Cisco Unified Communications Component Overviews

This chapter provides brief descriptions of the following Cisco Unified Communications system components:

Cisco 1700, 2600, 2800, 3700, and 3800 Series Integrated Access Routers

Cisco 1700, 2600, 2800, 3700, and 3800 series integrated access routers can be deployed as voice gateway routers as part of the Cisco IP Communications solution. Deployments can use these routers as voice gateways with Cisco Unified CallManager.

Cisco 1700 (including Cisco 1751 and 1750 routers), 2600, 2800, 3700, and 3800 series voice gateway routers communicate directly with Cisco Unified CallManager, allowing for the deployment of IP telephony solutions for large enterprises and service providers that offer managed network services. These routers provide a highly flexible and scalable solution for small and medium-sized branches and regional offices.

The Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers support a wide range of packet telephony-based voice interfaces and signaling protocols, providing connectivity support for more than 90 percent of PBX and PSTN connection points. Signaling support includes T1/E1 Primary Rate Interface (PRI), T1 channel associated signaling (CAS), E1-R2, T1/E1 QSIG protocol, T1 Feature Group D (FGD), Basic Rate Interface (BRI), foreign exchange office (FXO), ear and mouth (E&M), and foreign exchange station (FXS). These voice gateway routers can be configured to support from 2 to 540 voice channels.

For additional information, go to:

http://www.cisco.com/en/US/products/hw/routers/index.html

Cisco Emergency Responder

Cisco Emergency Responder enhances emergency calling from Cisco Unified CallManager. It helps assure that Cisco Unified CallManager sends emergency calls to the appropriate Public Safety Answering Point (PSAP) for the caller's location, and that the PSAP can identify the caller's location and, if necessary, return the call. Cisco Emergency Responder can also notify customer security personnel of an emergency call in progress and of a caller's location.

Cisco Emergency Responder helps Cisco Unified CallManager customers comply more effectively with their legal or regulatory obligations and reduce their risk of liability related to emergency calls. It includes these key features:

- Eliminates administration for IP phone relocation
- Alerts customer security personnel to emergency calls in progress
- Automatically tracks IP phone location
- Provides emergency call routing instructions to Cisco Unified CallManager
- Identifies caller location to local exchange carriers and PSAPs
- Supports Emergency callback
- Logs emergency calls and location record changes

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps842/index.html

Cisco FAX Server

The Cisco Fax Server is an easy-to-use, easy-to-manage fax and e-document delivery solution that helps enterprises integrate voice, fax, data, and desktop applications as part of an enterprise IP communications architecture. It enables users to send, receive, and manage documents directly from desktop, e-mail, and other business applications. Based on the Captaris RightFax 9.0 Enterprise Suite, the Cisco Fax Server can be coupled with enterprise messaging applications such as Cisco Unity software to create a powerful unified messaging solution.

For additional information, go to:

http://www.cisco.com/en/US/products/ps6178/index.html

Cisco Multiservice IP-to-IP Gateway

The Cisco Multiservice IP-to-IP Gateway is an integrated application within Cisco IOS software load that runs on the Cisco Integrated Service Routers—Cisco 2800 and Cisco 3800 series for integrated voice, video and data services, and on Cisco 2600XM, Cisco 3700, Cisco 7200VXR and Cisco 7301 series of routing and gateway platforms.

Designed to meet enterprise and service provider Session Border Controller (SBC) needs, the Cisco Multiservice IP-to-IP Gateway facilitates simple and cost-effective connectivity between independent VoIP and video networks. It provides a network-to-network interface point for:

- Signaling interworking (H.323, SIP)
- Media interworking (DTMF, fax, modem and codec transcoding)
- Address and port translations (privacy and topology hiding)
- Billing and CDR Normalization
- QoS and bandwidth management (QoS marking using TOS, DSCP and bandwidth enforcement using RSVP and codec filtering)

The Cisco Multiservice IP-to-IP Gateway integrates SBC needs into the network layer to interconnect VoIP and video networks in intra-enterprise, inter-enterprise, enterprise to service provider, and inter-service provider architectures.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps5640/index.html

Cisco RSVP Agent

Cisco RSVP Agent is a Cisco IOS Software feature that uses the network to deliver call admission control and quality of service for Cisco Unified CallManager deployments. Cisco RSVP Agent enables dynamic adjustment to changes in the network, supports complex network topologies, and enables unified data, voice, and video network designs.

Cisco RSVP Agent employs Resource Reservation Protocol (RSVP), an IETF standards-based signaling protocol for reserving bandwidth in an IP network. Cisco RSVP Agent offer benefits such as:

- Guaranteed WAN bandwidth for Cisco Unified CallManager calls
- Supports for complex network topologies, including meshed designs, redundant links, and dynamically changing topologies

- Unified physical and logical design of a data, voice, and video network, which simplifies deployments and reduces infrastructure and management costs
- Control of the quality and availability of voice and video calls, and authorization of calls
- Support for SIP, H.323, Media Gateway Control Protocol (MGCP), and Skinny Client Control Protocol (SCCP) call-signaling protocols

For additional information, go to:

http://www.cisco.com/en/US/products/ps6832/index.html

Cisco IP Communicator

Cisco IP Communicator provides personal computers with the functionality of IP phones. This Microsoft Windows-based application provides high-quality voice calls to users from wherever they have access to the corporate network. It can serve as a supplemental telephone, a telecommuting device, or a primary desktop telephone.

When registered to Cisco Unified CallManager, Cisco IP Communicator has the functionality of a full-featured Cisco Unified IP Phone, including the ability to transfer calls, forward calls, and conference additional participants to an existing call. In addition, a Cisco IP Communicator that is registered to Cisco Unified CallManager can be provisioned like any other Cisco Unified IP Phone, which greatly simplifies phone management.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps5475/index.html

Cisco Unified CallManager

Cisco Unified CallManager software is the call-processing component of the Cisco Unified Communications system. Cisco Unified CallManager extends enterprise telephony features and capabilities to packet telephony network devices such as IP phones, media processing devices, voice over IP (VoIP) gateways, and multimedia applications. Additional services such as unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems are made possible through Cisco Unified CallManager open telephony APIs. Cisco Unified CallManager offers a suite of integrated voice applications and utilities, including the Cisco Unified CallManager Attendant Console, an ad-hoc conferencing application, the Cisco Unified CallManager Bulk Administration Tool, the Cisco Unified CallManager CDR (call detail record) Analysis and Reporting Tool, the Cisco Unified CallManager Real-Time Monitoring Tool, and the Cisco Unified CallManager Assistant application.

Key features and benefits of Cisco Unified CallManager include:

- An enterprise IP telephony call-processing solution that is scalable, distributable, and highly available. Multiple Cisco Unified CallManager servers are clustered and managed as a single entity on an IP network, which yields scalability of 1 to 30,000 IP phones per cluster, load balancing, and call-processing service redundancy. Interlinking multiple clusters allows system capacity to reach 1 million users in a system of more than 100 sites. Clustering aggregates the power of multiple distributed Cisco Unified CallManager installations, enhancing the accessibility of the servers to phones, gateways and applications, and triple call-processing server redundancy improves overall system availability.

- Call Admission Control (CAC) helps ensure that voice quality of service (QoS) is maintained across constricted WAN links, and it automatically diverts calls to alternate public switched telephone network (PSTN) routes when WAN bandwidth is not available. A web interface to the configuration database enables remote device and system configuration. HTML-based online help is available for users and administrators.
- The appliance model provides a platform for call processing with the software preloaded on a Cisco Media Convergence Server (MCS) platform. The appliance comes with a single firmware image that includes the underlying operating system and the Cisco Unified CallManager application, and is accessed through a GUI. A command-line interface enables diagnostics and basic system management. All systems management activities, such as disk space monitoring, system monitoring, and upgrades, are either automated or are controlled through the GUI. To further enhance security, Cisco Security Agent for Unified CallManager is preloaded on the appliance. The appliance also includes a host-based firewall has been added and IP Security (IPSec) connectivity between all cluster members.
- Session Initiation Protocol (SIP) provides support of line-side devices, including IETF RFC 3261 compliant devices. Cisco SIP-compliant devices include many Cisco Unified IP phone models.
- The SIP trunk interface conforms to RFC 3261, which allows of video calls over the SIP trunk and enhanced conferencing and application support when used with Cisco Unity and Cisco Unified MeetingPlace.
- Support for RSVP (Reservation Protocol) agent capability. The Cisco RSVP agent on a Cisco router extends CAC capability beyond a hub-and-spoke topology within a cluster and allows a call can be routed directly between two locations without having to traverse the hub.
- SNMP support, which allows managers to set and report traps on conditions that could affect service and send them to the remote monitoring systems.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/index.html

Cisco Unified CallManager Express

Cisco Unified CallManager Express is an entry-level call processing system that provides a wide range of IP telephony features for small to medium-sized businesses and autonomous small enterprise branch offices with up to 240 phones.

All files and configurations for IP phones are stored internally on a single Cisco Integrated Services router for a cost-effective, highly reliable, IP communications solution. Cisco Unified CallManager Express helps ensure investment protection and offers scalability because all hardware and software is fully compatible with Cisco Unified CallManager and Cisco Unified Survivable Remote Site Telephony.

Cisco Unified CallManager Express provides key system and PBX modes of operation on a single network and several industry-unique features, including:

- Call processing for local IP and analog phones attached to a Cisco router
- Support for analog phones in SCCP mode, Session Initiation Protocol (SIP) line side support with supported Cisco Unified IP phones, and a robust set of PSTN interfaces
- Call routing over a WAN with calling party name and number information, and compressed voice for reduced WAN bandwidth utilization
- Support for peripheral services such as voice mail, automated attendant, and IP-based XML and Telephony Application Programming Interface (TAPI) applications

- Interoperability with Cisco Unified CallManger and the Cisco Unity Express
- Simple software configuration change on the Cisco router converts system to a highly available survivable telephony gateway for a remote site in a centralized Cisco Unified CallManager deployment

System management features in the Cisco Unified CallManager Express environment enable you to:

- Accomplish initial installation of Cisco Unified CallManager Express easily using a setup tool that prompts for answers to pertinent
- Perform everyday administration and remote troubleshooting using the Cisco IOS software command-line interface (CLI)
- Add users, phones, and extensions or make changes for system and integrated voice-mail using a single web-based GUI designed for nontechnical staff

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps4625/index.html

Cisco Unified Contact Center Enterprise

Cisco Unified Contact Center Enterprise provides a full-featured distributed contact center infrastructure, which segments customers, monitors resource availability, and delivers each contact to the most appropriate resource. It provides a VoIP contact center solution that integrates inbound and outbound voice applications with Internet applications, including real-time chat, web collaboration and e-mail.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/custcosw/ps1844/index.html

Cisco Unified Contact Center Express

Cisco Unified Contact Center Express provides departmental, enterprise branch, or small to medium-sized companies with easy-to-deploy, easy-to-use, and sophisticated customer interaction management for up to 300 agents. These applications securely support a virtual contact center with integrated self-service applications across numerous sites. They provide support for powerful agent-based assisted service and fully integrated self-service applications and offer distributed automatic call distributor (ACD), interactive voice response (IVR), computer telephony integration (CTI), and agent and desktop services.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/custcosw/ps1846/index.html

Cisco Unified Customer Voice Portal

The Customer Unified Voice Portal provides call-management and call-treatment solutions with self-service IVR capabilities, allowing callers to obtain personalized answers to complex questions and to conduct business without interacting with a live agent.

The Cisco Unified Customer Voice Portal includes support for agent queuing and for multisite call switching capabilities. It uses standard Internet technologies to provide a smooth customer experience even when transferring calls between several locations. With support for the Cisco Unified Intelligent Contact Management and Cisco Unified Contact Center products, the Cisco Unified Customer Voice Portal delivers self-service as part of a comprehensive customer contact strategy that provides unique, personalized interactions.

The Cisco Customer Voice Portal supports speech-enabled and touch-tone applications, which can be quickly integrated with back-end data and business rules that are available on the web. Using the standard Java 2 Platform, Enterprise Edition (J2EE) and Voice Extensible Markup Language (VoiceXML) with the graphical development tools provided with the portal (which are compliant with the Eclipse standard for building web applications), you can develop complex voice applications quickly and cost-effectively.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/custcosw/ps1006/index.html

Cisco Unified IP Phones

Cisco Unified IP Phones are full-featured telephones that provide voice communication over an IP network. They function much like digital business phones, allowing you to place and receive phone calls and to access features such as mute, hold, transfer, speed dial, call forward, and more. In addition, because Cisco IP Phones are connected to your data network, they offer enhanced IP telephony features, including access to network information and services, and customizeable features and services. Many phone models also support security features that include file authentication, device authentication, signaling encryption, and media encryption.

The Cisco Unified Communications system supports these Cisco Unified IP Phone models:

- Cisco Unified IP Phone 7902G—Suitable for public spaces, lobbies, workshops, and warehouses where the phone is not assigned to any one user. Includes one line (DN), four fixed feature keys, and no display. Offers basic security features and several power options.
- Cisco Unified IP Phones 7905G / 7906G—Suitable for a user who conducts low to moderate telephone traffic and does not have a PC connected to the phone. Includes one line (DN), four dynamic softkeys, a pixel-based display, and support for Extensible Markup Language (XML) services. Offers security features and several power options, including IEEE 802.3af power on the 7906G.
- Cisco Unified IP Phones 7911G / 7912G—Suitable for a user who conducts low to moderate telephone traffic and has a PC connected to the phone. Includes one line (DN), Ethernet switch and PC port, four dynamic softkeys, a pixel-based display, and support for XML services. Offers security features and several power options, including IEEE 802.3af power on the 7911G.
- Cisco Unified IP Phone 7940G / 7960G—Suitable for the needs of a transaction-type worker. Provides two lines (DNs) or a combination of lines and direct access to telephony features on the 7940G, and six lines or combination of lines and direct access to telephony features on the 7960G. Also includes a large LCD display, programmable line and feature keys, dynamic softkeys, high-quality speakerphone, built-in headset port, integrated Ethernet switch, and audio controls for full-duplex speakerphone, handset, and headset. Supports XML services, security features, and power options.
- Cisco Unified IP Phone 7941G / 7941G-GE / 7961G / 7961G-GE—Provides similar capabilities to the Cisco Unified IP Phone 7940G / 7960G, with the addition of a higher-resolution grayscale pixel-based LCD and IEEE 802.3af power. Includes two lines (DNs) or combination of lines and

direct access to telephony features on the 7941G / 7941G-GE, and six lines or combination of lines and direct access to telephony features on the 7961G / 61G-GE. GE models include a gigabit Ethernet port for integration with PCs or desktop servers.

- Cisco Unified IP Phone 7970G / 7971G-GE—Suitable for managers and executives. Includes a backlit, high-resolution color touch-screen display. Supports up to eight telephone lines or combination of lines and direct access to telephony features and provides hands-free speakerphone, built-in headset connection, and many telephony features. Offers security features and several power options, including IEEE 802.3af power. GE model includes a gigabit Ethernet port.
- Cisco Unified IP Phone 7985G—A personal desktop video phone for the Cisco Unified IP Communications solution. Provides all components needed for video calls, including camera, LCD screen, speaker, keypad, and handset. Includes one line (DN) and a backlit, high-resolution color video display. Supports hands-free speakerphone, high-quality voice codecs, built-in headset connection, and many telephony features. Offers security features and several power options including IEEE 802.3af power.
- Cisco Unified Wireless IP Phone 7920G (IEEE 802.11b wireless IP phone)—Supports up to six lines, security features, XML services, and many telephony features.
- Cisco Unified IP Conference Station 7936G—Suitable for a 20-foot by 30-foot room with 360 degree coverage. Includes one line (DN), a backlit LCD display, and two ports for optional extension microphones.

For additional information, go to:

http://www.cisco.com/en/US/products/hw/phones/ps379/index.html

Cisco Unified MeetingPlace

Cisco Unified MeetingPlace is a complete rich-media conferencing solution that integrates voice, video, and web collaboration capabilities. It allows users from any location to meet at any time and to easily integrating web, voice, and video conferencing into everyday communications.

Cisco Unified MeetingPlace provides intuitive interfaces for setting up, attending, and managing meetings. It allows immediate or future voice, video, and web conferences to be set up and attended in a single step—from Cisco Unified IP Phones, instant messaging clients, web browsers, and Microsoft Outlook and Lotus Notes calendars. Meeting participants have complete control over voice, video, and web conferences from a single browser interface.

Cisco Unified MeetingPlace can be deployed “on network,” behind a firewall, and integrated directly into an organization’s private voice and data networks and collaborative applications. This deployment enables cost savings because organizations can use their IP network infrastructures to reduce transport costs paid to service providers. In addition, on-network deployment results in a secure meeting environment by allowing organizations to isolate confidential meetings and content behind the firewall while providing the flexibility to meet with external parties. To prevent unauthorized access and toll fraud, Cisco Unified MeetingPlace integrates with the corporate directory to provide synchronized updates as an employee’s status changes.

Cisco MeetingPlace can be located in on-premises or hosted in off-site facilities. It can be managed in-house or management can be outsourced.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/ps5664/ps5669/index.html

Cisco Unified MeetingPlace Express

Cisco Unified MeetingPlace Express is an integrated voice and web conferencing solution for medium-sized organizations. It allows users to communicate and collaborate from any place at any time, using only a phone and a web browser. Cisco MeetingPlace Express integrates meeting management and control capabilities directly into web and Cisco Unified IP Phone interfaces.

Cisco Unified MeetingPlace Express supports industry standard protocols to ensure connectivity with a range of telephony systems, including Cisco Unified CallManager and Cisco Unified CallManager Express.

For additional information, go to:

http://www.cisco.com/en/US/products/ps6533/index.html

Cisco Unified Mobility Manager

Cisco Unified MobilityManager makes Cisco Mobile Connect services available to Cisco Unified CallManager users who want to consolidate all their business calls with a single enterprise IP phone number. The Cisco Mobile Connect service helps mobile workers direct their inbound business calls to their IP phone number and initiate outbound business calls as if they were at their IP phone—all from a mobile phone or other remote phone destination. To support Cisco Mobile Connect, Cisco Unified MobilityManager software includes an integrated suite of mobility application services, including web-based system administration and user profile configuration utilities to create, access, and control user profile information for each enterprise mobile worker.

Cisco Unified MobilityManager includes these features:

- Simultaneous desktop ringing
- Desktop pickup
- Mobile call pickup
- Security and privacy for Cisco Mobile Connect calls
- Cisco Mobile Voice Access
- Single enterprise voice mailbox
- Allowed and blocked call filters
- Caller identification
- System administrator-controllable user profile access
- Remote on/off control
- Voice-based access with user identification and personal identification number protection
- Call tracing

For additional information, go to:

http://www.cisco.com/en/US/products/ps6567/index.html

Cisco Unified Personal Communicator

Cisco Unified Personal Communicator transparently integrates a wide array of communications applications and services into a single desktop computer application. It provides access to a variety of communications tools, including voice, video, web conferencing, call management, directories, and presence information. Cisco Unified Personal Communicator offers an easy-to-use interface that streamlines the communications experience and facilitates collaboration. With Cisco Unified Personal Communicator, users can communicate virtually anytime, from anywhere, and can easily escalate communication methods as required.

For additional information, go to:

www.cisco.com/go/unifiedpersonalcomm

Cisco Unified Presence Server

Cisco Unified Presence Server enables the deployment of Session Initiation Protocol (SIP) technology to support new voice services in an enterprise environment. SIP enhances the voice network by providing a core set of behaviors for session establishment and control that can be applied in a wide array of features and services. In addition to core SIP support, Cisco Unified Presence Server uses SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions) technology to support instant messaging (IM) and presence.

Cisco Unified Presence Server consists of a SIP presence engine and a SIP proxy function. The presence engine collects user presence information (such as busy, idle, away, or available status) and user capabilities (such as the ability to support voice, video, instant messaging, and web collaboration), and compiles the data in a repository for each user. This repository is accessed by the applications and features that each user employs. A user can apply unique user rules and privacy to ensure that only authorized applications and users have access to presence information. The SIP proxy function allows for efficient and accurate routing of presence and general SIP messaging through the enterprise.

Cisco Unified Presence Server integrates with various desktop clients and applications. It enables Cisco Unified Personal Communicator to perform functions such as click-to-dial and phone control as well as voice, video, and web collaboration. In addition, Cisco Unified Presence Server provides a core IM service for Cisco Unified IP Phones that are connected to Cisco Unified CallManager. Cisco Unified Presence Server also supports interoperability with Microsoft Live Communications Server (LCS) 2005 and the Microsoft Office Communicator client, enabling specific Office Communicator functions to work with Cisco Unified IP Phones supported on Cisco Unified CallManager. Finally, the SIP and SIMPLE interface on the Cisco Unified Presence Server can provide value add presence and call control capabilities to any SIP/ SIMPLE standards based application or service.

For additional information, go to:

http://www.cisco.com/en/US/products/ps6837/index.html

Cisco Unified Survivable Remote Site Telephony

Cisco Unified CallManager with Cisco Unified Survivable Remote Site Telephony (SRST) allows companies to extend high-availability IP telephony to their remote branch offices with a cost-effective solution that is easy to deploy, administer, and maintain. The SRST capability is embedded in the Cisco IOS Software that runs on the Cisco 1700, 2600, 2800, 3700, and 3800 integrated services routers.

SRST software automatically detects a connectivity failure between Cisco Unified CallManager and IP phones at a branch office. SRST initiates a process to automatically configure the Cisco 1700, 2600, 2800, 3700, and 3800 series integrated services routers to provide call-processing backup redundancy for the IP phones and PSTN access in the affected office. The router provides essential call-processing services for the duration of the failure, helping ensure that critical phone capabilities are operational. Upon restoration of the connectivity to the Cisco Unified CallManager, the system automatically shifts call-processing functions back to the primary Cisco Unified CallManager cluster.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps2169/index.html

Cisco Unified Video Advantage

Cisco Unified Video Advantage brings video telephony functionality to the Cisco Unified IP Phone 7900 Series and to Cisco IP Communicator. It is composed of Cisco Unified Video Advantage software and Cisco VT Camera II, a video telephony USB camera. System administrators provision a Cisco Unified IP Phone with Cisco Unified Video Advantage just as they would provision a phone for audio calls. Users make and receive calls on their Cisco Unified IP Phones using the familiar phone interface, and calls display with video on user PCs without additional user action required.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps5662/index.html

Cisco Unified Videoconferencing

Cisco Unified Videoconferencing provides organizations with a reliable, easy-to-manage, versatile, and cost-effective network infrastructure for videoconferencing applications. In addition to integrating legacy and IP-based room systems over a single infrastructure, Cisco Unified Videoconferencing solutions offer video-enable telephony endpoints and rich media applications, enabling participants to collaborate and share information in real time. Cisco Unified Videoconferencing offers simple dialing options, a range of dynamic layouts, and many in-conference controls. Cisco Unified Videoconferencing provides support for H.323, H.320, SIP and SCCP video endpoints with a variety of formats, speeds and functionality

The Cisco Unified Videoconferencing product family is composed of the Cisco Unified Videoconferencing MCU 3511, 3521, 3526, and 3540. These products work with Cisco IOS gatekeepers and gateways.

For additional information, go to:

http://www.cisco.com/en/US/products/hw/video/ps1870/index.html

Cisco Unity

Cisco Unity is a messaging platform designed for enterprises of all sizes. It provides unified messaging (e-mail, voice, and fax messages sent to one inbox) and full-featured voice mail. Cisco Unity interoperates with most legacy TDM PBXs and with Cisco Unified CallManager to enable a transition to IP telephony while protecting existing infrastructure investments.

Key features of Cisco Unity include:

- Integration with Outlook or Lotus Notes desktop clients.
- Telephone interface (TUI) for DTMF-based control of messages. An intuitive interface allows accessing, creating, replying to, and forwarding messages using a traditional telephone, and allows managing and customizing mailbox features.
- Web-based desktop interface that allows users to manage and customize their mailbox features and to access their voice messages directly from a PC.
- Text-to-speech (TTS) for telephone access to e-mail messages.
- Integration with Exchange or Lotus Domino to provide a single location to store and manage all of messages.
- Unity Digital Networking using integration into a common Active Directory or Lotus Domino Directory to provide seamless message exchange between users at several sites on different Cisco Unity servers.
- Mobile message access for Unified Messaging subscribers using Blackberry or Treo devices.
- Cisco FAX server support or integration with third-party FAX vendors to provide FAX messages in a single, unified inbox.
- Interoperability with a wide range of legacy TDM PBX systems using analog DTMF, serial SMDI, or digital set emulation.
- Interoperability with a wide range of legacy voice messaging system using AMIS, VPIM, or Cisco Unity Bridge (for Octel node emulation).

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps2237/index.html

Cisco Unity Connection

Cisco Unity Connection provides messaging capabilities for mid-size offices and small enterprises. It includes an intuitive telephone interface, voice-enabled navigation of messages, and desktop access to messages directly from a PC. Cisco Untiy Connection integrates with Cisco Unified CallManger, Cisco Unified CallManger Express, and various legacy PBX models (using the PIMG) to support a variety of deployment models and configurations.

Key features of Cisco Unity Connection include:

- Voice-enabled message navigation (such as play, delete, reply, forward)
- Voice-enabled dialing to other system users
- Desktop messaging with the Unity Inbox web client
- Desktop messaging with IMAP-based e-mail clients
- Personal call transfer rules, which allow call routing based on caller, time of day, Outlook calendar status, and other parameters
- Text-to-speech (TTS), which allows access to Exchange e-mails from a telephone
- Message notifications to pagers, SMS phones, and other devices
- Automated attendant capabilities

For additional information, go to:

http://www.cisco.com/en/US/products/ps6509/index.html

Cisco Unity Express

Cisco Unity Express provides integrated, entry-level, voice mail and automated attendant services for small and medium offices or branches in Cisco Unified CallManager or Cisco Unified CallManager Express environments. In Cisco Unified CallManager environments, Cisco Unity Express provides local storage and processing of voice mail and automated attendant services, alleviating WAN bandwidth and QOS concerns for the branch office. Combining Cisco Unified CallManager Express with Cisco Unity Express provides a core set of phone features for everyday business needs while offering a variety of telephony feature sets that have been provided by traditional key systems and hybrid PBXs.

Cisco Unity Express voice messaging and auto-attendant includes the following key features:

- Networking across several sites—Voice Profile for Internet Mail version 2 (VPIMv2) provides support for voice mail messaging interoperability between Cisco Unity Express sites and between Cisco Unity Express and Cisco Unity, with Non-Delivery Record (NDR) for networked messages and blind addressing
- Distribution lists—Public and private distribution lists of local and remote users can be created for sending messages to more than one subscriber
- Broadcast messages—Privileged subscribers can send messages to all users on the network
- Password and PIN length flexibility—Network administrators can set minimum lengths and expiry times for passwords and personal identification numbers (PINs) for greater network security
- SNMP MIB support—Network administrators can remotely monitor the health and performance of the Cisco Unity Express system.
- Support for caller ID information in incoming messages—Permits playing of caller identification information as part of the message envelope for new incoming voice mail messages
- Addition of remote users to the local directory—The voice-mail administrator can add frequently called remote users to the local directory, which permits local users to address voice mail messages to remote users using dial-by-name and to receive spoken name verification of the remote user address
- Undelete voice messages—Voice-mail users can restore a voice-mail message that was deleted during the current voice message retrieval session.
- Audio prompts in a variety of languages.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps5520/index.html

Cisco VG224

The Cisco VG224 Analog Phone Gateway combines a high-density RJ21 analog interface with Cisco IOS Software manageability to provide a cost-effective platform for maximum functionality of existing analog phone equipment. It offers the following key benefits:

- High-density 24-port gateway for analog phones, fax machines, modems, and speakerphones
- DSP technology for fax and modem support
- Enhances an enterprise voice system architecture that is based on Cisco Unified CallManager or Cisco Unified CallManager Express
- Compact, 19-inch rack-mount chassis

For additional information, go to:

http://www.cisco.com/en/US/products/hw/gatecont/ps2250/ps5627/index.html