sent from the CO.

E&M Signaling — Continuous and Pulsed

Provides continuous and pulsed E&M signaling. Continuous and pulsed E&M signaling is a modification to the E&M signaling used in the United States. Continuous E&M signaling is intended for use in Brazil, but can also be used in Hungary. Pulsed E&M signaling is intended for use in Brazil.

Distributed Communications Systems (DCS) Protocol — Italy

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

ISDN/DATS Network Support — Russia

This feature supports ISDN/DATS trunk networks when the tone generated field is set to 15 (Russia) on the system-parameters country-options screen. It modifies the overlap sending delay and ISDN T302 and T304 timers to support the Russian trunk network.

Multi-Frequency Packet (MFP) Signaling — Russia

Multi-Frequency Packet (MFP) address signaling is provided in Russia on outgoing CO trunks. Calling party number and dialed number information is sent on outgoing links between local and toll switches. Russian MFP is set on each trunk group on the Type field on the trunk screen. Russian MFP does not apply to PCOL trunks.

National Private Networking Support — Japan

Provides support for Japanese private ISDN networks. The Japanese private network ISDN protocol is different from the standard ISDN protocol. The switch now supports extensions to the ISDN protocol for switches using the Japanese country code.

Public Network Call Priority

Provides call retention, forced disconnect, intrusion, mode-of-release control, and re-ring to switches on public networks. Different countries frequently refer to these capabilities by different names.

World Class Tone Detection

Enables the MultiVantage to identify and handle different types of call progress tones, depending on the system administration. You can use the tone detector and identification to display on Data Terminal Dialing and to decide when to send digits on trunk calls through Abbreviated Dialing, ARS, AAR, and Data Terminal Dialing.

9 — Message Integration

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.

Centralized Voice Mail via Mode Code Integration

The Centralized Voice Mail feature eliminates the need for a voice mail system at each of the sites in a network. It does so by allowing a MultiVantage network to use a single Intuity AUDIX or Octel 100 Voice Messaging System as a centralized voice mail system that serves the whole network. The Intuity AUDIX or Octel 100 system can also serve as a centralized voice mail system within a hybrid network of MultiVantage, DEFINITY BCS, and Merlin Legend/Magix switches.

Dual DCP I-Channels

Support the use of dual DCP I-channels for AUDIX networking. In this case, networking refers to the ability to send data files between AUDIX systems, not to communications with the switch.

Intuity AUDIX

Intuity Messaging Solutions essentially offers the same user features as the EMBEDDED AUDIX System, plus the following features:

- *Fax Messaging* allows you to handle faxes as easily as you handle voice mail. You can send, receive, store, scan, delete, skip, or forward faxes. This feature is fully integrated with voice messaging, so you can attach faxes to voice messages, for example. You can also create special mailboxes for each of your fax machines. These mailboxes accept fax telephone calls when the fax machine is busy and then deliver the fax to the fax machine when the fax machine is available
- *Turn off AUDIX Call Answering* allows you to turn off Call Answering in order to conserve system resources. You can create a message that tells callers they cannot leave a message, giving them another number to call, for example
- *Pre-Addressing* allows you to address a message before recording it
- *Integrated Messaging* allows you access and manage incoming voice, fax, and e-mail messages and file attachments from your personal computer or your telephone. A voice message will thus appear in your e-mail mailbox, for example, and vice versa. You can also set options to have just the message headers appear in the alternate mailbox. You can also create a voice or fax message by telephone and send it to an e-mail recipient
- *Text-to-Speech* allows you listen to a voice rendering of text messages sent from a supported e-mail system and/or Intuity Message Manager
- *Print Text* allows you to print messages sent from a supported e-mail system and/or Intuity Message Manager
- *Enhanced Addressing* allows you to send a message to up to 1500 recipients
- *Transfer Restrictions* allow you to control toll fraud by restricting transfers going through the voice messaging system
- *Internet Messaging* allows you to exchange messages (voice and text) with any e-mail address via the World Wide Web.
- *Avaya Voice Director* allows you to address messages via spoken name, in addition to using touchtones to enter extensions or names. It also supports transferring to AUDIX subscribers, including those in other locations, by speaking a name.

Intuity Call Accounting System

If you are using any of the INTUITY voice messaging products, the INTUITY Call Accounting System is probably the best call-accounting solution for you. The system works exclusively with INTUITY products, which reside on MAP/40 or MAP/100 computers. Offering many of same features as the Call Accounting System for Windows, the system also serves to help integrate your INTUITY applications.

Intuity Conversant

The INTUITY CONVERSANT Voice Information System is an interactive voice-response system that automates phone-call transactions from simple tasks, like routing to the right department, to complex tasks, such as registering college students or providing bank balances. It communicates with customers in natural-sounding, digitally recorded speech. And it performs — 24 hours a day and without the services of an operator.

The system can handle single or multiple voice-response applications simultaneously, and can serve up to 48 callers at once. It can operate by itself to dispense information or collect data, or it can work with a host computer to access a large database such as bank account records. With its speech-recognition capability, even rotary telephone users can have access to sophisticated phone-based services. Advanced telephone features provide intelligent call-transfer capabilities and allow you to use the system in your existing telephone environment.

Intuity Lodging

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

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

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

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

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

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

Intuity Lodging Call Accounting System

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

Leave Word Calling (LWC)

Allows internal system users to leave a short preprogammed message (usually “Call” with the calling user’s name, extension number, and the time of the call) for other internal users. When the message is stored on MultiVantage, 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.

Leave Word Calling (LWC) — QSIG/DCS

The Leave Word Calling feature is extended to enterprise networks with QSIG as the private network infrastructure, as well as those with DCS. For enterprise networks that are mixed or in transition from DCS to QSIG, interworking of the LWC feature between the protocols can be provided. LWC also works within a single non-networked switch.

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.

Message Demand Print

Allows you to print your undelivered messages without calling the Message Center.

Message Retrieval

With the Message-Waiting Lamp on their telephones, employees always know when they have messages. Messages can be retrieved in a variety of ways. These message retrieval options can be assigned to individual users.

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.

Mode Code Interface

MultiVantage supports an analog Mode Code interface for communications with Intuity AUDIX and other voice mail systems using the same interface. This interface employs DTMF tones, line signals, and feature access codes, and allows Intuity AUDIX to exchange data with MultiVantage without using a data link. Other adjunct vendors can engineer their products to use this interface.

EMBEDDED AUDIX

While many voice messaging systems require separate equipment and connections, the EMBEDDED AUDIX System easily installs directly into your MultiVantage cabinet to support advanced voice messaging capabilities without the need for an adjunct processor. Each EMBEDDED AUDIX system supports up to 2000 mailboxes and stores up to 100 hours of recorded messages.

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

Special voice-processing features include Voice Mail, Call Answering, Outcalling, Multi-Level Automated Attendant, and Bulletin Board. The following is a summary of EMBEDDED AUDIX capabilities:

- *Shared Extensions* provide personal mailboxes for each person sharing a phone
- *Multiple Personal Greetings* allows you to prepare a pool of up to nine personal greetings to save time and provide more personal customer service. Separate messages can indicate you are on the phone, away from the desk, on vacation, etc. You can assign different messages to internal, external, or after-hours calls

- *Priority Messaging* places important messages ahead of others. Internal and External callers can mark the message as priority.
- *Outcalling* automatically dials a prearranged phone number or pager when you have messages in your voice mailbox
- *Priority Outcalling* automatically dials a prearranged phone number or pager when you have *priority* messages in your voice mailbox
- *Broadcasting* allows you to send a single message to multiple recipients or to all users on the system
- *System Broadcast* allows you to send broadcast messages as regular voice messages, or as messages that recipients hear as they log in
- *AUDIX Directory,* allows you to look up the extension number of any other user by entering their name on the telephone keypad
- *Personal Directory* allows you to create a list of nicknames for quick access to telephone numbers
- *Call Answering for Nonresident Subscribers* provides voice mailboxes for users who do not have an extension number on the MultiVantage
- *Full Mailbox Answer Mode* informs callers whenever messages cannot be left because there is no room in a subscriber's mailbox
- *Name Record by Subscriber* lets you record your own name on the system
- *Automatic Message Scan* can play all new messages in part or in their entirety without requiring you to press additional buttons*, which is particularly useful when you are getting messages from your mobile phone*
- *Sending Restrictions by Community* enables you to limit the communities of callers who can communicate via AUDIX Voice Messaging
- *Group Lists* allows you to create mailing lists of up to 250 people to use for broadcasting messages
- *Message Forwarding* allows you to forward messages with or without attached comments
- *Name Addressing* allows you to address messages by name if you don't know the extension
- *Private Messaging* is a special coding feature that prevents recipients from forwarding messages
- *Leave Word Calling* allows you to press a button on your telephone in order to leave a standard *call me* message on any extension

- *Online Help* provides you with instant access to voiced instructions at any time when you are using the system.
- Multiple Language Support allows you to install up to nine languages on the system, from a superset of 30 available languages.
- *Enhanced Message Handling* gives you the flexibility for handling messages. Two of these features are Optional Advance/Rewind, which lets you advance through and rewind individual messages, and Undelete Messages, which lets you retrieve any messages that you may have accidentally deleted.

Octel Integration

MultiVantage systems integrate with the entire line of Octel Messaging systems including Octel 50 Message Server, Octel 100 Message Server, Octel 200/300 Message Server, and Octel 250/350 Message Server.

QSIG/DCS Voice Mail Interworking

QSIG/DCS Voice Mail Interworking is an enhancement to the current QSIG feature. It integrates DCS and QSIG Centralized Voicemail via the new DCS+/QSIG gateway. Switches labeled DCS+/QSIG integrate multi-vendor PBXs into a single voice messaging system. QSIG/DCS Voice Mail Interworking works on G3r, G3si, and G3csi. It provides network flexibility, DCS functionality without a dedicated T1.

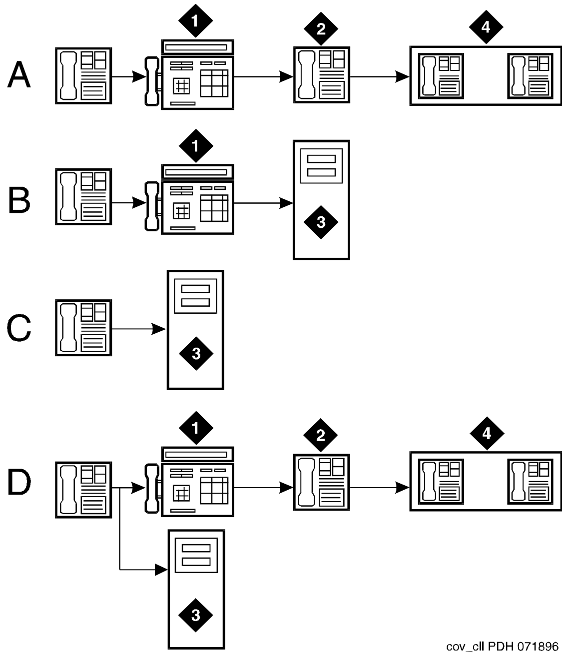
Voice Message Retrieval

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

Voice Messaging and Call Coverage

Often an AUDIX system is set up as the last point on a call-coverage path, as shown in Figure 4 on page 68. 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.



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

Figure 4. Typical MultiVantage Call Coverage Options

10 — Mobility

IP telephones or IP Softphones allow you to access the features of MultiVantage without having to be tied to one location. One of the major benefits of IP telephones is that you can move the telephones around on a LAN just by unplugging them and plugging them in somewhere else. One of the main benefits of IP Softphones is that you can load them on a laptop PC, and then use the PC's modem to connect them to the switch from almost anywhere.

For more information, see ''4600-Series IP Telephones'' on page 45 and ''Avaya IP Softphones'' on page 47.

Administration Without Hardware (AWOH)

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

Automatic Customer Telephone Rearrangement (ACTR)

Automatic Customer Telephone Rearrangement (ACTR) allows a phone to be unplugged from one location and moved to a new location without additional switch administration. The switch automatically associates the extension to the new port. ACTR works with the 2420 DCP telephone and the 6400 serialized telephones. The 6400 Serialized phone is stamped with the word Serialized on the faceplate for easy identification. The 6400 Serialized phone memory electronically stores its own part ID (comcode) and serial number. ACTR uses the stored information and associates the phone with new port when the phone is moved.

ACTR makes it easy to identify and move phones.

DEFINITY Wireless Business System (DWBS)

The DEFINITY Wireless Business System (DWBS) relies on the MultiVantage system to manage mobility. It uses Personal Wireless Telephony technology, which is a leading protocol in the United States. This standard, which has the primary advantage of permitting up to 12 simultaneous conversations per base station, defines the radio interface between the portable telephones and the base stations in the system.

The DWBS is fully integrated with the MultiVantage and offers users full access to the MultiVantage features. The system has the following maximum capacities:

- 1500 wireless telephones
- 300 base stations
- 7,000 to 40,000 calls per busy hour (depending on MultiVantage configuration)
- 12-million-square-foot (3.6-million-square-meter) coverage area

Cluster ID Administration

This feature 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

This feature 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. This enhancement affects R10r systems only.

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 MultiVantage as if the telephones were directly connected to the switch. The telephones are administered to be of the type XMOBILE 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 features as call-associated display, bridging, message waiting, call redirection, and so forth. X-Station Mobility is currently used for non cellular wireless offers (DECT and PHS) in EMEA and APAC regions, and the EC500 Extension to Cellular offer globally.

EC500 Extension to Cellular

EC500 is an integrated mobility solution that offers users the freedom to work anywhere, anytime, using any type of cellular or wireless phone. With EC500, calls to an office number are extended to a cell phone, allowing users to receive work-related calls wherever they are and whenever they need to. Additionally, the cell phone can be administered so that when a user calls into the office, the user’s name and office telephone number appear in the caller ID display of the phone being called. When the EC500 cell phone is administered to send office caller ID, the user also has the option of picking up an ongoing EC500 cell phone call on the office phone upon entering the office.

EC500 works over PRI as well as an IP trunk interface. The cell phone user receives the same features and capabilities for incoming calls as a caller ID-enabled analog telephone connected directly to the Avaya Communications Server. EC500 provides this capability regardless of the cell phone’s Cellular Service provider or the cellular standard in use.

EC500 offers the following capabilities:

- XMOBILE station administration enhancements to include the Dial Prefix, Cell Phone Number, Mapping Mode, XMOBILE Type and Configuration set fields
- User Control of EC500 through EC500 Activation and Deactivation Feature Access Codes
- Office caller ID for call originations from the cell phone
- EC500 enabled/disabled status in the status station screen
- Administration of XMOBILE Configuration Set options
- XMOBILE Duplicate Station support for bulk administration
- List XMOBILE mappings by cell phone number
- EC500 information in the Display Capacities form

Personal Station Access (PSA)

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 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.

Don’t Answer Reason Code (for PSA-disassociated stations)

PSA uses Administered Without Hardware, a feature that allows the MultiVantage switch administrator to assign a station without specifying a physical port — for example, use “X” as the port. If a station is disassociated, it means that it is not currently mapped to a particular physical endpoint such as a digital telephone. If a caller dials into an station extension that is currently disassociated, they are provided a message that indicates “Don't Answer” instead of “Busy”.

Name/Number Permanent Display

When a person uses PSA to associate their extension with a station, a display appears on the station indicating their name and extension number. This information is displayed until the user disassociates their extension from the station using the PSA-associate feature access code.

Terminal Translation Initialization (TTI)

MultiVantage provides Terminal Translation Initialization (TTI), a feature that works with Administration Without Hardware. 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 terminal that is connected to a wired (but untranslated) jack.

TransTalk 9000 Digital Wireless System

The TransTalk 9000 is a single-zone or dual-zone, in-building wireless system that provides a mobility solution on MultiVantage-based systems. It delivers the benefits and accessibility of a wireless phone with all the power and functionality of a wired desk telephone.

11 — Networking and Connectivity

Private Networking and Connectivity

Communication Device Support

Circuit Switched

Analog 6200-Series

See “6200-Series Analog Telephones” on page 45.

Digital Telephones

2420 DCP Telephones

See “2420 DCP Telephones” on page 45.

6400-Series Telephones

See “6400-Series DCP Telephones” on page 46.

6400 Tip/Ring Interface Module

See “6400 Tip/Ring Interface Module” on page 46.

8400-Series Telephones

See “8400-Series Telephones” on page 46.

Internet Protocol (IP)

4600-Series IP Telephones

See “4600-Series IP Telephones” on page 45.

Avaya IP Agent

See “Avaya IP Agent” on page 47.

Avaya IP Softphones

See “Avaya IP Softphones” on page 47.

IP Endpoint — Road-warrior mode

Enables use of the full MultiVantage 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 MultiVantage over an IP network. The single network connection between the PC and MultiVantage carries two channels, one for the signaling path and one for the voice path. On MultiVantage, the road-warrior application requires the C-LAN circuit pack for signaling and the IP media processor for voice processing.

IP Endpoint — Telecommuter mode

Enables telecommuters to use the full MultiVantage feature set from home. It consists of a PC and a telephone with separate connections to MultiVantage. 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 C-LAN circuit pack for signaling. The telecommuter application does not use the IP media processor.

Wireless

MultiVantage supports wireless devices. For specific information, see ‘‘Mobility’’ on page 69.

Port Network and Gateway Connectivity

Asynchronous Transfer Mode (ATM)

The Asynchronous Transfer Mode (ATM) switch is a replacement option for the CSS or the direct-connect switch. Several Avaya ATM switch types can provide MultiVantage port network connectivity. Non-Avaya ATM switches that comply with the ATM standards set by the European Union can also provide MultiVantage port network connectivity.

Avaya ATM WAN Survivable Processor Manager

See “Avaya ATM WAN Survivable Processor Manager” on page 143.

Port Network Connectivity (ATM-PNC)

ATM Port Network Connectivity (ATM-PNC) provides an alternative to the Center Stage Switch (CSS) configurations for connecting the Processor Port Network (PPN) to one or more Expansion Port Networks (EPNs). ATM-PNC replaces the CSS in a DEFINITY R8r and later network with an ATM switch or network. ATM-PNC is available with all three MultiVantage reliability options — standard, high, and critical. In addition, it offers ATM-PNC duplication.

ATM-PNC integrates delivery of voice, video, and data via ATM over a converged large-bandwidth network, providing reduced infrastructure cost and improved network manageability. ATM-PNC uses standards-based open interfaces that can be provisioned with either new or existing MultiVantage systems.

Port Network Connectivity (ATM-PNC) over WAN

ATM-PNC over a public WAN represents an environment where the customer uses a service provider's ATM Network between privately-owned ATM switches. The customer does not control the ATM switches in the network, including traffic policing policies and product quality.

Using a public WAN, Permanent Virtual Paths (PVP) may be set up between customer-owned ATM switches similar to the dedicated circuits in a Private WAN. However, ATM cell processing occurs in a Public WAN so the customer is dependent on ATM switches owned and managed by the Service Provider.

Switched Virtual Circuits (SVC) use the ATM protocol to transmit voice-like applications over ATM networks. The advantage of the SVC solution is that MultiVantage can dynamically signal the ATM network to provide more bandwidth as needed to handle peaks in the call traffic. If the ATM Network cannot handle the additional traffic, calls will be denied.

WAN Spare Processor (WSP)

An ATM WAN spare processor (WSP) provides a disaster recovery option for MultiVantage G3r expansion port networks deployed over an ATM WAN. An ATM WSP acts as a PPN in the event of a catastrophic failure in the network. The ATM WSP continually monitors a path to the PPN to determine if it is on-line and reachable. The WSP functions as a PPN if the main PPN is not functional or is not communicating to one or more of the other EPNs. From one to 15 ATM WSPs can be placed in a MultiVantage ATM port network configuration to provide a backup arrangement of PPNs, thus maintaining the availability of the MultiVantage features and functions.

NOTE:
ATM WSPs cannot be used with a conventional CSS.

Circuit Switched

Center Stage Switching

MultiVantage supports CSS as a method to interface between the PPN and EPNs using circuit switched technology to carry the voice traffic.

Internet Protocol (IP)

H.248 Media Gateway Control

MultiVantage uses standards based H.248 to perform call control to Avaya Media Gateways such as the G700. H.248 defines a framework of call control signaling between the intelligent Media Servers and multiple “unintelligent” Media Gateways.

IP Port Network Connectivity

MultiVantage allows Control Channel Message Set (CCMS) messages to be packetized over IP LAN and WAN connections to control multiple Port Networks.

Trunk Connectivity

Asynchronous Transfer Mode (ATM)

See “Asynchronous Transfer Mode (ATM)” on page 77.

Circuit Emulation Service (ATM-CES)

ATM-Circuit-Emulation Service (ATM-CES) lets the MultiVantage emulate ISDN-PRI trunks on an ATM facility. These virtual trunks can serve as integrated access, tandem, or tie trunks. ATM-CES trunk emulation maximizes port network capacities by consolidating trunking. For example, the CES interface can define up to eight virtual circuits for tie-line connectivity, consolidating onto one circuit card network connectivity that usually requires multiple circuit packs. ATM-CES is available on all platforms (r, si, and csi).

CMS Measurement of ATM

See “CMS Measurement of ATM” on page 22.

Circuit Switched

DS1 Trunk Service

See “DS1 Trunk Service” on page 89.

Echo Cancellation — with UDS1 Circuit Pack

See “Echo Cancellation — with UDS1 Circuit Pack” on page 89.

E1

See “E1” on page 90.

T1

See “T1” on page 90.

Internet Protocol (IP)

IP Trunks

IP trunk groups may be defined as a virtual private network’s tie lines between MultiVantage systems or ITS-E servers. Each MultiVantage IP Trunk circuit pack provides a basic 12-port package that can be expanded up to a total of 30 ports. The number of ports that are defined will correspond to the total number of simultaneous calls transmitted over the IP Trunk Interface.

The benefits of IP Trunk include a reduction in long distance voice and fax expenses, facilitating global communications, providing a full function network with data and voice convergence and optimizing networks by using the available network resources.

IP trunking is a good choice for basic, corporate voice and fax communications, where cost is a major concern. IP-trunk calls travel over a company’s intranet rather than the Public Telephone Network. So, for the most common types of internal, corporate communications, IP trunks offer considerable savings.

IP trunking is usually not a good choice for applications where calls have to be routed to multiple destinations (as in most conferencing applications) or to a voice messaging system. IP-trunk calls are compressed to save network bandwidth. Repeated compression and decompression results in a loss of data at each stage and degrades the final quality of the signal. The maximum number of compression cycles acceptable on a call is three, and three compression cycles can compromise voice quality. Normal corporate voice or fax calls typically go through fewer than three compression cycles. However, multipoint conference calls and most voice messaging systems add too many compression cycles for acceptable quality.

H.323 Trunk

A TN802B in MedPro mode or a TN2302AP IP Interface enables H.323 trunk service using IP connectivity between two MultiVantage systems. The H.323 trunk groups can be configured as system-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.

Trunk Types and Signaling

Auxiliary Trunks

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

Advanced Private Line Termination (APLT)

Provides access to and termination from CO (Central Office)-based private networks; namely, Common Control Switching Arrangements (CCSA) and Enhanced Private Switched Communications Service (EPSCS). APLT trunks are physically the same as those used for analog tie trunks, where the trunk signaling is compatible with EPSCS and CCSA network switches. The outgoing APLT trunk repeats any number of digits to the private network as dialed. APLT trunks can tandem through the PBX from EPSCS network only; CCSA networks require an Attendant to complete the call.

Central Office (CO)

Central Office (CO) trunks connect MultiVantage to the local Central Office for incoming and outgoing calls.

Digital Multiplexed Interface

Supports two signaling techniques: bit-oriented signaling and message-oriented signaling for direct connection to host computers.

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

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

Bit-Oriented Signalling

Digital Multiplexed Interface Bit-Oriented Signalling carries framing and alarm data and signalling information for connections to host computers and other vendor equipment.

Message-Oriented Signalling

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

Direct Inward Dialing (DID)

Direct Inward Dialing (DID) trunks connect MultiVantage to the local Central Office for incoming calls dialed directly to stations without Attendant assistance.

Direct Inward/Outward Dialing (DIOD)

Traditionally, Central Office (CO) trunks and Direct Inward Dialing (DID) trunks interface a PBX with a Central Office. A CO trunk services outgoing calls and accepts incoming calls that are terminated at the Attendant. A DID trunk is used for calls that need to be terminated without an Attendant interaction.

E&M Signaling — Continuous and Pulsed

See “E&M Signaling — Continuous and Pulsed” on page 56.

E911 CAMA Trunk Group

This form administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller’s Emergency Service Identification (CESID) information to the local community’s Enhanced 911 system through the local Central Office.

Foreign Exchange (FX)

Foreign Exchange trunks connect MultiVantage to a Central Office other than the local one.

ISDN Trunks

Gives you access to a variety of public and private network services and facilities. The ISDN standard consists of layers 1, 2, and 3 of the Open System Interconnect (OSI) model. MultiVantage can be connected to an ISDN using standard frame formats: Basic Rate Interface (BRI) and the Primary Rate Interface (PRI).

An ISDN provides end-to-end digital connectivity and uses a high-speed interface which provides service-independent access to switched services. Through internationally accepted standard interfaces, an ISDN provides circuit or packet-switched connectivity within a network and can link to other ISDN supported interfaces to provide national and international digital connectivity.

Automatic Termination Endpoint Identifier (TEI)

The user side will support automatic TEI assignment by the network. Both fixed and automatic TEI assignment will be supported on the network side.

Call-by-Call Service Selection

Enables a single ISDN-PRI trunk group to carry calls to a variety of services, rather than requiring each trunk group to be dedicated to a specific service. It allows you to set up various voice and data services and features for a particular call.

ETSI Functionality

The full set of ETSI public-network and private-network ISDN features is officially supported. This includes Look-Ahead Interflow, Look-Ahead Routing, and Usage Allocation. It also includes all QSIG supplementary services:

- Name Identification
- Call Diversion (including rerouting)
- Call Transfer
- Path Replacement

It does not include:

- DCS
- Non-Facility Associated Signaling
- D-Channel Backup
- Wideband Signaling

Facility and Non-Facility Associated Signaling

Allows an ISDN-PRI DS1/E1 interface D-channel to carry signaling information for B-channels (voice or data). D-Channel Backup can also be administered to increase system reliability.

Feature Plus

Enables those without DID to direct dial users on a remote PBX via the public network. ISDN Feature Plus eliminates the need for Attendant intervention for those without DID capabilities.

ISDN-Basic Rate Interface (ISDN-BRI)

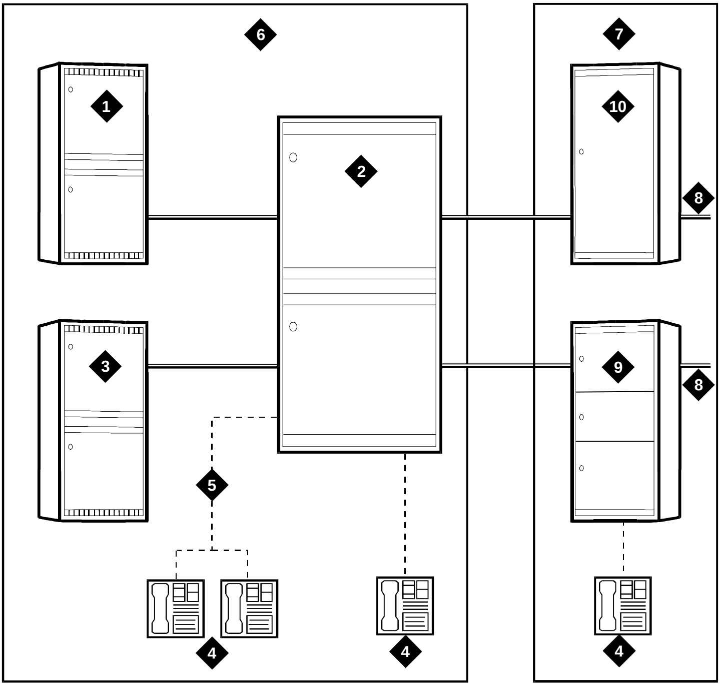
Enables connection of the system to equipment or endpoints that support an Integrated Services Digital Network (ISDN) by using a standard format called the Basic Rate Interface (BRI). This feature is a 192-Kbps interface that carries two 64-Kbps B-channels and one 16-Kbps D-channel.

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

The ISDN-BRI Trunk circuit pack allows MultiVantage to support the T interface and the S/T interface as defined by ISDN standards (ITU-T recommendation I.411). The circuit pack provides eight ports to the network and supports two B channels and one D channel. The ISDN-BRI Trunk provides the following advantages:

- Provides an inexpensive way to connect to ISDN services provided by the network provider
- Meets almost all ETSI Country protocol requirements
- Supports essential (not supplementary) ISDN services

BRI trunks support public-network access outside the U.S. on point-to-midpoint connections, with the restriction that MultiVantage must not be configured in a passive bus arrangement with other BRI endpoints. ISDN-BRI trunks can be used as inter-PBX tie lines using the QSIG peer protocol.



1	MultiVantage	6	Private ISDN (can be carried over ATM-CES)
2	MultiVantage	7	Public ISDN (can be carried over ATM-CES)
3	MultiVantage	8	Public and Private Networks
4	Basic Rate Interface Telephone	9	Central Office Switch
5	Passive Bus	10	Tandem Switch

Figure 5. MultiVantage and ISDN

Multiple Subscriber Number (MSN) - Limited

The ISDN standard MSN feature lets customers assign multiple extension to a single BRI endpoint. The MSN feature works with BRI endpoints that allow the Channel ID IE to be encoded as “preferred.”

NT Interface on TN556C

MultiVantage supports the NT (network) side of the T interface using the TN556C circuit pack. This gives the switch full tie trunk capability using BRI trunks. MultiVantage supports leased BRI connections through the public network, with a TN2185 on each end of the leased connection. MultiVantage will not, however, allow customers to administer both endpoints and trunks on the same TN556C circuit pack.

Presentation Restriction

Restricts the display of calling/connected numbers over ISDN trunks. ISDN trunk groups can be administered to control the display of calling/connected numbers. Each trunk group can be administered to display “Presentation restricted,” “Number no available due to networking,” or an administered text string instead of the calling/connected number.

Wideband Switching

Provides the ability to dedicate 2 or more ISDN B-channels or DSO endpoints for applications that require large bandwidth. Certain applications, such as video conferencing and high-speed data transmission, require extra bandwidth and it becomes necessary to put several ISDN-PRI narrowband channels into one wideband channel to accommodate the needs of these applications. This feature supports both European and North American standards.

Multi-Frequency Packet (MFP) Signaling — Russia

See “Multi-Frequency Packet (MFP) Signaling — Russia” on page 57.

National Private Networking Support — Japan

See "National Private Networking Support — Japan" on page 57.

Personal Central Office Line (PCOL)

Provides a dedicated trunk circuit between multi-appearance telephones and a CO or other switch via the network.

Release Link Trunks

Release Link trunks are used between switch locations to provide Centralized Attendant Service or Automatic Call Distribution group availability.

Remote Access Trunks

Tie Trunks

Tie trunks carry communications between MultiVantage and other switches in a private network. Several types of trunks can be used, depending on the type of private network you establish.

Wide Area Telecommunications Service (WATS)

Wide Area Telecommunications Service (WATS) trunks allow you to place long-distance outgoing voice-grade calls to telephones in defined service areas. The calls are priced according to distance in the service area, length of the call, time of day, and the day of the week.

Public Networking and Connectivity

Caller ID (ICLID) on Digital Trunks

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 MultiVantage digital telephones (DCP and BRI) equipped with a 40-character or a 32-character alphanumeric display.

Caller ID (ICLID) on Analog Trunks

Allows the system to accept calling name information from a local exchange carrier (LEC) network that supports the Bellcore calling name specification. The system can send calling name information in the format if Bellcore Calling Name ID is administered.

DS1 Trunk Service

Bit-oriented signaling that multiplexes 24 channels into a single 1.544-Mbps stream. DS1 can be used for voice or voice-grade data and for data-transmission protocols. E1 trunk service is bit-oriented signaling that multiplexes 32 channels into a single 2.048-Mbps stream. Both DS1 and E1 provide a digital interface for trunk groups. Digital Service 1 (DS1) trunks can be used to provide T1 or ISDN Primary Rate Interface (PRI) service.

Echo Cancellation — with UDS1 Circuit Pack

The Universal DS-1 (UDS1) circuit pack (TN464GP/TN2464BP) available for all MultiVantage platforms has echo cancellation circuitry. The echo cancellation capability of the circuit pack is intended only for channels supporting voice communication. It is not desirable to provide echo cancellation over channels supporting data communication.

The TN464GP/TN2464BP is intended for MultiVantage customers who are likely to encounter echo over circuits connected to the public network. The occurrence of echo is likely if the MultiVantage is configured for complex services such as ATM or IP. In addition, echo is likely to occur if the MultiVantage interfaces to local service providers who do not routinely install echo cancellation equipment in all their circuits.

E1

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

T1

When planning your networking requirements, one of the options you should consider is multiplexing over Digital Services 1 (DS1) facilities.

Flexible Billing

See “Flexible Billing” on page 16.

Local Exchange Trunks

Local exchange trunks connect MultiVantage to a Central Office. The following are some of the types available:

800-Service Trunks

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

Central Office (CO) Trunks

See “Central Office (CO)” on page 81.

Digital Service 1 (DS1) Trunks

See “DS1 Trunk Service” on page 89.

Direct Inward Dialing (DID) Trunks

See “Direct Inward Dialing (DID)” on page 82.

Direct Inward/Outward Dialing (DIOD) Trunks

See “Direct Inward/Outward Dialing (DIOD)” on page 82.

Foreign Exchange (FX) Trunks

See “Foreign Exchange (FX)” on page 83.

Wide Area Telecommunications Service (WATS)

See “Wide Area Telecommunications Service (WATS)” on page 88.

Intelligent Networking

Avaya VoIP Monitoring Manager

See “Avaya VoIP Monitoring Manager” on page 144.

Distributed Communications System (DCS)

Distributed Communications System (DCS) allows you to configure 2 or more MultiVantage switches as if they were a single, large MultiVantage system. DCS provides attendant and voice-terminal features between these switch locations. DCS simplifies dialing procedures and allows transparent use of some of the MultiVantage features. (Feature transparency means that features are available to all users on DCS regardless of the switch location.)

See also, ‘‘Centralized Attendant Service (CAS)’’ on page 10 and ‘‘Inter-PBX Attendant Calls’’ on page 7.

Attendant with DCS

Direct Trunk Group Selection

See “Direct Trunk Group Selection” on page 6.

Display

See “Display” on page 6.

DCS Automatic Circuit Assurance

Allows a user or Attendant at one node to activate or deactivate Automatic Circuit Assurance referral calls for the entire DCS network. This transparency allows the referral calls to originate at a node other than the node that detects the problem.

DCS Over ISDN-PRI D-channel (DCS+)

Enhances DCS by allowing access to the public network for DCS connections between DCS switch nodes. With this feature (also known as DCS Plus or DCS+), DCS features are no longer restricted to private facilities. The ISDN-PRI B-channel is used for voice communications, and the ISDN-PRI D-channel is used to transport DCS control information.

DCS Protocol — Italy

See “Distributed Communications Systems (DCS) Protocol — Italy” on page 56.

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. This feature allows for rerouting the call after a transfer or rerouting during a call. DCS With Reroute is similar to the rerouting capabilities used with QSIG.

QSIG/DCS Voice Mail Interworking

See “QSIG/DCS Voice Mail Interworking” on page 66.

Electronic Tandem Network (ETN)

In an Electronic Tandem Network (ETN) — also known as Private Network Access (PNA) — MultiVantage provides a variety of features on a network-wide basis. It allows calls to other systems in a private network. These calls do not use the public network. Instead, they are routed over your dedicated facilities.

Automatic Alternate Conditional Routing

You can control the routing of particular calls using conditional routing. For example, you can limit the number of communications satellite hops (communications satellite links used as trunks) in any end-to-end private network routing pattern. Limiting the number of satellite hops may be desirable for controlling transmission quality or call delay in both voice and data calls.

Trunk Signaling and Error Recovery

The reliability of Electronic Tandem Network calls is improved by allowing a trunk call to be retried on another circuit when signaling failures occur.

- **tandem switch**: A switch within an electronic tandem network (ETN) that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.
- **tandem through**: The switched connection of an incoming trunk to an outgoing trunk without human intervention.
- **tandem tie-trunk network (TTTN)**: A private network that interconnects several customer switching systems.

See also, “Port Network Connectivity (ATM-PNC)” on page 77.

Extension Number Portability

When employees move within the network, they can retain their extension numbers. The ability to keep extension numbers, and even Electronic Tandem Network and Direct Inward Dialed numbers, when moving to other locations within the company eliminates missed calls and saves valuable time.

Internet Protocol (IP)

The capabilities and applications of MultiVantage are extended using IP. MultiVantage IP supports audio/voice over a LAN or WAN, and it ensures that remote workers have access to communication system features from their PCs. MultiVantage also provides standards based control between Media Server and Media Gateways allowing communications infrastructure to be distributed to the edge of the network. The MultiVantage IP engine offers features that enables users to increase the quality of voice communications. The Quality of Service (QoS) feature enables users to administer and download the Differentiated Services Type-of-Service value to optimize voice quality. The Quality of Service feature reduces latency by implementing buffers in the audio-processing board, and assists some routers in prioritizing audio traffic.

MultiVantage IP also includes 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.

IP Solutions supports trunks, IP Communications Devices, IP Port Networks and IP control for Media Gateways. IP Solutions is implemented using various IP-media processor circuit packs inside the MultiVantage servers or the Avaya Media Gateways. The IP media processors provides H.323 trunk connections and .323 voice processing for IP telephones. The features that use the IP media processor also require the C-LAN circuit pack or native processor Ethernet connectivity.

The IP LAN can also connect via VPN and WAN facilities to extend the customer IP network across geographically disparate locations. With MultiVantage ISDN, Distributed Communication Services (DCS+), or QSIG Services, MultiVantage can extend feature transparency, centralized voice mail, Centralized Attendant Service, call center applications, and enhanced call routing across IP Trunks.

NOTE:

To maximize voice quality using IP, 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. The IP network must be a switched Ethernet infrastructure and have the appropriate engineering to accommodate bandwidth, latency and packet loss requirements to effectively provide for real-time voice over IP traffic.

Alternate Gatekeeper and Registration Addresses

When an IP endpoint (including softphones, IP phones, and Avaya R300) registers with the switch, the switch sends back an IP registration address. The switch sends a different IP address for each registration according to a cyclic algorithm.

If registration with the original C-LAN circuit pack IP address is successful, the switch sends back the IP addresses of all the C-LAN circuit packs in one network region, not including interconnected regions. These C-LAN addresses are called gatekeeper addresses. These addresses can also be used if the call signaling on the original C-LAN circuit pack fails.

NOTE:

On switches using the LAN region based on IP Address feature, it's likely that the network region number assigned to an IP phone would be different from the network region number of the TN799 that the phone is registering through. That difference would mean the list of TN799 addresses in the same network region as the IP phone would be empty. The alternate gatekeeper feature would send a blank list to the IP phone. To prevent that from happening, an IP terminal registers with MultiVantage, MultiVantage sends to the endpoint the IP addresses of the CLANs in the same region as the terminal, followed by network regions interconnected with the network region of the terminal.

If the network connection to one C-LAN circuit pack fails, the IP endpoint re-registers with a different C-LAN. Alternate gatekeeper and registration addresses, and C-LAN circuit pack load sharing, spread IP endpoint registration across more than one C-LAN circuit pack, increasing performance and reliability.

Classless Interdomain Routing (CIDR)

CIDR is a redefinition of the subnet mask, allowing for the aggregation of contiguous classful networks under a single network definition. This allows for more efficient routing table management when administering IP address on MultiVantage.

Multiple Network Regions per C-LAN

See “Multiple Network Regions per C-LAN” on page 119.

Multiple Location Support for Network Regions

Multiple Location Support for Network Regions allows remote Avaya Media Gateways connected to a central Avaya Media Servers to retain local user time, local ARS Public Analysis Tables for local trunking, automatic Daylight Savings Time, and local touch tone receivers for IP communications devices such as Avaya IP Telephones. MultiVantage allows administrators to map locations to IP Network Regions.

Network Regions

Network Regions provide the administrative foundation on which MultiVantage features are allocated to IP endpoints. A Network Region is a collection of IP endpoints and switch IP interfaces interconnected by an IP network. Endpoints that share network regions typically represent users with common interests. For example, a customer might have two separate small campuses in a large metropolitan area, interconnected by a WAN, and both served by the same server running MultiVantage. MultiVantage allows the customer to define a network region for each campus, and associate each of their C-LAN and IP Media Processor circuit packs with these regions.

Quality of Service (QoS)

MultiVantage provides the best possible end-to-end audio experience, when all or part of the audio path is carried over packet facilities, by employing a variety of Quality of Service (QoS) features. “Best,” in this context, is defined by the customer as represented by the system administrator, and represents a trade-off between audio reproduction quality, audio path delay (latency), audio loss, and network resource consumption.

802.1p/Q

IEEE standard 802.1Q and 802.1p provide the means to specify both a virtual LAN (VLAN) and a frame priority at layer 2 for use by LAN hubs, or bridges, which can do routing based on MAC addresses. 802.1p/Q provides for 8 levels of priority (3 bits) and a large number (12 bits) of VLAN identifiers. The VLAN identifier at layer 2 permits segregation of traffic to reduce traffic on individual links. Because 802.1p operates at the MAC layer, its presence may vary from LAN segment to LAN segment within a single network region. Flexibility requires that 802.1p/Q options be administered individually for each network interface.

Codecs

Codecs provide the means by which audio is compressed and are typically used in VoIP. Codecs supported by MultiVantage include G.711, G.723, and G.729.

Differentiated Services (DiffServ)

With the DiffServ option, the system administrator can administer (by region) and download, to the TN2302AP, the DiffServ Type-of-Service (TOS) value. This allows data networking equipment to prioritize the audio stream at the IP level to promote voice quality. DiffServ makes use of the Type-of-Service (TOS) octet in the existing IP Version 4 header. As such, it may be set by information senders and used by IP (layer 3) routers within the network.

Dynamic Jitter Buffers

Propagation delay and jitter is caused when a human’s voice is sampled, encoded, and packetized for transport over the IP network, but is received and decoded at different rates. Jitter buffers are used to buffer the audio output to smooth the conversions. MultiVantage provides dynamic jitter buffers to balance both delay of conversation and rapid bursts that may occur.

Integration with Cajun Rules

Cajun Rules provides a central repository for QoS parameters and allows comprehensive QoS management across routers, switches, and endpoints. QoS parameters and policies are assigned according to network regions on a Network Region and are distributed through Enterprise Directory Gateway to the MultiVantage and to routers and switching devices.

QoS for Call Control

MultiVantage allows QoS for the signaling packets coming from gatekeepers such as the C-LAN by employing the same standards based DiffServ and 802.1p/Q schemes as with audio channels. This QoS services further improve the users VoIP audio experience.

QoS for VoIP

MultiVantage implements QoS through the selection of audio codec such as G.711, G.723 and G.729, and by requesting network prioritization through the Layer 3 Differentiated Services (DiffServ) scheme, as well as the Layer 2 IEEE 802.1p/Q prioritization. Diffserv and 802.1p/Q are supported on voice packets coming to/from the gateway, all the way down to the endpoints such as IP Telephones. Dynamic Jitter Buffers are also used.

QoS to Endpoints

Users can set operating parameters to optimize the audio performance, or Quality of Service (QoS), on calls made over your IP network. These parameters include the audio codec, network priority through DiffServ capability, and the IEEE 802.1p/Q MAC-layer prioritization and segregation.

Default QoS parameters are downloaded to the IP Telephone R1.5 and the IP Softphone R3 when the values are not provided by the endpoint installer or the user. Certain options can be set locally by the endpoints or via the gatekeeper. The endpoints receive the parameters when the endpoints register, and once they are registered, whenever the administered values of the QoS parameters are modified.

RSVP

Resource Reservation Protocol (RSVP) is a protocol that allows an endpoint to negotiate with a RSVP-capable network to allocate protected resources for traffic that the endpoint will generate. RSVP provides protection for traffic flows, which is also essential for providing the quality of service required by VoIP. One of the primary purposes of RSVP is to mediate between an end-point and the set of routers on the path to another end-point to reserve resources to provide a guaranteed QoS for a session. RSVP is used for IP endpoints and is supported for configurations including MultiVantage Server and Avaya Media Gateways.

Shuffling and Hairpinning

Hairpinning and Shuffling can improve traffic handling performance and improve voice quality by more efficiently using both MultiVantage switching fabric by allocating, when possible, available IP network resources.

Hairpinning means rerouting the audio channel connecting two IP endpoints so that the bearer (audio) packets are routed through the TN2302AP IP Media Processor board in IP format, without having to go through the IP to TDM conversion or traverse the TDM bus.

Shuffling means rerouting the audio channel connecting two IP endpoints. After shuffling, the audio which previously was carried in a mixed connection of IP signaling and TDM bus signaling, now goes directly through the LAN or WAN between the two IP endpoints. Shuffling also can mean reversing this process if an endpoint requests a resource to support a feature such as conferencing that requires the TDM bus.

Variable Length Ping

Provides an enhancement to the ping command included in R7.1. This enhancement specifies a longer packet to be sent by ping and shows if a router or host has a problem fragmenting or integrating transferred packets.

Variable Length Subnet Mask (VLSM)

VSLM is a redefinition of the subnet mask, allowing for a more efficient allocation of IP addresses within a traditional classful block when administering IP address on MultiVantage.

QSIG

Basic

QSIG provides compliance to the International Organization for Standardization (ISO) ISDN-PRI private-networking specifications. QSIG is defined by ISO as the worldwide standard for private networks. QSIG features are supported on BRI trunks.

QSIG is the generic name for a family of signaling protocols. The Q-reference point or interface is the logical point where signaling is passed between 2 peer entities in a private network. QSIG signaling can provide feature transparency in a single-vendor or multi-vendor environment.

QSIG provides call-related Supplementary Services. These are services that go beyond voice or data connectivity and number transport and display. Examples of Supplementary Services include Name Identification, Call Forwarding (Diversion), and Call Transfer.

Call Completion

Call Completion utilizes the QSIG Platform enhancement Call Independent Signaling Connections and is functionally equivalent to the Distributed Communications System (DCS) feature: AutoCallback. The Call Completion feature includes a connection release method. The connection release method clears the temporary signaling connection (TSC) after each phase of call-independent signaling and establishes a new TSC for each subsequent phase.

Call Forwarding (Diversion)

QSIG Call Forwarding (Diversion) is based on the MultiVantage Call Forwarding feature. It extends the feature transparency aspects of Call Forwarding over a QSIG trunk:

- If QSIG Call Forwarding is activated, all calls are diverted immediately.
- If QSIG Call Forwarding with Busy/Don’t Answer is activated and a station is busy, a call is diverted immediately.
- If QSIG Call Forwarding with Busy/Don’t Answer is activated and a station is idle but the call is not answered, a call is diverted after a specified number of rings.

These features are activated either by dialing a Feature Access Code (FAC) or by pressing a button. See Call Forwarding for detailed descriptions of how to use these features.

Call Independent Signaling Connections (CISC)

Call Independent Signaling Connections (CISC) are used to pass QSIG Supplementary Service information that is independent of an active call between two QSIG compliant nodes. Implementation is based on the ISO standard for CISC. It is possible to determine the status of a QSIG TSC by using the “status Signaling group” command on the SAT.

Call Offer

This feature, on request from the calling-user (or on that user’s behalf), enables a call to:

- Be offered to a busy called-user
- Wait for a busy called-user to accept the call when the necessary resources have become available

Call Transfer

QSIG Call Transfer differs from the standard MultiVantage Transfer feature in that additional call information is available for the connected parties after the transfer completes. However, the information is only sent for QSIG trunks. If one call is local to the transferring switch, that user receives the name of the party at the far end.

Called Name ID

The QSIG Called Name feature presents the called party's name on the calling party's display while the call is ringing. It then reverts to "connected name" when answered.

Centralized Attendant Service (CAS)

Provides you with the capability to have all your Attendants in one location, serving users in multiple locations. QSIG CAS does not utilize separate Release Link Trunks (RLT). This feature will not restrict calls from going out over non-QSIG trunks; however, the full functionality of the QSIG CAS will not be available.

Attendant Display of Class of Restriction (COR)

While on a call, the Attendant can press a "COR display" button to see the Class of Restriction of the user. The Attendant will not block the transfer of the restricted line to the user. This feature is used for informational purposes only.

Attendant Return Call

If a call that is transferred by the Attendant goes unanswered for a specified period of time, the call is returned to the Attendant. Preferably the call will transfer back to the Attendant who transferred the call.

Priority Queue

QSIG MSI will pass more information to the main PBX. This information enables calls coming in from a QSIG CAS branch to be placed in the appropriate place in the queue, as if the call originated on the main PBX.

RLT Emulation via a PRI

ISDN QSIG trunks will route calls from the branch PBX to the main PBX. You no longer have to specify a dedicated RLT network. The QSIG Path replacement takes care of the trunk optimization. You have the flexibility to route calls to the main PBX.

MultiVantage/Octel QSIG Integration

MultiVantage enables integration of Octel Messaging servers via QSIG.

Leave Word Calling (LWC)

See “Leave Word Calling (LWC)” on page 63.

Manufacturer-Specific Information (MSI)

QSIG handles non-standardized information that is specific to a particular PBX or network. This information is known as Manufacturer Specific Information (MSI). A manufacturer can define manufacturer-specific supplementary services operations after it has:

- Applied to a sponsoring and issuing organization (ECMA or European Computer Manufacturers Association in this case)
- Been assigned an organization identifier. This organization identifier is used as the root of the manufacturer-specific service-operation value.

All MSI operation values should be unique to that manufacturer.

Manufacturer-specific supplementary services can be created using specific operations encoded with the manufacturer’s identifier. MultiVantage supports non-QSIG applications that transport information across QSIG networks in MSI. Applications now have the same functionality over QSIG networks that they have over non-QSIG networks. Applications that use MSI include Centralized Attendant Service, Transfer to Audix, Best Service Routing, and QSIG VALU.

Message Waiting Indication (MWI)

The system indicates that a guest’s phone has received one or more messages in their voice mailbox. An automatic message waiting lamp light at the called party’s telephone.

Name and Number Identification

Allows a switch to send and receive the calling number, calling name, connected number, and connected name. Additional parameters that control the display of the connected name and number are administered on the Feature-Related System-Parameters form. QSIG Name and Number Identification displays up to 15 characters for the calling and connected name and up to 15 digits for the calling and connected number across ISDN-PRI interfaces.

Path Replacement with Path Retention

With this feature, a call’s connections between switches in a private network can be replaced with new connections while the call is active. This feature is invoked when a call is transferred and improvements may be made in costs. For example, after a call is transferred, the two parties on the transferred call can be connected directly and the unnecessary trunks are dropped off the call. The routing administered at the endpoints may allow for a more cost-effective connection.

Earlier versions of DEFINITY could not route a call over the original trunk when path replacement was turned on. Path Replacement features Path Retention, which allows MultiVantage to use the original trunk group path when the routing analysis performed by the switch shows the original trunk group to be the best route.

QSIG/DCS Voice Mail Interworking

QSIG/DCS Voice Mail Interworking is an enhancement to the current QSIG feature. It integrates DCS and QSIG Centralized Voicemail via the new DCS+/QSIG gateway. Switches labeled DCS+/QSIG integrate multi-vendor PBXs into a single voice messaging system. QSIG/DCS Voice Mail Interworking works on G3r, G3si, and G3csi. It provides network flexibility, DCS functionality without a dedicated T1.

Reroute After Diversion to Voice Mail

Supports path optimization for calls that are diverted to a QSIG voice mail hunt group. That is, the switch moves the call to the shortest route between the caller and the voice mail system. For example, if user A on switch A calls user B on switch B and the call goes to a voice mail system attached to switch C, then the call is using up two trunks: A-B and B-C. If there is a trunk that directly connects switches A and C, this feature will drop the A-B and B-C connection and set up a new call from switch A to switch C, thus saving one trunk. The reroute happens automatically; the user never knows that the extra trunk was dropped.

Stand-alone Path Replacement

Path Replacement is the process of routing an established call over a new, more efficient path, after which the old call is torn down leaving those resources free. Path Replacement offers potential savings by routing calls more efficiently, saving resources and trunk usage.

Path replacement can exist as a stand-alone feature, or occur in the following additional cases:

- Call Forwarding by Forward Switching supplementary service, including the case where Call Diversion by Rerouting fails, and Call Forwarding is accomplished via forward switching
- Gateway scenarios where MultiVantage, serving as an incoming or outgoing gateway, invokes PR to optimize the path between the gateways
- Calls in queue/vector processing even though no true user is on the call yet
- QSIG Lookahead Interflow call, Best Service Route call, or adjunct route

Supplementary Services and Rerouting

The QSIG standard defines Supplementary Services as those service beyond voice or data connectivity and number transport and display. Examples include call forwarding, transfer and call hold.

VALU

Call Coverage

Provides similar call coverage as DCS Call Coverage and Call Coverage Remote Off Net or C-CRON. Call will come back if covered over QSIG. The functionality will only be complete when all the switches are MultiVantage and using QSIG VALU. The Covered-to party can still receive Distinct Alerting.

Call Coverage and CAS

When a trunk has both CAS and VALU Call Coverage activated, the coverage display information is provided on calls that cover from a branch PBX to the main PBX. Path replacement will be attempted after coverage.

Distinctive Alerting

Provides distinctive ringing, internal and external, to the remote called party when the call is routed over the QSIG network.

Uniform Dial Plan (UDP)

A unique four- or five-digit number assigned to each station on the network. Uniform numbering gives each station a unique number (location code plus extension) that can be used at any location in the Electronic Tandem Network to access that station, MultiVantage enhances the standard UDP with the unrestricted 5-digit Uniform Dial Plan, which allows up to five digits to be parsed for call routing.

MultiVantage supports Uniform Dial Plans of up to 7-digits in length for local extensions, including stations, data modules, vectors, agent login IDs, etc. Administrators have the flexibility to administer dial plans between 3 and 7 digits in length, and MultiVantage supports mixed digit lengths in the same dial plan. QSIG is the required networking protocol if the dial plan between networked switches exceeds 5 digits in length. In other words, DCS only supports 3-5 digit dial plan, whereas QSIG supports between 3-7 digit dial plans.

Extended Trunk Access

Used with Uniform Dial Plan, allows the system to send any unrecognized number (such as an extension not administered locally) to another system for analysis and routing. Such unrecognized numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the Uniform Dial Plan table. Non-Uniform Dial Plan numbers are administered on either the First Digit Table (on the Dial Plan Record form) or the Second Digit Table. They are not administered on the Extended Trunk Access Call Screening Table. Extended Trunk Access helps you make full use of automatic routing and Uniform Dial Plan.

Extension Number Portability — When employees move within the network, they can retain their extension numbers. The ability to keep extension numbers, and even Electronic Tandem Network and Direct Inward Dialed numbers, when moving to other locations within the company eliminates missed calls and saves valuable time.

Data Interfaces

Administered Connections

Automatically establishes an end-to-end connection between two access or data endpoints based on administered attributes. This feature provides capabilities such as alarm notification, including an administrable alarm type and threshold; automatic restoration of connections established over a Software-Defined Data Network; ISDN-PRI trunk group [service may be referred to as ISDN-PRI (AC/AE) Service]; scheduled as well as continuous connections; and administrable-retry interval for failed connection attempts.

Data Call Setup

Enables the setting up of data calls using a variety of methods, such as: keyboard dialing, telephone dialing, Hayes command dialing, permanent switched connections, administered connections, automatic calling unit interface, and Hot Line dialing. Data Call Setup is provided for both DCP and ISDN-BRI telephones.

Data Hot Line

Provides for automatic placement of a data call when the originator hangs up. Data Hot Line may be used for security purposes. This feature offers fast and accurate call placement to commonly called data endpoints. Data terminal users who constantly call the same number can use Data Hot Line to automatically place the call when they hang up the telephone.

Data Modules

Data modules connect MultiVantage with other communications equipment, changing protocol, connections, and timing as necessary.

MultiVantage supports the following types of data module:

- High Speed Links
- Data stands
- Modular-processor data module
- 7000-series data modules
- Modular-trunk data module
- Asynchronous Data Unit
- Asynchronous Data Module (for ISDN-Basic Rate Interface telephones)
- Terminal adapters

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

Data Privacy

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data Privacy is activated when you dial an activation code at the beginning of the call.

Data Restriction

Protects analog data calls from being disturbed by any of the system's overriding or ringing features. It is administered at the system level to selected analog and multi-appearance telephones and trunk groups.

Default Dialing

Provides data terminal users who dial a specific number the majority of the time a very simple method of dialing that number. This feature enhances Data Terminal (Keyboard) Dialing by allowing a data terminal user to place a data call to a pre-administered destination in several different ways, depending on the type of data module. Data Terminal Dialing and Alphanumeric Dialing are unaffected.

IP Asynchronous Links

IP Asynchronous Links enable MultiVantage to transfer existing asynchronous adjunct connectivity to an Ethernet (TCP/IP) environment. IP Asynchronous Links support switch server applications, as well as client applications. MultiVantage can connect to System Management applications such as the Avaya Visibility Suite over the LAN. Call detail recording (CDR) devices, property management systems (PMS) and printers can be connected using asynchronous TCP/IP links.

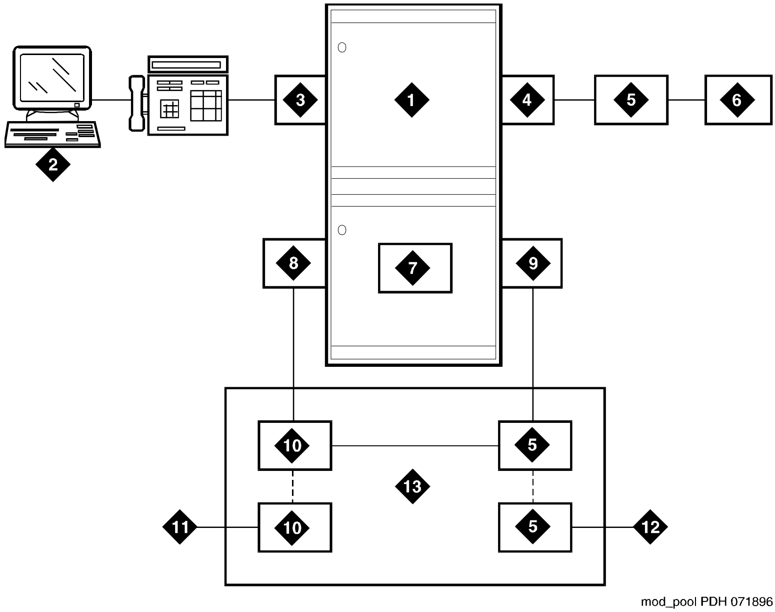
IP Asynchronous Links:

- Reduce the cost of connecting to MultiVantage for various adjuncts
- Allow for an open architecture to transport information and increases the speed at which data is transferred
- Allow customers to manage applications from on-site or remote locations
- Allow several system management applications to run on a single PC, thereby reducing hardware requirements
- Guarantee data delivery through a reliable session-layer protocol
- Support customers' existing serial hardware investment through use of Network Terminal Servers

Modem Pooling

Enables switched connections between digital data endpoints (data modules) and analog data endpoints and acoustic coupled modems. Data transmission between a digital data endpoint and an analog endpoint requires a conversion since the DCP format used by the data module is not compatible with the modulated signals of an analog modem. A modem translates DCP format into modulated signals and vice versa. The Modem Pooling feature provides a set of modems for such conversions.

MultiVantage modem pools are assigned into modem pool groups. A group can have up to 32 modems, called “members.” MultiVantage can have as many as 63 modem pool groups. See Figure 6 on page 110.



1	MultiVantage	7	Integrated Pooled Modem
2	Asynchronous Terminal	8	Data Line Port
3	Digital Port	9	Analog Port
4	Analog Trunk	10	7400A
5	Modem	11	Digital Communications Protocol
6	Remote Application	12	Analog
		13	EIA Standard

Figure 6. MultiVantage Modem Pooling

Multimedia Applications Server Interface

The Multimedia Applications Server Interface provides a link between the MultiVantage and one or more Multimedia Communications eXchange nodes. A Multimedia Communications eXchange is a stand-alone multimedia call processor produced by Avaya. This new link to MultiVantage enhances the capabilities of each Multimedia Communications eXchange system by enabling it to share some of the MultiVantage features. In particular, the interface provides the following advantages:

- Call Detail Recording (CDR)— The capture of call detail records so you can analyze the call patterns and usage of multimedia calls just as MultiVantage administrators analyze normal calls.
- Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) — The intelligent selection of the most cost-effective routing for calls, based on available resources and your carrier preference. The system may select public trunks via DEFINITY Multimedia eXchange (MMCX)
- Voice Mail Integration — You can access your EMBEDDED AUDIX or Intuity AUDIX voice messaging system from a Multimedia Communication eXchange (MMCX).

Multimedia Calling

Multimedia calls are initiated with voice and video only. Once a call is established, one of the parties may initiate an associated data conference to include all of the parties on the call who are capable of supporting data. The data conference is controlled by an adjunct device called an Expansion Services Module (ESM).

Multimedia Call Early Answer on Vectors and Stations

Early Answer is a feature applied to multimedia calls in conjunction with conversion to voice. Early Answer:

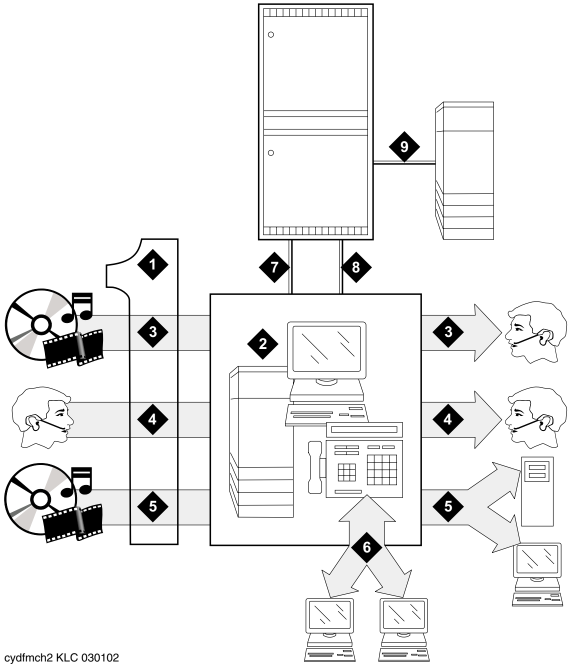
- Answers the data call
- Establishes the multimedia protocol prior to completion of a converted call
- Ensures that a voice path to/from the originator is available when the (voice) call is answered

For an incoming call, Early Answer answers the dynamic service-link calls when the destination endpoint answers, unless Early Answer is specified during routing or termination processing.

NOTE:
The “destination voice endpoint” might be an outgoing voice trunk if the destination voice station is forwarded or covered off-premises.

Multimedia Call Handling (MMCH)

Multimedia Call Handling (MMCH) enables you to control voice, video, and data transmissions using your telephone set. The feature buttons on a multi-function MultiVantage telephone enable you to conduct video conferences, and forward, cover, hold, or park multimedia calls much as you would a standard voice call. You can also share PC applications so that you and colleagues can collaborate while working from remote sites. See Figure 7 on page 113.



1 One number access

2 Multimedia call complex

3 Multimedia to voice conversion

4 Standard voice call handling

5 Call redirection

6 Multimedia conferencing

7 BRI data connection

8 DCP voice connection

9 ESM data collaboration

Figure 7. MultiVantage Multimedia Call Handling

Multimedia Call Redirection to MM Endpoint

A dual port multimedia station may be a destination of call redirection features such as call coverage, forwarding, and station hunting. The station can receive and accept full multimedia calls or data calls converted to multimedia.

Multimedia Data Conferencing (T.120) via ESM

The data conference is controlled by an adjunct device called an Expansion Services Module (ESM). The Expansion Services Module is used to terminate T.120 protocols [including Generalized Conference Call (GCC), a protocol standard for data conference control] and provide data conference control and data distribution. The MultiMedia Interface circuit pack, TN787, is used to rate adapt T.120 data to/from the ESM.

Multimedia Hold, Conference, Transfer, and Drop

Station users have the ability to activate hold, conference, transfer, or drop on multimedia calls. Multimedia endpoints and voice-only stations may participate in the same conference.

Multimedia Multiple-Port Networks

MultiVantage supports the equivalent of 580 Basic mode complexes operating at 6CCS traffic level. All enhanced mode complexes operate with soft-mode service links since the use of hard-mode service links reduces capacities. G3si limits are 1/3 to 1/2 of the G3r limits, depending on memory limitations and port network limitations.

Pass Advice of Charge Information to World Class BRI Endpoints

Provides Advice of Charge (AOC) information to World Class BRI (WCBRI) endpoints. On a call using a WCBRI endpoint, AOC information will be displayed on the endpoint after the call has completed and the far end has hung up.

Call Routing

Alternate Facility Restriction Levels

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

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

Automatic Routing Features

MultiVantage 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)

Allows private network calls to originate and terminate at one or many locations without accessing the public network. When you dial an access code and phone number, AAR selects the most desirable route for the call and performs digit conversion as necessary. If the first choice route is unavailable, another route is chosen automatically.

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

Automatic Route Selection (ARS)

ARS selects carriers automatically and routes calls inexpensively over the public network. When there are one or more long-distance carriers and Wide-Area Telecommunications Services (WATS) provided, MultiVantage selects the most preferred route for the call. Long-distance carrier-code dialing is not required on routes selected by the system. You assign long-distance carrier-codes and MultiVantage translates them. The system inserts codes as needed to guarantee automatic-carrier selection. ARS can route calls to a variety of types-of-numbers and access a variety of types of trunk groups.

AAR/ARS Overlap Sending

MultiVantage supports overlap sending for AAR and ARS calls that are routed over ISDN-PRI trunk groups. ISDN-PRI call-address information is sent one digit at a time instead of in one block. In countries with complex public-network numbering plans, this allows for a significant decrease in call setup time. When overlap receiving is enabled, this is especially significant for tandem calls.

AAR/ARS Partitioning

Allows AAR and ARS to be partitioned into 8 user groups within a single MultiVantage and provides individual routing treatment for each of these user groups.

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

Generalized Route Selection

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

Look-Ahead Routing

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

Node Number Routing

Allows you to specify the route pattern associated with each node in a private network. It is a required capability for Extension Number Portability and is used in conjunction with Automatic Route Selection, AAR and ARS Partitioning, Private Networking, and Uniform Dial Plan. Uniform Dial Plan extensions can be routed to a specified node using its associated pattern. Node Number Routing allows a Uniform Dial Plan route pattern based on node numbers or on location codes. On the AAR and ARS Digit Analysis Tables, you also can specify a Node Number instead of a Route Pattern.

Time of Day Routing

Provides the most economical routing of ARS and AAR calls. This routing is based on the time of day and day of the week that each call is made. Up to 8 TOD routing plans may be administered, each scheduled to change up to 6 times a day for each day in the week.

This allows you to take advantage of lower calling rates during specific times of the day and week. In addition, companies with locations in different time zones can use different locations that have lower rates at different times of the day or week. This feature is also used to change patterns during the times an office is closed in order to reduce or eliminate unauthorized calls.

Multiple Location Support

Multiple Location Support enables local user time, local ARS Public Analysis Tables for local trunking, automatic Daylight Savings Time, and enhances shared resource algorithms (touch tone receivers) when Remote Expansion Port Networks (EPNs), ATM Port Networks, and Avaya Media Gateways are remoted off of a central server at a different location.

Traveling Class Marks

Traveling Class Marks are a mechanism for passing a caller’s facility restriction level from one Electronic Tandem Network switch to another. Traveling Class Marks allow privilege checking to be passed across switches through the Electronic Tandem Network.

Miscellaneous

Answer Detection

For purposes of Call-Detail Recording (CDR), it is important to know when the called party answers a call. MultiVantage provides three ways to determine whether the called party has answered an outgoing call.

Answer supervision by time-out

You set a timer for each trunk group. If the caller is off-hook when the timer expires, the MultiVantage assumes that the call has been answered. This is the least accurate method. Calls that are shorter than the timer duration do not generate call records, and calls that ring for a long time produce call records whether they are answered or not.

Call-classifier board

A call-classifier board detects tones and voice-frequency signals on the line and determines whether a call has been answered. This method is fairly accurate.

Network answer supervision

The Central Office (CO) sends back a signal to indicate that the far end has answered. If a call has traveled over a private network before reaching the CO, the signal is transmitted back over the private network to the originating system. This method is extremely accurate, but is not available in the United States over CO, FX, or WATS trunks.

12 — Reliability and Survivability

Alternate Gatekeeper

The Alternate Gatekeeper enhancement can provide survivability between MultiVantage and IP communications devices such as IP Telephones and IP Softphones. This is accomplished by providing alternate gatekeepers (C-LAN) in the event of network or gatekeeper failure and by load balancing endpoint traffic among multiple gatekeepers. It is important to recognize that calls will drop during that interval while the communication is re-established to the switch.

ATM WAN Spare Processor (WSP)

See “WAN Spare Processor (WSP)” on page 78.

Multiple Network Regions per C-LAN

Multiple Network Regions per C-LAN enables a single C-LAN to provide registration and call control to IP endpoints in multiple network regions. MultiVantage implements this approach by allowing IP address to be mapped to network regions in a mapping form, instead of just to a C-LAN. When an IP phone registers, the switch will determine the phone's network region number based on the phone's IP address.

Power Failure Transfer

Provides service to and from the local telephone company Central Office, including Wide Area Telecommunications System, during a power failure. This allows you to make or answer important or emergency calls during a power failure. This feature is also called Emergency Transfer.

Survivable Remote EPN (SREPN)

The Survivable Remote Expansion Port Network (SREPN) allows a DEFINITY ECS (R6r or later) EPN to provide service to the customer when the link to the main processor fails or is severed or when the processor or CSS fails. When the links to the system are restored and stable, the logic switch is manually reset and the EPN is reconnected to the links from the switch. There are both command and manual resets. The resets can be done remotely at the SAT or manually at the equipment.

The SREPN must be administered separately (not as a duplicated PPN) to function in a disaster recovery scenario. It does not function as a survivable remote EPN without the administration (stations, trunks, features) to support its operation.

NOTE:

SREPN is not compatible with ATM Port Network Connectivity (ATM-PNC). If that’s the case, see ‘‘WAN Spare Processor (WSP)’’ on page 78.

13 — Security, Privacy, and Safety

System Administrator

Access Security Gateway

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

Alternate Facility Restriction Levels

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

CAUTION:

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

Alternate Operations Support System Alarm Number

Allows you to establish a second number for the MultiVantage to call when an alarmable event occurs. This feature is useful for alerting a second support organization, such as INADS or OneVision.

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.

Authorization Codes — 13 Digits

Authorization codes extend calling-privilege control and enhance security for remote-access callers. Authorization codes can be up to 13 digits in length.

Avaya Site Administration Authorization codes may be used to:

- Override facility restriction levels assigned to originating stations or trunks
- Restrict individual incoming tie trunks and remote-access trunks from accessing outgoing trunks
- Track CDR calls for cost-allocation purposes
- Provide additional security control

Call Restrictions

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

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

Class of Restriction (COR)

Defines many different classes of call origination and termination privileges. MultiVantage may have no restrictions, only a single COR, or may have as many classes of restrictions as necessary to effect the desired restrictions. Many different types of classes of restriction can be assigned to many types of facilities on the switch. For example, you can use a calling-party COR to prevent callers from accessing the public network.

Block Collect Call

See “Block Collect Call” on page 56.

Customer-Provided Equipment Alarm

Provides you with an indication that a system alarm has occurred and that the MultiVantage has attempted to contact a service organization. A device that you provide, such a lamp or a bell, is used to indicate the alarm situation. You can administer the level of alarm about which you want to be notified.

Data Privacy

Protects analog data calls from being disturbed by any of the system’s overriding or ringing features. Data Privacy is activated when you dial an activation code at the beginning of the call.

Data Restriction

Protects analog data calls from being disturbed by any of the system’s overriding or ringing features. It is administered at the system level to selected analog and multi-appearance telephones and trunk groups.

Facility Restriction Levels and Traveling Class Marks

Allows certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed to use Central Office trunks to other corporate locations while other users may be restricted to less expensive private-network lines. You can administer up to eight levels of restriction for users of AAR and ARS.

Malicious Call Trace

Allows you to trace malicious calls. You define a group of terminal users who can notify others in the group when they receive a malicious call. These users can then retrieve information related to the call. Using this information, you can identify the malicious call source or provide information to personnel at an adjacent system to complete the trace. It also allows you to record the malicious, call as well as trace a malicious call over ETSI PRI.

Restriction — Controlled

Allows an Attendant or telephone user, with console permission, to activate and deactivate for an individual telephone or a group of telephones, the following restrictions: outward, total, station-to-station, and termination restrictions.

Security Violation Notification (SVN)

Security Violation Notification (SVN) allows you to set security-related parameters and to receive notification when the limits that you have established are violated. You can run reports related to both valid and invalid access attempts. You can also disable a login ID or remote access authorization that is associated with a security violation.

Station Security Codes

To provide additional security around the customer options the “init” login has been provided with additional security for the purpose of establishing an authentication procedure for attempts to remotely log into the system.

End 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.

Barrier Codes

A security code used with Remote Access to prevent unauthorized access to your system. To increase your system's security, use a 7-digit barrier code with Remote Access Barrier Code Aging. A barrier code automatically expires if an expiration date or number of accesses has exceeded the limits you set. If both a time interval and access limits are administered for a barrier code, the barrier code expires when one of the conditions is satisfied.

NOTE:
Barrier codes are not tracked by Call Detail Recording (CDR). Barrier codes are incoming access codes, whereas, authorization codes are primarily outgoing access codes.

Calling/Connected Party Number (CPN) Restriction

Per Call CPN Restriction

Users may indicate Calling Number privacy information. For ISDN calls, the CPN Presentation Indicator is encoded accordingly. For non-ISDN calls going to a public network that supports the CPN Restriction feature, the network specific Feature Activation Code gets passed to the network for interpretation and activation of the desired feature.

If Per Call CPN Restriction is activated for an outgoing call, it will override any Per Line CPN Restriction administration for the calling station, and will override any ISDN Trunk Group administration for sending Calling Number.

Per Line CPN Restriction

Users may block the Calling Party Number when originating calls. For ISDN calls, the CPN Presentation Indicator is encoded accordingly. For non-ISDN calls, going to a public network that supports the CPN Restriction feature, the network specific Feature Activation Code gets passed to the network for interpretation and activation.

If Per Line CPN Restriction is administered for a station, it will override any ISDN Trunk Group administration for sending Calling Party Number.

Crisis Alerts To a Digital Numeric Pager

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.

Crisis Alerts To a Digital Station

Crisis Alert uses both audible and visual alerting to notify administered digital display stations 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 (or room). 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.

When crisis alerting is active, the station is placed in position-busy mode so that other incoming calls can not interfere with the emergency call notification. The station can still originate calls to allow notification of other personnel.

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 Alerts To an Attendant Console

Crisis Alert uses both audible and visual alerting to notify attendant consoles 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 (or room). 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. Though often used in the hospitality industry, it can be set up to work with any standard attendant console.

When crisis alerting is active, the console is placed in position-busy mode so that other incoming calls can not interfere with the emergency call notification. The console can still originate calls to allow notification of other personnel. Once a crisis alert call has arrived at a console, the console user must press the position-busy button to unbusy the console, and press the crisis-alert button to deactivate audible and visual alerting.

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.

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.

E911 CAMA Trunk Group

See “E911 CAMA Trunk Group” on page 83.

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.

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.

Restriction — Controlled

See “Restriction — Controlled” on page 124.

Station Lock

Station Lock allows users to lock their phones to prevent unauthorized outgoing calls. Users can block outgoing calls and still receive incoming calls. This feature is activated by pressing a phone button or dialing a feature access code (FAC). Station Lock allows users to block all outgoing calls except for emergency calls. Phones can be remotely locked and unlocked.

14 — Special Applications

Special Applications are those custom features developed by the Avaya Global Rapid Response Team to meet a particular customer's need. Each feature is ordered through the Rapid Response Team as an al la carte item. Special ordering and provisioning procedures apply. Contact your Avaya Sales representative or Authorized Avaya Business Partner for more information.

The features available include the following:

- Support Connectivity with Northern Telecom DMS100/250 Names Display (Setup Method - Names Transparency)
- Pickup of Attendant Calls Using TAAS During Day Service
- Record Actual Answering Party on Call Detail Recording
- Flash to Answer Call Waiting/Hold
- Cancel ARS by Dialing "*"
- External Coverage Path Changes to Internal Path when Night Service Active
- Enhanced Emergency Alert to a Station
- External Coverage Path to be Used When a Trunk Originated Call is on Soft Hold
- Expand DS1's to 332 Line Side Only (DEFINITY Server R Only)
- Integrated Directory Service Over DCS (IDS+)
- Administrable Conference Tones by Class of Service
- Enhanced Display for 8434 Terminals on Redirected Calls
- Enhanced Display for 8434, 8434D, 7444D, 7407+, 7407D and Callmaster Terminals on Redirected Calls and Bridged Appearances
- Display Incoming Digits for ISDN Trunk Groups
- Night Service on DID Trunk Group
- Display UUI Information
- Enhanced DID Routing
- Vector Collect # and * Literally Option
- Service Observe Physical Set

- Busy Tones on Send All Calls with No Available Coverage Points
- 80000 UDP Extension Records (DEFINITY Server R Only)
- Dial by Name
- Variable Length Account Codes
- 25,000 Facility Busy Indicators (DEFINITY Server R Only)
- ISDN Redirecting Number
- Enhancement added to Support Country Version 1a
- Russia Power Industry - Russia ONLY
- Support Calling Party Category on QSIG Code Set 5
- Attendant Dial 0 Redirect
- Listed Directory Number (LDN) Attendant Queue Priority
- Omit Designated Extensions from Station Displays
- Update Display for Redirected Calls
- Priority Attendant Queuing by COR
- Toll-Free Announcements until Answered (in Vectoring with ISDN Trunks)
- CDR Start Time/Provide Date and Time in Hour, Minute & Seconds
- Prime Line Preference
- Idle Appearance Preference Display Enhancement
- Allow Station Users to Program their Own Facility Busy Indicators on 6400 and 8400 Series Terminals
- XSTATION Support with the DENSO 300M - Japan ONLY
- UUI for Universal Caller ID in Codeset 6
- Station User Button Ring Control
- Delay ISDN Connect on Agent Answer/Prevent Vector ISDN Alerting
- Forward Held-Call CPN (Calling Party Number) for Call Transfer/Conference
- Enhancement to QSIG Rerouting for Call Forwarding - Don’t Strip ARS/AAR Access Code (9) when Forwarding Digits from DEFINITY ECS to IPC Turret
- Expand the Number of Coverage Paths (DEFINITY Server R Only) to 2000 and Remote Cover Points.
- Media Encryption for VoIP

15 — System Management

MultiVantage System Management provides the administrator powerful tools to maintain their communication solutions and to drive down the Total Cost of Ownership.

Administration Without Hardware (AWOH)

See "Administration Without Hardware (AWOH)" on page 69.

Alternate Facility Restriction Levels

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

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

Announcements

Multiple Music Sources

This feature allows the customer to provide multiple distinct music sources for use with the Call Vectoring features, calls placed on hold, calls awaiting pickup, etc.

Music-on-Hold Access

Automatically provides music, silence, or tone to a caller. Music lets the caller know that the connection is still valid. Many different music options can be administered to accommodate different tenants on MultiVantage. See ‘‘Tenant Partitioning’’ on page 141 for more information.

Recorded Announcement

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

Voice Announcement over LAN (VAL)

Voice Announcement over LAN (VAL) introduces the TN2501AP, a new integrated announcement circuit pack that:

- plays announcements over the TDM bus, similar to the TN750C.
- has up to 1 hour of announcement storage time per circuit pack.
- has 33 ports (31 playback, 1 record, and 1 ethernet).
- supports a 10/100 Mb ethernet interface, allowing announcement and firmware file portability over a LAN (FTP server functions).
- supports generated .wav announcement files.

Avaya Voice Announcement over LAN (VAL) Manager

Avaya Voice Announcements over LAN (VAL) Manager is part of the Avaya Visability Suite of products. It enables you to the use of a LAN to transfer recorded announcements to Avaya Media Servers.

Announcements can be stored in .wav files, which can be sent to a Voice Announcement over LAN board without conversion. The Voice Announcement over LAN Manager also provides a repository to backup and restore announcement files, and simplifies administration. With Voice Announcement over LAN Manager, you can view the current status of announcements, easily add, change, and remove announcements, and copy and backup announcement files from Avaya Media Servers to the Voice Announcement over LAN Manager and back, through the LAN.

Authorization Codes — 13 Digits

See “Authorization Codes — 13 Digits” on page 122.

Automatic Circuit Assurance

Assists in identifying possible trunk problems. MultiVantage maintains a record of the performance of individual trunks and automatically calls a designated user when a possible failure is detected. This feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time.

Automatic Transmission Measurement System

Measures voice and data trunk facilities for satisfactory transmission performance. The measurement report contains data on trunk signal loss, noise, signaling return loss, and echo return loss. Acceptable performance, the scheduling of tests, and report contents are administrable.

Barrier Codes

A security code used with Remote Access to prevent unauthorized access to your system. To increase your system’s security, use a 7-digit barrier code with Remote Access Barrier Code Aging. A barrier code automatically expires if an expiration date or number of accesses has exceeded the limits you set. If both a time interval and access limits are administered for a barrier code, the barrier code expires when one of the conditions is satisfied.

NOTE:
Barrier codes are not tracked by Call Detail Recording (CDR). Barrier codes are incoming access codes, whereas, authorization codes are primarily outgoing access codes.

Bulletin Board

Provides a place on the switch where you can post information and receive messages from other switch users, including Avaya personnel. Anyone with appropriate permissions can use the bulletin board for everyday messages. In addition, Avaya personnel can leave high-priority messages, which are displayed on the first 10 lines of the bulletin board.

Busy Verification of Telephones and Trunks

Allows Attendants and users of multi-appearance telephones to make test calls to trunks, telephones, and hunt groups to check the status of an apparently busy resource. With this feature, an Attendant or multifunction telephone user can distinguish between a telephone that is truly busy and one that only appears busy because of some problem. You can also use the feature to quickly identify faulty trunks.

Call Charge Information

Provides two ways to know the approximate charge for calls made on outgoing trunks:

- Advice of Charge — For ISDN trunks

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

- Periodic Pulse Metering — For non-ISDN trunks

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

Call-charge information helps you to account for the cost of outgoing calls without waiting for the next bill from your network provider. This is especially important in countries where telephone bills are not itemized. You can also use this information to let employees know the cost of their phone calls, and so encourage them to help manage the company's telecommunications expenses.

NOTE:
This feature is not offered by the public network in some countries, including the United States.

In addition, the Pass Advice of Charge to BRI endpoints feature will transparently pass AOC information that has been received from PRI networks to WCBRI endpoints.

Call Detail Recording (CDR)

Records detailed call information on incoming and outgoing calls for the purpose of call accounting and sends this call information to a Call Detail Recording (CDR) output device. You can specify the trunk groups and extensions for which you want records to be kept as well as the type of information to be recorded. You can keep track of both internal and external calls. This application contains a wide variety of administrable options and capabilities.

Call Restrictions

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

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

Calling Party/Billing Number (CPN/BN)

Allows the system to transmit Calling Party Number/Billing Number (CPN/BN) information to an ISDN-PRI trunk group. The CPN is the calling party’s telephone number. BN is the calling party’s billing number. The CPN/BN may contain international country codes. It is used with an adjunct application.

Class of Restriction (COR)

See “Class of Restriction (COR)” on page 123.

Class of Service (COS)

Defines whether or not telephone users *can* access the following features and functions: Automatic Callback, Call Forwarding, Data Privacy, Priority Calling, Restrict Call Forwarding Off-Net, Call Forward Busy/Don’t Answer, Personal Station Access, Extended Forwarding and Busy/Don’t Answer, Trunk-to-Trunk Transfer Restriction Override, Off-Hook Alert, Console Permission, or Client Room.

Classless Interdomain Routing (CIDR)

See “Classless Interdomain Routing (CIDR)” on page 95.

Concurrent User Sessions

In order to increase the efficiency of administration and maintenance functions, the MultiVantage switch accommodates multiple concurrent administration and maintenance user sessions. Three or more devices (management terminals or operation support systems) can be connected to the switch to perform administration and/or maintenance tasks simultaneously. MultiVantage supports eight concurrent administration and maintenance users — five can perform concurrent administration, and three can perform concurrent maintenance. The eight concurrent sessions can be in any combination of local and remote connections.

Customer-Provided Equipment Alarm

See “Customer-Provided Equipment Alarm” on page 123.

Customer Telephone Activation (CTA)

Enables customers to install their own phones, eliminating the need for a service technician to do the installation. This feature is based on the TTI feature and allows the customer to associate a physical phone with a station translations switch. CTA is a streamlined version of TTI; it has a fixed feature-access code but does not require a security code. In addition, CTA allows only for “merging” of phones with station translations, whereas TTI allows for both “merging” and “unmerging” of phones with station translations. CTA applies only to DCP and analog touch-tone, circuit-switched phones.

DCS Automatic Circuit Assurance

Allows a user or Attendant at one node to activate or deactivate Automatic Circuit Assurance referral calls for the entire DCS network. This transparency allows the referral calls to originate at a node other than the node that detects the problem.

External Device Alarming

Allows you to assign analog ports to alarm interfaces for external devices. You can specify a port location, information to identify the external device, and the alarm level to report when a contact closure occurs.

Facility Busy Indication

Allows users of multi-appearance telephones to see which lines, trunk groups, terminating extension groups, hunt groups, or paging zones (called resources or facilities) are busy. When the lamp associated with the resource is lit, the resource is busy.

You can store extension numbers, trunk group access codes, and Loudspeaker Paging access codes in a Facility Busy Indication button. The Facility Busy Indication button provides direct access to any of the facilities.

Facility Restriction Levels and Traveling Class Marks

Allows certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed to use Central Office trunks to other corporate locations while other users may be restricted to less expensive private-network lines. You can administer up to eight levels of restriction for users of AAR and ARS.

Facility Test Calls

Allows telephone users to make test calls to access specific trunks, dual tone multifrequency receivers, time slots, and system tones. The user dials an access code and makes the test call to make sure the facility is operating properly. Security measures are included to prevent unauthorized use.

Firmware Download

The firmware download feature makes it possible to download an image from a remote or local source into MultiVantage and use that image to reprogram the application code of a port circuit pack. This feature makes updating firmware more cost effective. It also reduces the expense of servicing the MultiVantage port circuit packs because it eliminates the need for a technician to be involved when a board is updated. Firmware download is achieved using the TN799C C-LAN interface.

NOTE:
Circuit packs that can be updated with the firmware download feature have a “P” at the end of their TN number.

Information and Reports

- Attendant Position Report

  The Attendant Position report lists the following:

  - Attendant usage
  - Number of calls answered
  - Total time the attendant was available to answer a new call
  - Average holding time on calls answered
- Blockage Study Report
- Call Coverage Reports

  The Call Coverage report displays measurements of the distribution of traffic offered to call-coverage groups. Separate reports for all calls and external calls are supplied.
- Coverage Points Report

The Coverage Points report differs based on whether “All Calls” or “External Calls” is selected. For each coverage point in the group, the number of calls offered, abandoned while at that coverage point, and overflowing to the next coverage point are listed.

- Display ARP Reports
- Emergency and Journal Reports

  The Emergency and Journal report is based on information from all crisis alerts.
- Hunt Group Measurements Report
- IP Reports
- Packet Error History

  Provides a 24-hour history of important packet level statistics that indirectly indicate some LAN performance characteristics. The 24-hour history gives the ability to look back at these measures if the trouble cleared.
- Port Network and Link Usage Report
- Processor Occupancy Report

  The Processor Occupancy report provides summary information on how heavily the processor is loaded.
- Recent Change History Report

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

  The Summary report provides a performance summary of MultiVantage.
- Tandem Traffic Report

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

Traffic reports show measurements 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.

- Trunk Group Detailed Measurements

IP Asynchronous Links

See “IP Asynchronous Links” on page 108.

Malicious Call Trace

Allows you to trace malicious calls. You define a group of terminal users who can notify others in the group when they receive a malicious call. These users can then retrieve information related to the call. Using this information, you can identify the malicious call source or provide information to personnel at an adjacent system to complete the trace. It also allows you to record the malicious call.

Malicious Call Trace Logging

Malicious Call Trace Logging allows a PC to receive information from MultiVantage to log malicious calls.

Restriction — Controlled

Allows an Attendant or telephone user, with console permission, to activate and deactivate for an individual telephone or a group of telephones, the following restrictions: outward, total, station-to-station, and termination restrictions.

Scheduling

Functional scheduling in MultiVantage allows you to specify the time a command will be executed or to 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.

Security Violation Notification (SVN)

Security Violation Notification (SVN) allows you to set security-related parameters and to receive notification when the limits that you have established are violated. You can run reports related to both valid and invalid access attempts. You can also disable a login ID or remote access authorization that is associated with a security violation.

Station Security Codes

See “Station Security Codes” on page 124.

Tenant Partitioning

Allows partitioning of the MultiVantage in order to lease the system’s services and features to tenants. This provides attractive new services and revenue for “virtual” landlords. It provides the robust features of a large system at affordable rates to small business tenants. MultiVantage supports up to 100 partitions and 27 Attendant Groups. Multiple Attendant Groups can be assigned to each partition. Stations, hunt groups, and other endpoints assigned to a Class of Service can be partitioned. Network routing pattern preferences also support the assigned Tenant Partitioning. Tenant Partitioning also allows you to assign a unique music source for each tenant partition for customers who are put on hold.

See also, ‘‘Music-on-Hold Access’’ on page 132.

Terminal Translation Initialization (TTI)

MultiVantage provides Terminal Translation Initialization (TTI), a feature that works with Administration Without Hardware. 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 terminal that is connected to a wired (but untranslated) jack.

Time of Day Clock Synchronization via LAN Source

Customers need accurate and common time of day time source across multiple switches in a network. This is especially important when customers are using a central Avaya Call Management System (CMS) to report events coming from multiple servers running MultiVantage.

The Time of Day Clock Synchronization via LAN Source feature is implemented in two different ways:

Linux Platforms

MultiVantage running on Linux-based Media Servers, such as the Avaya S8300 Media Server or the Avaya S8700 Media Server, synchronizes time directly from a LAN source such as a server.

UNIX Platforms

MultiVantage running on DEFINITY Servers which use Oryx/Pecos operating system (proprietary UNIX-based OS) receives a command from Avaya Site Administration to adjust the time. Avaya Site Administration is synchronized to the LAN PC's clock.

Trunk Group Circuits

Trunks provide the communications links between MultiVantage 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 MultiVantage via port circuit packs.

Variable Length Ping

See "Variable Length Ping" on page 99.

Variable Length Subnet Mask (VLSM)

See "Variable Length Subnet Mask (VLSM)" on page 99.

Avaya VisAbility Management Suite

Avaya VisAbility Software is Systems Management software that contains applications to manage a converged voice and data network. The applications include network management, fault management, performance management, configuration management, directory management and policy management functionality.

Avaya ATM WAN Survivable Processor Manager

Avaya ATM-WAN Survivable Processor Manager can be a key part of your emergency restoration and business continuity planning. This application enables users to download translations from a main MultiVantage server, and simultaneously upload those translations to multiple (up to 15) ATM WAN Spare Processors (WSPs) over a LAN connectivity, according to a schedule specified by the administrator.You can schedule translations to run once, now or for a specified time and date in the future, or schedule regular daily or weekly updates.

The module also provides the ability to schedule regular daily or weekly updates of the MultiVantage translations. The ATM WAN Survivable Processor Manager provides the current status of the main MultiVantage server and any defined WSP devices in the network. A complete history log is created listing each of the switches, and the time and the resulting message from the scheduled action. On-line help is embedded into the module for ease of use.

Avaya MultiVantage Configuration Manager

Avaya MultiVantage Configuration Manager provides centralized management of distributed network and campus environments, using a single point of entry and graphical Web-based interface for configuration and administration of multiple Avaya Media Servers.

Avaya MultiVantage Fault/Performance Manager

Avaya MultiVantage Fault/Performance Manager integrates with Avaya MultiService Network Manager to provide a system view of your converged network. Fault Manager displays a hierarchical view of devices and their status, allowing you to view and isolate alarms and errors. Performance Manager provides a comprehensive set of performance reports for trending and isolation of performance issues.

Avaya Site Administration

Avaya Site Administration is a Microsoft Windows-based graphical user interface for making changes, adding or moving users, and performing basic traffic analysis.

Avaya Voice Announcement over LAN (VAL) Manager

See “Avaya Voice Announcement over LAN (VAL) Manager” on page 132.

Avaya VoIP Monitoring Manager

Avaya VoIP Monitoring Manager (VMON) provides the ability to monitor VoIP network quality. This web-based application receives QoS statistics from Avaya IP end points and displays the data via graphs and reports, so administrators can isolate voice quality problems and send traps when poor voice quality is detected.

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.

Administration Change Notification

Enables MultiVantage to communicate with the Avaya Directory Enabled Management (DEM) client. This feature enables the client to have real-time, integrated, directory-based, read/write access to MultiVantage administration data based on rules defined by the customer. Administration Change Notification enables the client to subscribe to notifications of changes to administration data in MultiVantage. It thus provides real-time updates whenever administration changes occur in a particular object (for example, a station).

Avaya Directory Enabled Management

Avaya Directory Enabled Management (DEM) is part of the Avaya VisAbility Management Suite, and provides real-time, integrated, directory-based read/write access to Avaya Media Servers and Intuity AUDIX messaging servers. It streamlines workflow and information management in an electronic environment using converged networks.

DEM creates a meta-directory for converged voice and data networks. It synchronizes directory information with data from MultiVantage and Intuity devices, and stores the information in an LDAP-compliant directory service (for example, Novell’s eDirectory or Microsoft’s Active Directory). Directory-enabled applications can then use the DEM to implement workflow processes that automate various system management functions and speed business operations.

LDAP

Lightweight Directory Access Protocol version 3 (LDAPv3) is an industry compliant protocol for accessing online directory services. A directory is like a database, but tends to contain more description information. MultiVantage integrates with LDAP datastores through the use of the Administration Change Notification feature and Avaya Directory Enabled Management client application to provide real-time, integrated, directory-based read/write access to MultiVantage and Intuity AUDIX messaging servers.

16 — Telecommuting and Remote Office

Avaya R300 Remote Office Communicator (R300)

The R300 feature offers a cost-effective method for providing full functionality at a remote site. The R300 provides remote telephony that has all the capabilities of telephony that is connected directly. Through the R300, voice and data can share the same WAN link between MultiVantage and the remote site, thus providing voice and data convergence.

The R300 acts like a simple switch at the remote site to connect remote stations and local access trunks. It supports VoIP and DCP, as well as analog line and trunk connections. In addition, each R300 unit supports 12 remote dial access data channels. A single MultiVantage switch can support multiple R300 units. The number of units supported by MultiVantage varies according to the MultiVantage model type.

Coverage of Calls Redirected Off-Net (CCRON)

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.

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.

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.

IP Endpoint — Road-warrior mode

See “IP Endpoint — Road-warrior mode” on page 76.

IP Endpoint — Telecommuter mode

See “IP Endpoint — Telecommuter mode” on page 76.

IP Softphones

See “Avaya IP Softphones” on page 47.

Off-Premises Station

A trunk-data module connects off-premises private-line trunk facilities and MultiVantage. 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.

See also, ‘‘Call Redirection’’ on page 152.

See also, ‘‘Call Vectoring’’ on page 23.

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.

17 — Telephony

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

Labeling of Abbreviated Dialing (AD) Buttons on Softkeys allows users of the 2420 DCP telephone, as well as the 4600-, 6400-, and 8400-series display telephone sets, to administer labels for the AD buttons that appear on their softkeys. These personalized labels appear on the menu display. These labels apply to any AD buttons you have administered on the 2420 DCP and the 4620 IP telephones.

Abbreviated Dialing On-Hook Programming

On-Hook Programming allows users of the 2420 DCP telephone, as well as the 4600-, 6400-, and 8400-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.

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.

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. The Administrable Timeout on Call Timer feature allows the user to specify how long to display the duration of the call.

Alphanumeric Dialing

See “Alphanumeric Dialing” on page 49.

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.

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.

Bellcore Calling Name ID

Allows the system to accept calling name information from a local exchange carrier (LEC) network that supports the Bellcore calling name specification. The system can send calling name information in the format if Bellcore Calling Name ID is administered. The following Caller ID protocols are supported.

- Bellcore (default) - US protocol (Bellcore transmission protocol with 212 modem protocol)
- V23-Bell - Bahrain protocol (Bellcore transmission protocol with V.23 modem protocol).

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.

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.

Changeable Coverage Paths

Changeable Coverage Paths allows the end user to modify the coverage points by using a Feature Access Code.

Time of Day

This feature allows a user to have multiple coverage paths depending on the time of day, and day of the week.

Call Redirection

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 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 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.

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 MultiVantage 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.

Caller ID (ICLID) on Analog Trunks

See “Caller ID (ICLID) on Analog Trunks” on page 89.

Caller ID (ICLID) on Digital Trunks

See “Caller ID (ICLID) on Digital Trunks” on page 89.

Circular Station Hunting

This feature will eliminate the “hot seat” in a hunt group. The MultiVantage will keep track of the last extension in the hunt group that has received a call. When another incoming call arrives, the next idle extension will receive the call, 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.

Conferencing

See “Collaboration” on page 35.

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.

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.

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.

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.

Special Ringing

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).

Enhanced Telephone Display

The Enhanced Telephone Display feature allows you to choose the character set that you want to see in MultiVantage 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.

Go to Cover

Allows users who call another internal extension to send the call directly to coverage.

Hold

Allows you to disconnect from a call temporarily, use your telephone for other call purposes, and then return to the original call.

Intercom — Automatic Answer

Automatic Answer Intercom Calls (Auto Answer ICOM) allows a user to answer an intercom call within the intercom group without pressing the intercom button. Auto Answer ICOM works with digital, BRI, and hybrid phones with built-in speaker, headphones, or adjunct speakerphone.

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 can be answered using Automatic Answer, but only attendants can use Automatic Answer to answer external calls directed to the attendant.

Last Number Dialed

Allows 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.

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.

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.

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.

Misoperation Handling

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

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

- If 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.

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.

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.

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

MultiVantage 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.

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.

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.

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.

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.

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.

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.

Ringer Cutoff

Allows the user of a multi-appearance telephone to turn audible ringing signals on and off. Visual alerting is not affected by this feature. When this feature is enabled, only Priority (three-burst) ring, Redirect Notification, Intercom ring, and Manual Signaling ring at the telephone. Internal and external calls do not ring.

Ringing — Abbreviated and Delayed

Allows you to manually or automatically assign one of four ring types to each call appearance on a telephone. Whatever treatment you assign to a call appearance is automatically assigned to each of its bridged call appearances.

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.

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.

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.

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.

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.

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.

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.

Telephone Self Administration

The telephone self administration capability allows you to program feature buttons on the telephone yourself.

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.

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.

Time of Day Routing

Provides the most economical routing of ARS and AAR calls. This routing is based on the time of day and day of the week that each call is made. Up to 8 TOD routing plans may be administered, each scheduled to change up to 6 times a day for each day in the week.

This allows you to take advantage of lower calling rates during specific times of the day and week. In addition, companies with locations in different time zones can use different locations that have lower rates at different times of the day or week. This feature is also used to change patterns during the times an office is closed in order to reduce or eliminate unauthorized calls.

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.

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.

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 may be a safer alternative to this feature.

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.

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.

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.

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 MultiVantage is connected. These services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the System station on an active trunk call. The Trunk Flash feature can help to reduce the number of trunk lines connected to the system. “Digit 1 as Flash” as used in Italy and the United Kingdom will not serve as the flash button in this application.

Index

Numerics

A

B

C

D

E

F

G

H

I

J

L

M

N

O

P

Q

R

S

T

U

V

W

X