



Cisco Unified Communications System Description

Release 6.1(1)

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Preface

Overview

This document provides an overview of the Cisco Unified Communications 6.1(1) system. It describes the Cisco Unified Communications system-level approach, lists key features of the Cisco Unified Communications components, and illustrates the various Cisco Unified Communications deployment models. This document also provides an overview of the steps you follow when you deploy a Cisco Unified Communications solution.

Organization

This manual is organized as follows:

Chapter 1, “Cisco Unified Communications System Overview”	Provides an overview of the Cisco Unified Communications system-level approach and architecture
Chapter 2, “Deployment Models”	Introduces various options for deploying the Cisco Unified Communications solution
Chapter 3, “Cisco Unified Communications Component Overviews”	Provides a general overview of the features provided by the Cisco Unified Communications solution and links to additional information
Chapter 4, “Component Protocols and APIs”	Lists the protocols and application program interfaces (APIs) that are supported by various Cisco Unified Communications solution components
Chapter 5, “Deployment Methodology”	Provides an overview of the steps that are involved in implementing a Cisco Unified Communications solution
Appendix A, “Cisco Unified Communications Architecture Basics”	Provides an overview of the technologies related to voice and video over an IP network

Related Documentation

The Cisco Unified Communications solution provides a suite of interactive documentation that covers details of the system architecture and components, installation and upgrade information, troubleshooting, related information. You can access this documentation at this URL:

<http://www.cisco.com/go/unified-techinfo>

Obtaining Documentation and Submitting a Service Request

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

<http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html>

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CHAPTER 1

Cisco Unified Communications System Overview

The Cisco Unified Communications 6.1(1) system securely integrates voice, video, and other collaborative data applications into intelligent network communications solutions. This system, which includes IP telephony, unified communications, rich-media conferencing, IP video broadcasting, and customer contact solutions, takes full advantage of the power, resiliency, and flexibility of an IP network. The elements of this system were designed, developed, documented, and tested as part of a comprehensive, end-to-end Unified Communications System.

The Cisco Unified Communications system reduces the cost and complexity associated with managing multiple and remote sites, meets stringent quality of service (QoS) requirements, and provides optimal availability and security when deployed as part of a converged network. In addition, the solution interoperates with existing time-division multiplexing (TDM)-based systems and enterprise business applications, allowing organizations to migrate to full-featured IP Communications while maintaining existing technology investments.

This topic provides an overview of the key features and benefits of Cisco Unified Communications. It includes these sections:

- [System Definition](#)
- [System Release Strategy](#)
- [Service Offerings](#)
- [Career Certifications](#)
- [Solution Bundling](#)
- [Intelligent Information Network](#)
- [Business Productivity Applications](#)
- [Customer Interaction Network](#)
- [IP Communications](#)
- [Security](#)
- [Deployment and Migration](#)

System Definition

The Cisco Unified Communications system is designed for a single, secure, converged network. Part of an integrated, comprehensive Cisco architecture, the communications applications reside “in” the network, not “on” the network, and can easily incorporate emerging business processes, applications, and new devices. Applications can be deployed in a single instance, rather than in multiple instances, and managed services offerings further increase deployment flexibility. Standards-based Cisco Unified Communications products let organizations migrate based on business needs, not technical limitations, to keep pace with new technology.

This release of the Cisco Unified Communications system introduces the Cisco Unified Communications Manager, Business Edition and deployment models for the medium-sized business. The Cisco Unified Communications Manager, Business Edition includes the features and capabilities of Cisco Unified Communications Manager, Cisco Unified Mobility, and Cisco Unity Connection co-resident on a single, low-cost Media Convergence Server. The Cisco Unified Communications Manager, Business Edition is designed to support 150 to 500 endpoints in one main and up to five remote locations.

The Cisco Unified Communications System also includes a suite of network management applications that allow you to monitor, manage, and troubleshoot your system. It also includes tools that allow you to analyze the readiness of your infrastructure to support the UC system.

System Release Strategy

The Cisco Unified Communications system includes the following types of releases:

- Major release—Marks the beginning of a major new release version. This release type typically is based on a major release of at least one of these components: Cisco Unified Communications Manager, Cisco Unity, Cisco Unified MeetingPlace, or Cisco Customer Response Solutions.
- Minor release—Adds features and fixes to an existing major release. This release type can consist of revisions to existing components and new versions of components.
- Maintenance release—Contains bug fixes for one or more of the components. This release type is based on an existing major or minor release.

For example, in Cisco Unified Communications release 6.1(1), “6” indicates the major release, the first “1” indicates the minor release, and the second “1” indicates the maintenance release.

Service Offerings

Using the Cisco Lifecycle Services approach, Cisco Systems and its partners offer a broad portfolio of end-to-end services. These services are based on proven methodologies for deploying, operating, and optimizing Unified Communications solutions. Planning and design services, for example, can help you meet aggressive deployment schedules and minimize network disruption during implementation. Operate services reduce the risk of communications downtime with expert technical support. Optimize services enhance solution performance for operational excellence. Cisco and its partners offer a system-level service and support approach that can help you create and maintain a resilient, converged network that meets your business needs.

Service offerings include:

- Cisco Unified Communications Software Subscription, which allows you to purchase major software version upgrades of various Cisco Unified Communications products at a reduced cost through a one-, two-, or three-year subscription.
- Cisco Unified Communications Essential Operate Service, which provides 24-hour, 365-day-a-year access to Cisco Systems engineers and certified partners who are highly trained and have a deep understanding of Cisco Unified Communications products and technologies.
- Cisco Unified Communications Select Operate Service, which provides a proactive support solution that combines 24-hour, 365-day-a-year access to technical support representatives plus a simple-to-install monitoring solution designed for Cisco Unified Communications.
- Cisco Unified Communications SMB Network Operate & Optimize Service, is a partner-led service offering (designed specifically for the medium-sized businesses) that enables the delivery of affordable, ongoing, high-availability network support.

Career Certifications

The Cisco Certified Voice Professional (CCVP) certification and related certifications are designed for IT professionals who are responsible for integrating voice technology into underlying network architectures. Individuals who earn a CCVP certification can help create a telephony solution that is transparent, scalable, and manageable. Earning a CCVP certification validates a robust set of skills in implementing, operating, configuring, and troubleshooting a converged IP network. The certification content focuses on many components of the Cisco Unified Communications system, including Cisco Unified Communications Manager, quality of service (QoS), gateways, gatekeepers, IP phones, voice applications, and utilities on Cisco routers and Cisco Catalyst switches.

Solution Bundling

In addition to providing traditional solution ordering, where you choose the individual components and quantities that you require, the Cisco Unified Communications system provides flexible bundling options. A bundled solution simplifies the way in which you order applications and services and makes it easy to add options.

This release of the Cisco Unified Communications system introduces a new bundling option for medium-sized businesses. This option includes the Cisco Unified Communications Manager, Business Edition, which is designed specifically to address the call processing and messaging needs of medium-sized businesses, as well as new network management tools.

Intelligent Information Network

The Cisco Intelligent Information Network facilitates the evolution of networking to systems. It allows the network to be used as a strategic asset and provides capabilities that include:

- Cisco Discovery Protocol (CDP)—A simple broadcast protocol that devices use to advertise their presence, it operates in the background and facilitates communication between a Cisco Unified IP Phone plugged into a network and the network switch.

- **QoS**—Cisco provides an end-to-end solution to ensure quality of service. QoS starts at the phone and LAN distribution layer, where packets are classified and marked as high priority traffic. Traffic markings originating from Cisco Unified IP Phones are automatically trusted by the Cisco switch infrastructure, which typically remarks traffic from nontrusted end user workstations. Configuration is made easier through Cisco AutoQoS, which automatically handles a range of tasks traditionally done manually, including classifying applications, generating policies, configuring the proper QoS configurations, monitoring and reporting to test QoS effectiveness, and enforcing service-level consistency.

As traffic flows through the access layer, priority queuing and buffer management ensure that real-time traffic is prioritized over less time-critical data. Where bandwidth is most restricted, across the WAN, the Cisco solution provides RSVP for reserving the bandwidth needed for voice. Fragmentation and interleaving of large blocks of data ensure a steady stream of voice traffic, and voice packet header compression minimizes bandwidth consumed.

- **VLAN**—When a Cisco Unified IP Phone boots up on the IP network, it advertises its presence using CDP, and it requests an IP address lease from a DHCP server. The Cisco LAN switch learns of the new phones via CDP and automatically reconfigures to add that port to the VLAN used for voice. With this feature, the LAN infrastructure can distinguish a phone from a PC and does not require manual configuration every time a phone is added, moved, or removed.
- **Wireless**—Cisco wireless access points allow Cisco wireless phone users to roam a campus without losing voice connectivity. If a user roams to a different site, the system will discover the new physical location for emergency 911 information purposes.
- **Power over Ethernet (POE)**—Eliminates the need for local power connections for every phone. Cisco switches can be configured with redundant power supplies connected to uninterruptible power supplies in a data center to ensure that the power to the phone is preserved, even when local power for other equipment at the desk is lost. Most Cisco Unified IP Phone models support the industry-standard 802.3af power and the Cisco pre-standard inline power.
- **Gigabit Ethernet (GigE)**—Allows certain Cisco Unified IP Phone models to take advantage of the emerging Gigabit Ethernet LAN infrastructure.

Business Productivity Applications

The Cisco Unified Communications system provides a wide array of applications that enhance business and organizational productivity and efficiency. These applications offer capabilities that include:

- **Rich-media conferencing**—Cisco Unified MeetingPlace provides intuitive interfaces for setting up, attending, and managing meetings. Extensive voice, video using Cisco Unified Videoconferencing, and web conferencing capabilities enable a range of meeting applications, including highly-collaborative meetings, training sessions, and presentations.
- **Messaging**—Cisco Unity provides users with access to voice, e-mail, and fax messages from a Cisco Unified IP Phone or from a PC. These solutions combine unified messaging with personal productivity tools to help manage communications quickly and conveniently. For midsize organizations, Cisco Unified Connection provides voice messaging, speech recognition, call routing rules, and desktop PC message access in a system that is easy to manage and deploy. For small organizations, Cisco Unity Express offers a voice messaging solution that integrates with your router.

- Common interface—Cisco Unified Personal Communicator is a presence-based desktop application that provides a focal point for phone services, directory services, messaging, and conferencing.
- Cisco Unified Presence—The focal point of all status processing, including attributes and capabilities. It links the various knowledge within each application to provide a ubiquitous and broad view of a defined user within the Cisco Unified Communications system.

Customer Interaction Network

The Cisco Customer Interaction Network component provides a single, integrated platform for all contact center locations. It is a distributed, IP-based customer-service infrastructure that easily integrates with legacy contact center platforms and networks, providing multi-channel services and integration with customer relationship management applications.

- Intelligent contact routing and multi-channel automatic call distribution (ACD)—Enables interaction with customers via phone (inbound or outbound), web, e-mail or chat. The application provides call handling tailored to different classes of customers and to individual customers, providing flexible contact center operational profiles based on varying business needs.
- Voice and web self-service—Extracts and parses web content and presents this data to customers through a telephony interface, allowing simple transactional requests to be handled by the interactive voice response (IVR) system instead of by agents. This application provides self-service automation with automatic speech recognition (ASR) and TTS. It also performs *prompt-and-collect* functions to obtain user data such as passwords or account identification that it can then pass to contact center agents, and it delivers proactive notification users through e-mail, fax, pager, and short message service (SMS).
- Agent and supervisor options—Provide full support for agent or supervisor interaction using chat capabilities. Instant messaging offers the capability to communicate with any or all the agents on a supervisor's team. Other options include:
 - Agent status monitoring
 - Silent monitoring
 - Barge-in
 - Intercept
 - Real-time and historical reporting
 - ACD

IP Communications

IP communications provides powerful and efficient voice, data, and video communications, and related capabilities. Key features include:

- Video telephony—Allows video calls to be placed and received over an IP telephony network using the familiar phone interface. Video endpoints support common call features such as forward, transfer, conference, and hold. Use of a single infrastructure also enables a unified dial plan and user directory for voice and video calls. This release of the Cisco Unified Communications system also includes Cisco Unified Conferencing for TelePresence, which is a new technology that combines rich audio, high-definition video, and interactive elements to deliver a unique in-person experience.

- Mobility—Provides for several forms of user mobility, including:
 - Extension Mobility—Allows users to access any phone within a single Cisco Unified Communications cluster as their own, by simply logging in to the phone. After log in, the phone assumes all of the user profile information, including line numbers, speed dials and service links.
 - Site/campus mobility—Allows users to access the Cisco Unified Communications network through the wireless Cisco Unified Wireless IP Phones 7920G and 7921G. In addition, this release includes enhanced mobile IP phone applications that allow users to:
 - Dynamically manage how and when mobile calls take place
 - Intelligently screen calls based on urgency, subject matter, and caller identity
 - Identify which users are available to talk and which users choose not to be disturbed
 - Increase accessibility of corporate calendar and contact information from mobile phones.
- Emergency caller response/safety and security—Enables emergency calls in an IP network to be directed to the appropriate Public Safety Answering Point (PSAP). In this way, emergency agencies can identify the location of 911 callers without a system administrator needing to keep location information current.

Security

The Cisco Unified Communications system takes a layered approach to protecting against various attacks, including denial of service (DOS), privacy, and toll fraud. Security features include:

- Encryption of signaling and media—Ensures that the signaling and the actual phone conversations are protected against unintended interception by third parties.
- Catalyst Integrated Security Features (CISF)—Includes private VLANs, port security, DHCP snooping, IPSource Guard, secure Address Resolution Protocol (ARP) detection, and dynamic ARP inspection. These features protect the network against attacks such as man-in-the-middle attacks and other spoofing.
- Integration with firewalls—Ensures that system platforms are accessible only by authorized devices. The firewall acts as a guardian between all IP devices and the Cisco Unified Communications system platforms, ensuring that only specific transactions are allowed.
- Secure platforms—Provides features, such as host-based intrusion detection, optional security scripts, and anti-virus software, that ensure that the platform is hardened against intruders and malicious code.
- Enhanced phone security features—Provides configurable levels of security. Options include configuring the phone to ignore Gratuitous Address Resolution Protocol (GARP) requests, disabling the PC port on the phone, disabling access to network configuration settings on a phone, and configuring a phone to accept only digitally signed firmware images.

Deployment and Migration

The Cisco Unified Communications system is designed to be deployed efficiently and effectively. The solution offers:

- Flexible deployment models—Cisco Unified Communications supports LAN and WAN connectivity and can be configured for single-site or multi-site networks. Headquarters, contact centers, branch offices, and telecommuter configurations can be interconnected without geographic constraints. Call processing and administration can be centralized or distributed.
- Integration with existing equipment and networks—Cisco Unified Communications provides gateway support to enable integration and interoperability with existing call processing equipment, phones, and TDM networks. This capability ensures compatibility with and migration from legacy systems, and supports:
 - Integration with PBXs through QSIG, Digital Private Network Signaling System (DPNSS), and PRI links
 - Integration with ACD platforms via CTI interface
 - Integration with legacy phones through gateways
 - Integration with TDM networks through gateways via T1, E1, and PRI links
- Open IP connectivity through SIP—Cisco Unified Communications provides enhanced support for SIP trunking and to a variety of SIP endpoints. An integrated Cisco Unified Presence provides user information and status and enables interconnection to popular messaging networks.
- High availability—Cisco Unified Communications networks can be built to meet high availability requirements as business needs dictate. Networks can be designed to ensure no single point of failure in either network topology or applications. Cisco Unified Survivable Remote Site Telephony (Unified SRST) allows remote branch offices to remain in service even when the WAN access link is lost.



CHAPTER 2

Deployment Models

This chapter provides an overview of the Cisco Unified Communications deployment models that Cisco has tested and verified. These models are not the only ways in which you can deploy the Cisco Unified Communications system, nor are they design recommendations. Rather, they are designed to provide sample configurations that address typical system-level requirements.

For additional guidelines, recommendations, and best practices for implementing enterprise networking solutions, refer to the *Cisco Solution Reference Network Design (SRND)* guides and related documents, which are available at this URL:

www.cisco.com/go/srnd

For additional information about the deployment models, including details about all components in each model, refer to the Cisco *Unified Communications System Technical Information* site at: <http://www.cisco.com/go/unified-techinfo>.

This chapter includes these sections:

- [Deployment Overview, page 2-1](#)
- [Single-Site Model, page 2-2](#)
- [Multisite Centralized Call Processing Model, page 2-4](#)
- [Multisite Distributed Call Processing Model, page 2-6](#)
- [Clustering Over IP WAN Call Processing Model, page 2-8](#)
- [Major Components of Deployment Models, page 2-10](#)

Deployment Overview

The sample Cisco Unified Communications deployments demonstrate a variety of business applications based on the following criteria:

- End-to-end IP communications requirement
- Interoperability between sites
- Administrative requirements (centralized or distributed)
- Messaging requirements
- Conferencing requirements
- Availability requirements
- Mobility requirements

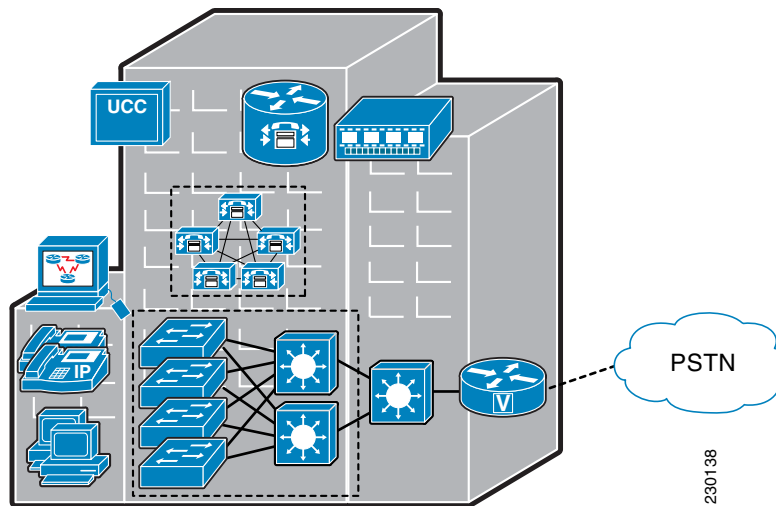
- Scalability requirements
- Customer interaction network requirements

Single-Site Model

The Single-Site model is designed for autonomous offices in which most or all employees are IPC users. This model supports up to 30,000 users.

Figure 2-1 shows an example of this model.

Figure 2-1 Single-Site Model



Organization Suitability

The Single-Site model is suitable for medium-sized businesses and government operations that reside at one site and that need basic call processing, some contact center capabilities, and basic messaging and conferencing. Such operations include legal and financial professional offices, and municipal government offices.

Design Characteristics

The Single-Site model is designed to be locally managed and administered. It can operate on a wired or wireless LAN. Local and long distance calling is achieved through gateway connectivity with the PSTN by various combinations of T1/E1 CAS and PRI.

User Roles and Endpoints

The Single-Site model provides flexible communications features for operators and administrative assistants. There are some executive phones, some of which are video-capable. Most other employees use digital telephones, including wireless telephones, and a voice messaging system, which this model also provides. In addition, some staff may take orders or provide technical support. This model provides basic contact center capabilities to handle these requirements.

Some users, such as building services and shipping and receiving employees, may require mobile phones. This model provides on-campus device mobility features for these users.

Supported Applications

The Single-Site model supports applications that provide a wide array of advanced features. These applications include:

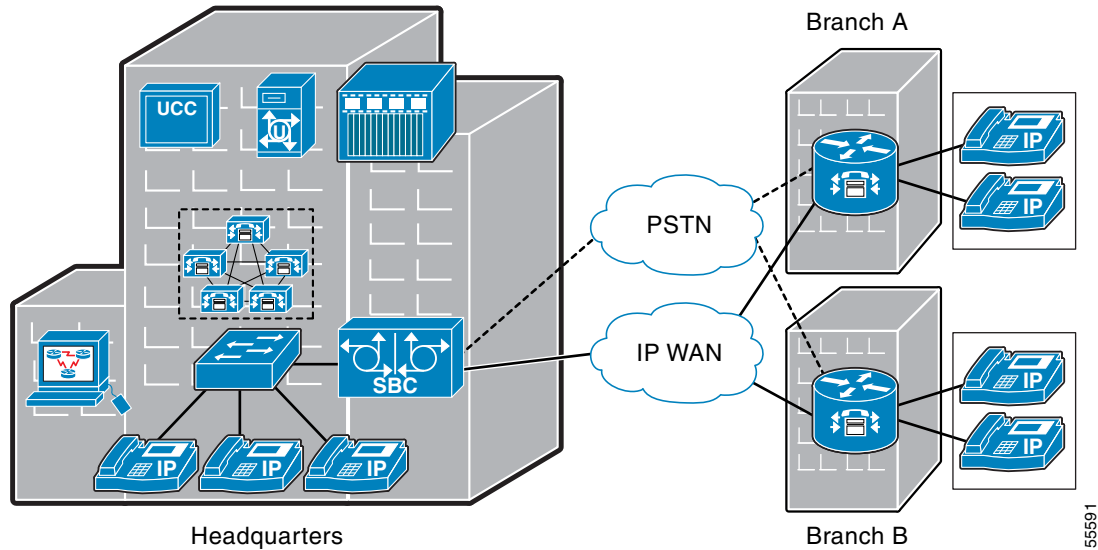
- Call processing:
 - Cisco Unified Communications Manager
 - Cisco Unified Communications Manager Express
 - Cisco Unified Communications Manager Business Edition
- Contact Center:
 - Cisco Unified Contact Center Express
 - Cisco Unified Contact Center Enterprise
 - Cisco Unified IP IVR
 - Cisco Unified Customer Voice Portal
- Messaging:
 - Cisco Unity
 - Cisco Unity Connection
 - Cisco Unity Express
- Instant messaging and presence: Cisco Unified Presence
- Conferencing:
 - Cisco Unified MeetingPlace Express
 - Cisco Unified Videoconferencing
- System management:
 - Cisco Unified Communications Manager Serviceability Tools
 - Cisco Unified Operations Manager
 - Cisco Unified Service Monitor
 - Cisco Unified Service Statistics Manager
 - Cisco Unified Provisioning Manager
 - Cisco Monitor Manager and Monitor Director (for small or medium size deployments with Cisco Unified Communications Manager Express and or Cisco Unity Express)

Multisite Centralized Call Processing Model

The Multisite Centralized Call Processing model is designed for distributed operations with a large central or headquarters site and multiple remote or branch sites. This model can support up to a total of 30,000 phones distributed among up to a maximum of 1000 sites. Based upon the bandwidth available, each site can support any number of users up to the overall total of 30,000 phones.

Figure 2-2 shows an example of this model.

Figure 2-2 Multisite Centralized Call Processing Model



Organization Suitability

The Multisite Centralized Call Processing model is suitable for businesses such as banks, which include a corporate headquarters and many local or regional offices.

Design Characteristics

In the Multisite Centralized Call Processing model, each branch site connects to the headquarters site or sites through a WAN. Branch sites receive call processing functions from the headquarters site. Failover capabilities at each branch site ensure that it can continue to operate if the WAN connection to the headquarters site is lost. Branch sites include small contact center capabilities.

The WAN connection between the headquarters and branch sites can be frame relay, MPLS, or site-to-site VPN. Each branch site can operate on a wired or wireless LAN.

Connectivity with legacy PBXs in the headquarters site can be provided T1/E1 CAS, PRI, Q SIG, and DPNSS. Connectivity to the PSTN in the headquarters site is provided through various combinations of T1/E1 CAS and PRI.

Local calling is achieved through gateway connectivity. Long distance calling for branch sites uses the WAN for on-net calling. Off-net long distance traffic is backhauled over the WAN to one or more drop-off gateways.

This model is designed to be administered at the headquarters location.

User Roles and Endpoints

Headquarters roles and endpoints are identical to those described in the “[Single-Site Model](#)” section on [page 2-2](#). Branch sites access the call processing capabilities in the headquarters site. While there are some executive phones, most employees use digital telephones and the central voice messaging system.

Some staff may take orders or provide technical support. This model provides basic contact center capabilities in the branches to handle these requirements.

Supported Applications

The Multisite Centralized Call Processing model supports applications that provide comprehensive features for all sites. These applications include:

- Call processing:
 - Cisco Unified Communications Manager (in central site)
 - Cisco Unified Communications Manager Express for fixed remote teleworker applications (in central site)
 - Cisco Unified Communications Manager Business Edition (in central site)
 - Unified SRST or Cisco Unified Communications Manager Express in SRST mode (as backup for Cisco Unified Communications Manager in branch sites and for Cisco Unified Communications Manager Business Edition in branch or central sites).
- Contact Center:
 - Cisco Unified Contact Center Enterprise (in headquarters)
 - Cisco Unified Contact Center Express (based in headquarters)
 - Cisco Unified Customer Voice Portal (for queueing and self-service at headquarters or branches). Unified Customer Voice Portal is an interactive voiceXML-based response (IVR) solution that provides carrier-class IVR and IP switching services on Voice over IP (VoIP) networks. You can integrate Unified CVP with Unified Contact Center Enterprise or can deploy as a self-service IVR solution.
 - Cisco unified IP IVR for centralized queuing.
- Messaging:
 - Cisco Unity (based in headquarters)
 - Cisco Unity Connection
 - Cisco Unity Express
- Instant messaging and presence: Cisco Unified Presence (based in headquarters)
- Conferencing:
 - Cisco Unified MeetingPlace (based in headquarters)
 - Cisco Unified Videoconferencing

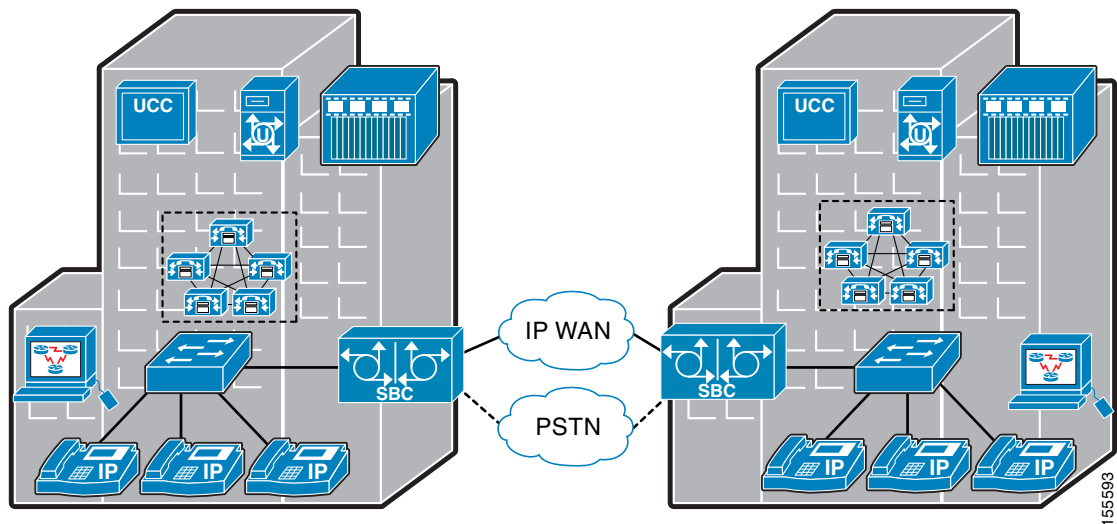
- System management:
 - Cisco Unified Operations Manager (based in headquarters)
 - Cisco Unified Service Monitor (based in headquarters)
 - Cisco Unified Service Statistics Manager (based in headquarters)
 - Cisco Unified Provisioning Manager
 - Cisco Monitor Manager and Monitor Director (for monitoring deployments with Cisco Unified Communications Manager Express and Cisco Unity Express or Cisco Unified Communications Manager Express in SRST mode as backup to Cisco Unified Communications Manager)
 - Cisco netManager - Unified Communications
 - LAN Management Solution

Multisite Distributed Call Processing Model

The Multisite Distributed Call Processing model is designed for organizations with large user populations or large numbers of geographically distributed sites resulting in the need for more than a single call processing entity. This model is suited for deployments that require multiple Cisco Unified Communications Manager clusters or Cisco Unified Communications Manager Express platforms. Each call processing entity in this model is configured as a Single-Site Model (see the “[Single-Site Model](#)” section on page 2-2) or Multisite Centralized Call Processing Model (see the “[Multisite Centralized Call Processing Model](#)” section on page 2-4) and each has a common dial plan and feature set.

Figure 2-3 shows an example of this model.

Figure 2-3 *Multisite Distributed Call Processing Model*



Organization Suitability

The Multisite Distributed Call Processing model is suitable for business operations that consist of multiple sites in various regions. Such operations include technology, manufacturing, transportation, and distribution and logistics companies.

Design Characteristics

Each site in the Multisite Distributed Call Processing model can operate on a wired or wireless LAN. The intersite WAN connection can be frame relay, MPLS, or site-to-site VPN. Each branch site can operate on a wired or wireless LAN.

Local calling is achieved through gateway connectivity at each site. Long distance calling for each site uses the WAN for on-net calling. Off-net long distance traffic is backhauled over the WAN to one or more drop-off gateways.

User Roles and Endpoints

Each site in the Multisite Distributed Call Processing model has the same user roles and endpoints that are described in the [“Multisite Centralized Call Processing Model” section on page 2-4](#).

Supported Applications

The Multisite Distributed Call Processing model supports applications that provide powerful, flexible, and scalable features. These applications include:

- Call processing:
 - Cisco Unified Communications Manager (large sites or deployments)
 - Cisco Unified Communications Manager Business Edition
 - Cisco Unified Communications Manager Express (smaller sites or deployments)
- Contact Center:
 - Cisco Unified Contact Center Enterprise (in one or more locations)
 - Cisco Unified IP IVR (for centralized queuing)
 - Cisco Unified Customer Voice Portal (for centralized or distributed queuing and self-service). Unified Customer Voice Portal is an interactive voiceXML-based response (IVR) solution that provides carrier-class IVR and IP switching services on Voice over IP (VoIP) networks. You can integrate Unified CVP with Unified Contact Center Enterprise or can deploy as a self-service IVR solution.
- Messaging:
 - Cisco Unity
 - Cisco Unity Connection
 - Cisco Unity Express
- Instant messaging and presence: Cisco Unified Presence (in one or more locations)
- Conferencing:
 - Cisco Unified MeetingPlace
 - Cisco Unified Videoconferencing
- System management:
 - Cisco Unified Operations Manager
 - Cisco Unified Service Monitor

- Cisco Unified Service Statistics Manager
- Cisco Unified Provisioning Manager
- Cisco Monitor Manager and Cisco Monitor Director
- Cisco netManager - Unified Communications
- LAN Management Solution

Clustering Over IP WAN Call Processing Model

The Clustering Over IP WAN Call Processing model is designed for organizations with large user populations across multiple sites that are connected by an IP WAN with the QoS features enabled. The Clustering Over IP WAN supports the two deployment models:

- Local Failover Deployment Model

Local failover requires that you place the Unified Communications Manager subscriber and backup servers at the same site, with no WAN between them. This deployment model is ideal for two to four sites with Unified Communications Manager.

- Remote Failover Deployment Model

Remote failover allows you to deploy primary and backup call processing servers split across the WAN. Using this deployment model, you may have up to eight sites with Unified Communications Manager subscribers being backed up by Unified Communications Manager subscribers at another site.

You can also use a combination of the two deployment models to satisfy specific site requirements. For example, two main sites may each have primary and backup subscribers, with another two sites containing only a primary server each and utilizing either shared backups or dedicated backups at the two main sites.

Organization Suitability

The Clustering Over IP WAN Call Processing model is suitable for business operations that consist of multiple sites in various regions connected over an IP WAN. Such operations include technology, manufacturing, transportation, and distribution and logistics companies.

Design Characteristics

The local failover and remote failover sites in the Clustering Over IP WAN Call Processing model operates over an IP WAN. The intersite WAN connection can be frame relay, MPLS, or site-to-site VPN.

The IP WAN must conform to the following maximum delay and minimum bandwidth requirements:

- The maximum allowed round-trip time (RTT) between any two servers in the Unified Communication Manager cluster is 80 ms.
- A minimum of 1.544 Mbps (T1) bandwidth is required for Intra-Cluster Communication Signaling (ICCS) for every 10,000 busy hour call attempts (BHCA) between sites that are clustered over the WAN. This is a minimum bandwidth requirement for call control traffic, and it applies to deployments where directory numbers are not shared between sites that are clustered over the WAN.

- In addition to the bandwidth required for Intra-Cluster Communication Signaling (ICCS) traffic, a minimum of 1.544 Mbps (T1) bandwidth is required for database and other inter-server traffic for every remote subscriber server.

The IP WAN network should also be engineered to provide sufficient prioritized bandwidth for all ICCS traffic, especially the priority ICCS traffic. Standard QoS mechanisms must be implemented to avoid congestion and packet loss. If packets are lost due to line errors or other conditions, the ICCS packet will be retransmitted because it uses the TCP protocol for reliable transmission. The retransmission might result in a call being delayed during setup, disconnect (teardown), or other supplementary services during the call.

For additional details on IP WAN delay, bandwidth requirements, and QoS engineering, refer to the *Clustering Over the IP WAN* section in the *Unified Communications Deployment Models* chapter of Unified Communications SRND at <http://www.cisco.com/go/srnd>.

User Roles and Endpoints

The local failover and remote failover sites in the Clustering Over IP WAN Call Processing model has the same user roles and endpoints that are described in the “[Multisite Centralized Call Processing Model](#)” section on page 2-4.

Some of the key advantages of clustering over the WAN are:

- Single point of administration for users for all sites within the cluster
- Feature transparency
- Shared line appearances
- Extension mobility
- Unified dial plan

Supported Applications

The Clustering Over IP WAN Call Processing model supports applications that provide powerful, flexible, and scalable features. These applications include:

- Call processing:
 - Cisco Unified Communications Manager (subscriber and backup)
 - Cisco Unified Communications Manager Express (smaller sites or deployments)
 - Unified SRST or Cisco Unified Communications Manager Express in SRST mode.
- Contact Center:
 - Cisco Unified Contact Center Enterprise
 - Cisco Unified IP IVR (for centralized queuing)
 - Cisco Unified Customer Voice Portal (for centralized or distributed queuing and self-service). Unified Customer Voice Portal is an interactive voiceXML-based response (IVR) solution that provides carrier-class IVR and IP switching services on Voice over IP (VoIP) networks. You can integrate Unified CVP with Unified Contact Center Enterprise or can deploy as a self-service IVR solution.
- Messaging:
 - Cisco Unity

- Cisco Unity Connection
- Cisco Unity Express
- Instant messaging and presence: Cisco Unified Presence
- Conferencing:
 - Cisco Unified MeetingPlace
 - Cisco Unified Videoconferencing
- System management:
 - Cisco Unified Operations Manager
 - Cisco Unified Service Monitor
 - Cisco Unified Service Statistics Manager
 - Cisco Unified Provisioning Manager
 - Cisco Monitor Manager and Cisco Monitor Director (for deployments of Cisco Unified Communications Manager Express in SRST mode as a backup to Cisco Unified Communications Manager)
 - Cisco netManager - Unified Communications
 - LAN Management Solution

Major Components of Deployment Models

Table 2-1 shows the major Cisco components in each Cisco Unified Communications deployment model.

Table 2-1 Deployment Models Components Summary

Components	Single-Site Model	Multisite Centralized Call Processing Model	Multisite Distributed Call Processing Model	Clustering Over IP WAN Call Processing model
Scale	<ul style="list-style-type: none"> • Up to 30,000 phones with Cisco Unified Communications Manager • Up to 240 phones with Cisco Unified Communications Manager Express • Up to 575 phones (and up to 500 users) with Cisco Unified Communications Manager Business Edition 	<ul style="list-style-type: none"> • Up to 30,000 phones and 1,000 sites with Cisco Unified Communications Manager • Up to 240 phones with Cisco Unified Communications Manager Express for small or branch site or fixed remote teleworker applications • Up to 575 phones (and 500 users) and up to a total of 20 sites with Cisco Unified Communications Manager Business Edition 	<ul style="list-style-type: none"> • Up to 30,000 phones per Cisco Unified Communications Manager instance • Up to 240 phones per Cisco Unified Communications Manager Express instance. 	<ul style="list-style-type: none"> • Up to 30,000 phones per Cisco Unified Communications Manager instance

Table 2-1 Deployment Models Components Summary (continued)

Components	Single-Site Model	Multisite Centralized Call Processing Model	Multisite Distributed Call Processing Model	Clustering Over IP WAN Call Processing model
Call Processing	<ul style="list-style-type: none"> • Cisco Unified Communications Manager • Cisco Unified Communications Manager Express • Cisco Unified Communications Manager Business Edition 	<ul style="list-style-type: none"> • Cisco Unified Communications Manager • Cisco Unified Communications Manager Business Edition (in central site) • Cisco Unified Communications Manager Express for fixed remote teleworker applications (in central site) • Unified SRST or Cisco Unified Communications Manager Express (as backup to Cisco Unified Communications Manager and Cisco Unified Communications Manager Business Edition) 	<ul style="list-style-type: none"> • Cisco Unified Communications Manager (in one or more locations) • Cisco Unified Communications Manager Express (in one or more locations) 	<ul style="list-style-type: none"> • Cisco Unified Communications Manager (subscriber and backup)
Contact Center	<ul style="list-style-type: none"> • Cisco Unified Contact Center Enterprise • Cisco Unified Contact Center Express • Cisco Unified IP IVR • Cisco Unified Customer Voice Portal 	<ul style="list-style-type: none"> • Cisco Unified Contact Center Enterprise (based in head- quarters) • Cisco Unified Contact Center Express (based in headquarters) • Cisco Unified Customer Voice Portal (in head- quarters or branches) 	<ul style="list-style-type: none"> • Cisco Unified Contact Center Enterprise (in one or more locations) • Cisco Unified IP IVR • Cisco Unified Customer Voice Portal 	<ul style="list-style-type: none"> • Cisco Unified Contact Center Enterprise • Cisco Unified Customer Voice Portal
Messaging	<ul style="list-style-type: none"> • Cisco Unity • Cisco Unity Connection • Cisco Unity Express 	<ul style="list-style-type: none"> • Cisco Unity (based in head- quarters) • Cisco Unity Connection • Cisco Unity Express 	<ul style="list-style-type: none"> • Cisco Unity (in one or more locations) • Cisco Unity Connection • Cisco Unity Express 	<ul style="list-style-type: none"> • Cisco Unity • Cisco Unity Connection • Cisco Unity Express
Instant Messaging and Presence	<ul style="list-style-type: none"> • Cisco Unified Presence 	<ul style="list-style-type: none"> • Cisco Unified Presence (in head- quarters) 	<ul style="list-style-type: none"> • Cisco Unified Presence (in one or more locations) 	<ul style="list-style-type: none"> • Cisco Unified Presence

Table 2-1 Deployment Models Components Summary (continued)

Components	Single-Site Model	Multisite Centralized Call Processing Model	Multisite Distributed Call Processing Model	Clustering Over IP WAN Call Processing model
Conferencing	<ul style="list-style-type: none"> • Cisco Unified Meeting- Place • Cisco Unified Video-conferencing 	<ul style="list-style-type: none"> • Cisco Unified Meeting-Place (based in head-quarters) • Cisco Unified Video-conferencing 	<ul style="list-style-type: none"> • Cisco Unified Meeting- Place (in one or more locations) • Cisco Unified Video-conferencing 	<ul style="list-style-type: none"> • Cisco Unified Meeting- Place • Cisco Unified Video-conferencing
System Management	<ul style="list-style-type: none"> • Cisco Unified Operations Manager • Cisco Unified Service Monitor • Cisco Unified Service Statistics Manager • Cisco Unified Provisioning Manager • Cisco Monitor Manager and Monitor Director (for small or medium size deployments with Cisco Unified Communications Manager Express and or Cisco Unity Express) • LAN Management Solution 	<p>Based in head- quarters:</p> <ul style="list-style-type: none"> • Cisco Unified Operations Manager • Cisco Unified Service Monitor • Cisco Unified Service Statistics Manager • Cisco Unified Provisioning Manager • Cisco Monitor Manager and Monitor Director (for monitoring deployments with Cisco Unified Communications Manager Express and Cisco Unity Express or Cisco Unified Communications Manager Express in SRST mode as backup to Cisco Unified Communications Manager) • Cisco netManager - Unified Communications • LAN Management Solution 	<p>Distributed:</p> <ul style="list-style-type: none"> • Cisco Unified Operations Manager • Cisco Unified Service Monitor • Cisco Unified Service Statistics Manager • Cisco Unified Provisioning Manager • Cisco Monitor Manager and Monitor Director • Cisco netManager - Unified Communications • LAN Management Solution 	<ul style="list-style-type: none"> • Cisco Unified Operations Manager • Cisco Unified Service Monitor • Cisco Unified Service Statistics Manager • Cisco Unified Provisioning Manager • Cisco Monitor Manager and Cisco Monitor Director (for deployments of Cisco Unified Communications Manager Express in SRST mode as a backup to Cisco Unified Communications Manager) • Cisco netManager - Unified Communications
Off-Premises Calling	<ul style="list-style-type: none"> • PSTN via gateway 	<ul style="list-style-type: none"> • Site to Site over IP WAN • PSTN as backup for branch sites 	<ul style="list-style-type: none"> • Site to Site over IP WAN • PSTN for off-network calling 	<ul style="list-style-type: none"> • Site to Site over IP WAN



CHAPTER 3

Cisco Unified Communications Component Overviews

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

- [Cisco 1800, 2800, and 3800 Series Integrated Services Routers, page 3-2](#)
- [Cisco Emergency Responder, page 3-2](#)
- [Cisco FAX Server, page 3-3](#)
- [Cisco Multiservice Cisco Unified Border Element, page 3-3](#)
- [Cisco RSVP Agent, page 3-4](#)
- [Cisco IP Communicator, page 3-4](#)
- [Cisco Unified Application Environment, page 3-4](#)
- [Cisco Unified Communications Manager, page 3-5](#)
- [Cisco Attendant Console, page 3-5](#)
- [Cisco Unified Communications Manager Express, page 3-5](#)
- [Cisco Unified Communications Manager Business Edition, page 3-6](#)
- [Cisco Unified Contact Center Enterprise, page 3-7](#)
- [Cisco Unified Contact Center Express, page 3-7](#)
- [Cisco Unified Customer Voice Portal, page 3-7](#)
- [Cisco Unified IP Phones, page 3-8](#)
- [Cisco Unified MeetingPlace, page 3-9](#)
- [Cisco Unified MeetingPlace Express, page 3-10](#)
- [Cisco Unified MeetingPlace Express VT, page 3-10](#)
- [Cisco Unified Mobility, page 3-10](#)
- [Cisco Unified Personal Communicator, page 3-12](#)
- [Cisco Unified PhoneProxy, page 3-12](#)
- [Cisco Unified Presence, page 3-12](#)
- [Cisco Unified Survivable Remote Site Telephony, page 3-13](#)
- [Cisco Unified Video Advantage, page 3-13](#)
- [Cisco Unified Videoconferencing, page 3-13](#)

- [Cisco Unity, page 3-14](#)
- [Cisco Unity Connection, page 3-14](#)
- [Cisco Unity Express, page 3-15](#)
- [Cisco VG224, page 3-16](#)
- [Security Components, page 3-16](#)
- [Management and Serviceability Components, page 3-16](#)

Cisco 1800, 2800, and 3800 Series Integrated Services Routers

Cisco 1800, 2800, and 3800 series integrated services 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 call component process for Cisco Unified Communications Manager.

Cisco 2800 and 3800 series integrated services routers communicate directly with Cisco Unified Communications Manager, 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 2800 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.

The Cisco 1800 Series integrated services routers are ideal for small to medium-sized businesses and small enterprise branch offices. The 1800 series routers help businesses to reduce costs by deploying a single, resilient system for fast, secure delivery of multiple mission-critical business services. The Cisco 1861 integrated services router is a modular platform that provides voice, data, voice-mail, automated attendant, video, and security capabilities while integrating with existing desktop applications such as calendar, e-mail, and customer relationship management (CRM) programs and with built-in security. It includes:

- Cisco Unified Communications Manager Express or Survivable Remote Site Telephony for call processing for up to 12 users
- Optional Cisco Unity Express, for voice messaging and automated attendant
- LAN switching with Power over Ethernet (PoE) expandable through Cisco Catalyst Switches
- Onboard voice ports for PSTN, PBX, and key system connections

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 Communications Manager. It helps assure that Cisco Unified Communications Manager 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 Communications Manager 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:

- Automatically tracks IP phone location
- Provides emergency call routing instructions to Cisco Unified Communications Manager
- Identifies caller location to local exchange carriers and PSAPs
- Alerts customer security personnel to emergency calls in progress
- 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 Cisco Unified Border Element

The Cisco Multiservice Cisco Unified Border Element (CUBE) 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 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 Cisco Unified Border Element 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 Cisco Unified Border Element 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 Communications Manager deployments. Cisco RSVP Agent employs Resource Reservation Protocol (RSVP), an IETF standards-based signaling protocol for reserving bandwidth in an IP network. The RSVP protocol enables dynamic adjustment to changes in the network, supports complex network topologies, and enables call admission decisions.

Cisco RSVP Agent offer benefits such as:

- Provides guaranteed WAN bandwidth for Cisco Unified Communications Manager calls
- Supports complex network topologies, including meshed designs, redundant links, and dynamically changing topologies
- Controls the quality and availability of voice and video calls, and authorization of calls
- Provides seamless interworking of any call control signaling that Cisco Unified Communications Manager supports such as SIP, H.323, Media Gateway Control Protocol (MGCP), and Skinny Client Control Protocol (SCCP).

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 Communications Manager, 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 Communications Manager 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 Application Environment

Cisco Unified Application Environment enables the rapid development, reliable execution, and automated management of applications that converge voice and video with enterprise applications and data. It is a suite of products including:

- Cisco Unified Application Designer—Enables developers to visually construct applications by dragging and dropping prebuilt functions onto a graphical communications business logic canvas and visually updating parameters associated with the graphical functions.

- Cisco Unified Application Server—Abstracts the complexity of telephony protocols, separates application logic from core call routing to protect Cisco Unified Communications Manager, and provides a standard way to manage all of an organization's unified communications applications.
- Cisco Unified Media Engine—Provides ready-to-use, sophisticated media processing capabilities for all applications built using the Cisco Unified Application Designer—functions such as interactive voice response (IVR), conferencing, transcoding, text-to-speech (TTS), speech recognition, and speaker verification.

For additional information, go to:

http://www.cisco.com/en/US/products/ps6789/Products_Sub_Category_Home.html

Cisco Unified Communications Manager

Cisco Unified Communications Manager software is the call-processing component of the Cisco Unified Communications system. Cisco Unified Communications Manager 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 Communications Manager open telephony APIs. Cisco Unified Communications Manager offers a suite of integrated voice applications and utilities, including the Cisco Unified Communications Manager Attendant Console, an ad-hoc conferencing application, the Cisco Unified Communications Manager Bulk Administration Tool, the Cisco Unified Communications Manager CDR (call detail record) Analysis and Reporting Tool, the Cisco Unified Communications Manager Real-Time Monitoring Tool, and the Cisco Unified Communications Manager Assistant application.

Cisco Attendant Console

The three types of Cisco Attendant Console supported by Cisco Unified Communications Manager are:

- Cisco Unified Communications Manager Attendant Console (AC)—native AC within Cisco Unified Communications Manager
- Cisco Unified Business Attendant Console—OEM'd version of ARC Console
- Cisco Unified Department Attendant Console—OEM'd version of ARC Console

Cisco Attendant Console integrates traditional time-division multiplexing (TDM) telephony functions with advanced IP telephony applications and services such as Lightweight Directory Access Protocol (LDAP) directory. Cisco Unified Communications Manager Attendant Console gives the user the ability to monitor the state of every line in the system and to efficiently dispatch calls.

For additional information, go to:

<http://www.cisco.com/en/US/products/sw/voicesw/ps555/index.html>

Cisco Unified Communications Manager Express

Cisco Unified Communications Manager 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 or on the new Unified Communications 500 Series router for a cost-effective, highly reliable, IP communications solution. Cisco Unified Communications Manager Express helps ensure investment protection and offers scalability because all hardware and software is fully compatible with Cisco Unified Communications Manager and Cisco Unified Survivable Remote Site Telephony.

Cisco Unified Communications Manager 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 Communications Manager deployment

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

- Accomplish initial installation of Cisco Unified Communications Manager Express easily using a setup tool that prompts for answers to pertinent questions
- 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 Communications Manager Business Edition

The Cisco Unified Communications Manager Business Edition is the call-processing, mobility, and messaging component of the Cisco Unified Communications system for medium-sized businesses. It includes the features and capabilities of Cisco Unified Communications Manager, Cisco Unified Mobility, and Cisco Unity Connection co-resident on a single, low-cost Media Convergence Server.

The Cisco Unified Communications Manager Business Edition is designed to support 150 to 500 endpoints in one main and up to five remote locations. Installation is simplified as the applications come pre-loaded onto the server. And management of all applications can be performed through a consolidated interface.

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. Cisco Unified Contact Center Enterprise is made up of Unified Intelligent Contact Management Enterprise, which provides pre-routing and post-routing capabilities, Cisco Unified Communications Manager, and Cisco Unified IP IVR.

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 customizable 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 SIP Phone 3911—Suitable for public spaces, lobbies, workshops, and warehouses where the phone is not assigned to any one user. Includes one line (DN), fixed feature keys, and a 2-line x 24-character display along with two menu select keys and a two-way rocker for scrolling control. The Cisco Unified SIP Phone 3911 supports IEEE 802.3af Power over Ethernet (PoE), or local power through an optional power adaptor.
- 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 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 Wireless IP Phone 7921G (IEEE 802.11a/b/g wireless IP phone)—Supports up to six lines, enhances security features, XML services and many telephony features as well as a full color display and built-in speaker phone. Offers security and QoS features and several power options, including IEEE 802.11a, b, and g standards that allow customers to use 7921G in the 2.4 GHz or 5 GHz bands.
- Cisco Unified IP Phone 7931G—Suitable for a user in retail, commercial, or manufacturing who conducts moderate telephone traffic. Includes twenty-four lighted line keys, each of which can be assigned to a different line, dedicated keys for hold, redial, and transfer, as well as four soft keys. Also includes 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 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.
- Cisco Unified IP Conference Station 7937G—Extends the capabilities of the 7936G, which includes access to network data, XML applications, and web-based services. Suitable for 40 feet by 40 feet conference rooms. Adds IEEE 802.3af inline power, wideband codec, internationalization and localization, IP Phone XML services, and wireless lapel microphone.

- Cisco Unified IP Phones 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 7942G / 7962G—Suitable for advanced business use, managers and executives. Provides support for high-quality wideband audio (G.722 / TIA920-adherent) and a high-resolution 4-bit grey scale display.
- Cisco Unified IP Phone 7945G / 7965G / 7975G—Suitable for enhanced business use, manager and executive. Provides support for high-quality wideband audio (G.722 / TIA920-adherent), a backlit TFT color display, and an integrated Gigabit Ethernet switch and a four-way navigation rocker with Select key.
- 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.

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, video (up to 150 concurrent video users), and web conferencing solution for medium-sized organizations. It allows users to communicate and collaborate from any place at any time through a phone or video and a web browser. Cisco MeetingPlace Express integrates meeting management and control capabilities directly into web and Cisco Unified IP Phone interfaces. Users can easily set up meetings from a variety of interfaces, including Microsoft Outlook, web browsers, telephones, and the included IP phone application.

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

For additional information, go to:

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

Cisco Unified MeetingPlace Express VT

Cisco Unified MeetingPlace Express VT is an integrated voice, video, and web conferencing solution for Cisco Unified Communications Manager environments in midsize organizations. It integrates rich-media conferencing functions that enable users to extend point-to-point voice and video communications to multiparty voice, video, and web conferences.

**Note**

Cisco Unified MeetingPlace Express VT supports only adhoc Voice and Video conferencing.

Cisco Unified MeetingPlace Express VT lets you integrate advanced conferencing capabilities with Cisco Unified Communications Manager. Installed on a single server, it supports common industry protocols to ensure connectivity with a range of video endpoints. And it maximizes video investments by supporting traditional room-based videoconferencing and desktop video telephony environments.

For additional information, go to:

<http://www.cisco.com/go/meetingplaceexpress>

Cisco Unified Mobility

Cisco Unified Mobility makes Cisco Mobile Connect services available to Cisco Unified Communications Manager 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 while Cisco Mobile Voice Access and Cisco Enterprise Feature Access 2-Stage Dialing provide users the ability to 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 Mobile Voice Access, and Enterprise Feature Access 2-Stage Dialing, Cisco Unified Mobility uses the web-based system administration and user configuration facilities built in to the Cisco Unified Communications Manager, which enables enterprise mobile workers to take advantage of these features.

Cisco Unified Mobility 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
- Cisco Enterprise Feature Access 2-Stage Dialing
- Mid-call features (Hold, Resume, Transfer, Conference)

For additional information, go to:

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

Cisco Unified Messaging Gateway

The Cisco Unified Messaging Gateway provides an open and secure method of intelligently routing messages and exchanging subscriber and directory information within a unified messaging network. It acts as the central hub in a network of Cisco unified messaging solutions and third-party gateways that interface with older voicemail systems.

The Cisco Unified Messaging Gateway:

- Unified messaging network scale as required for branch office customers and larger distributed enterprises
- Centralize voicemail system management
- Transparently integrate Cisco Unified Communications solutions into existing voicemail installations
- Integrates small to large-scale unified messaging deployments that consist of 5 Cisco Unity Express systems and above and supports up to 10,000 mixed Cisco Unity Express, Cisco Unity, and Cisco Unity Connection systems.

For additional information, go to:

http://www.cisco.com/en/US/partner/products/ps8605/tsd_products_support_series_home.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 PhoneProxy

Cisco Unified PhoneProxy acts as an application layer gateway, transparently proxying all SCCP, RTP, HTTP, and TFTP communications between Cisco IP phones or Cisco IP Communicator and Cisco Unified Communications Manager, while simultaneously obscuring Cisco Unified Communications Manager information from these endpoints. It supports both secure SCCP and secure RTP and provides a built-in Web application for authentication of users who are activating IP phones for secure use.

For additional information, go to:

<http://www.cisco.com/en/US/products/ps7057/index.html>

Cisco Unified Presence

Cisco Unified Presence 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 uses SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions) technology to support instant messaging (IM) and presence.

Cisco Unified Presence 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 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 provides a core IM service for Cisco Unified IP Phones that are connected to Cisco Unified Communications Manager. Cisco Unified Presence also supports interoperability with Microsoft Live Communications Server (LCS) 2005, Microsoft Office Communications Server (OCS) 2007, and Microsoft Office Communicator client, enabling specific Office Communicator functions to work with Cisco Unified IP Phones supported on Cisco Unified Communications Manager. Finally, the SIP and SIMPLE interface on the Cisco Unified Presence 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 Communications Manager 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 2800 and 3800 integrated services routers.

SRST software automatically detects a connectivity failure between Cisco Unified Communications Manager and IP phones at a branch office. SRST initiates a process to automatically configure the Cisco 2800 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 Communications Manager, the system automatically shifts call-processing functions back to the primary Cisco Unified Communications Manager 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 3515, 3522, 3527, and 3545. 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 Communications Manager 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 Unity 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 Communications Manager or Cisco Unified Communications Manager Express environments. In Cisco Unified Communications Manager 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 Communications Manager 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 Communications Manager or Cisco Unified Communications Manager 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>

Security Components

To provide voice security, the Cisco Unified Communications System includes the following components:

- Cisco Firewall Services Module—Integrated services module for the Cisco Catalyst 6500 Series switches and Cisco 7600 Series routers that tracks the state of all network communications and prevents unauthorized network access. It delivers strong application-layer security through intelligent, application-aware inspection engines that examine network flows at Layers 4-7, including protection for voice, multimedia, instant messaging, and peer-to-peer applications.
- Cisco Adaptive Security Appliance—Integrates the features of the Cisco PIX 500 Series security appliances, Cisco IPS 4200 Series sensors, and Cisco VPN 3000 Series concentrators to provide proactive threat defense, network activity and application traffic control, and VPN connectivity.
- Cisco Intrusion Prevention System—Stand-alone device (IPS 4200) and integrated services module (IDSM) for the Cisco Catalyst 6500 Series switches that detects, classifies, and stops threats including worms, spyware/adware, network viruses, and application abuse.
- Cisco Network Admission Control Appliance (formerly Cisco Clean Access)—Automatically detects, isolates, and cleans infected or vulnerable devices that attempt to access the network. It determines whether machines are compliant with security policies and repairs any vulnerabilities before permitting them to access the network.
- CiscoWorks Management Center for Cisco Security Agents—Provides centralized security event log collection from and centralized policy or version upgrades for Cisco Security Agents.

For more information about these components, go to:

<http://www.cisco.com/en/US/products/hw/vpndevc/index.html#products>

Management and Serviceability Components

The Cisco Unified Communications Solution includes the following complementary products, solutions, and services to help centrally manage an entire deployment:

- **Unified Readiness Manager**—Used to perform network assessments and identify gaps to be addressed prior the implementation of a unified communications system. It generates a gap analysis report that provide detailed information about the status of the network analyzed against predefined rules that verify layer 2 and 3 components, redundancy in the network, WAN circuit performance, and device performance. In addition, it allows users to simulate voice bearer traffic between agents deployed in the network and provides details about jitter, packet loss, and delay.
- **Resource Management Essentials**—Allows network administrators to view and update the status and configuration of all Cisco devices, including switches, access servers, and routers, from anywhere on the network through a standard web client.
RME can rapidly and reliably deploy Cisco software images and view configurations of Cisco routers and switches.
- **Cisco Unified Campus Manager**—Provides graphical views of network topology and manages VLANs.
- **Cisco Unified Operations Manager**—Used for comprehensive monitoring with proactive and reactive diagnostics for the Cisco Unified Communications system. It provides:
 - Built-in rules, which provide contextual diagnostics and enable troubleshooting of service-impacting outages.
 - A real-time, service-level view of the Cisco Unified Communications system, including the current operational status of each element.
 - Capabilities for application-level testing of telephony functions, which can be used proactively and reactively to identify problems and ensure that applications are functioning properly, for dial-plan validation, as well as for monitoring video-enabled endpoints.
- **Cisco netManager Unified Communications**—Provides monitoring for data systems in Mid-Market deployments. This tool allows you to visualize, monitor, threshold, and alert your data network.
- **Cisco Unified Service Monitor**—Used to monitor and evaluate the quality of voice in Cisco Unified Communications solutions. It provides:
 - Continuous monitoring of active calls supported by the Cisco Unified Communications system with near-real-time notification when the voice quality of a call fails to meet a user-defined mean opinion score (MOS).
 - Reports that characterize the user experience as measured by the system and details on the endpoints that are most frequently related to voice-quality alerts.
- **Cisco Unified Network Manager**—Used for proactive, operational and performance monitoring for small and medium sized businesses that deploy unified communications. It provides:
 - A visual representation of the network topology,1 highlighting unified communications application dependencies with the ability to inspect and act on operational outages from within the topology
 - Inventory and status information of all the devices and phones in the deployment.
- **Cisco Unified Provisioning Manager**—Used for the provisioning and activation of Cisco Unified Communications products. It allows administrators to manage initial deployments and implementations, and then permits delegation of the ongoing operational provisioning and activation tasks that are required for changes to services for individual subscribers. It provides:
 - A single, consolidated view of subscribers across the organization.
 - A set of business-level, policy-driven management abstractions for managing subscriber services across the Cisco Unified Communications infrastructure.

- Cisco Unified Service Statistics Manager—Provides statistics management, analysis, and reporting capabilities for a Unified Communications deployment. It leverages the data collection capabilities of Unified Operations Manager and Service Monitor to gather Cisco Unified Communications statistics information from a variety of Cisco devices and systems (including Unified Communications Manager, Unity, Unity Connection, Unified Communications Manager Express, Unity Express, Unified Contact Center, and Unified Contact Center Express). It stores the statistics in a database and provides statistical analysis and reporting.
- CiscoWorks LAN Management Solution (LMS)—Provides a suite of management tools that simplify configuring, administrating, monitoring, and troubleshooting Cisco networks. These tools provide an integrated system for sharing device information across applications, and offer capabilities that include:
 - Network discovery, topology views, end-station tracking, and VLAN management
 - Hardware and software inventory management, centralized configuration tools, and syslog monitoring
 - Network response time and availability monitoring and tracking
 - Real-time device, link, and port traffic management, analysis, and reporting
 - Presentation of current operational status of an IP Communications deployment and service-level views of the network
 - Contextual diagnostic tools to assist with troubleshooting
 - Presentation of service-quality alerts by using the information available through Cisco Unified Service Monitor (when deployed)
 - Presentation of current information about connectivity- and registration-related outages that are affecting IP phones in the network, and information that identifies the IP phones
 - Tracking of IP Communications devices and the IP phone inventory, tracking of IP phone status changes (providing reports that document move, add, and change operations on IP phones in the network)
 - Real-time notifications using SNMP traps, syslog notifications, and e-mail
 - Real-time voice quality monitoring and real-time voice quality alerts
 - Network discovery, topology views, end-station tracking, and VLAN management
- Cisco Unified Communication Essential Operate service—Provides hardware and software maintenance and support for Cisco voice applications. Support activities include:
 - Incident troubleshooting
 - Incident remediation
 - Network infrastructure device replacement
 - Access to applications software updates
 - Assistance using leading practices
- Cisco Unified Communications Select Operate Service—Provides proactive support for Cisco voice technologies that combines Cisco technical support with voice application monitoring and reporting. Support activities include:
 - Incident troubleshooting
 - Incident remediation
 - Provisioning monitoring solution
 - Monitoring and notification

- Network infrastructure device replacement
- Access to applications software updates
- Assistance using leading practices
- Cisco Unified Communications Remote Management Service—Provides a remote management service that offers comprehensive monitoring, issue resolution, and day-to-day management of voice applications and converged networks. Support and management activities include:
 - IPC system monitoring
 - Incident diagnosing
 - Defining remediation actions required to resolve incident
 - Incident resolution, which can include managing break/fix service request, applying software updates and patches, or managing hardware replacements
 - Day-to-day operational changes in a network, including logical move, adds, changes, and deletions
 - Daily backup configurations for Cisco OS, Cisco Catalyst OS, and servers
 - Reporting
 - Maintenance management of third-party equipment

For more information about these components, go to:

<http://www.cisco.com/en/US/products/sw/netmgtsw/index.html#products>



CHAPTER 4

Component Protocols and APIs

This chapter lists the protocols and call control application program interfaces (APIs) that are supported by various Cisco Unified Communications components.

This chapter includes these topics:

- [Call Control Signaling Protocols, page 4-1](#)
- [Cisco Unified Communications Application Program Interfaces, page 4-3](#)

Call Control Signaling Protocols

Cisco Unified Communications components support an array of call control signaling protocols. [Table 4-1](#) shows the call control signaling protocols that are supported by each component.

Table 4-1 Call Control Signaling Protocol Support

	DPNSS	H.320	H.323	ISDN	MGCP	SCCP	SIP	QSIG	T1 CAS
Cisco Emergency Responder						◆	◆		
Cisco IP Communicator						◆	◆		
Cisco Unified Communications Manager	◆		◆	◆	◆	◆	◆	◆ ¹	◆
Cisco Unified Communications Manager Express			◆			◆	◆		
Cisco Unified Communications Manager Business Edition	◆		◆	◆	◆	◆	◆	◆	◆
Cisco Unified Contact Center Enterprise			◆			◆	◆		

Table 4-1 Call Control Signaling Protocol Support (continued)

	DPNSS	H.320	H.323	ISDN	MGCP	SCCP	SIP	QSIG	T1 CAS
Cisco Unified Contact Center Express			◆			◆	◆		
Cisco Unified Customer Voice Portal			◆			◆	◆		
Cisco Unified IP Phones						◆	◆		
Cisco Unified MeetingPlace			◆				◆		
Cisco Unified MeetingPlace Express			◆				◆		
Cisco Unified MeetingPlace Express VT						◆			
Cisco Unified Personal Communicator							◆		
Cisco Unified Presence							◆ ²		
Cisco Unified Survivable Remote Site Telephony			◆			◆	◆		
Cisco Unified Video Advantage						◆ ³			
Cisco Unity						◆	◆		
Cisco Unity Connection						◆	◆		
Cisco Unity Express							◆		
Gateways	◆	◆	◆	◆	◆	◆ ⁴	◆	◆	◆

1. Cisco Unified Communications Manager does not support QSIG protocol directly, but only through a MGCP gateway. In such cases Cisco Unified Communications Manager also supports DPNSS, ISDN, and T1 CAS protocols.
2. Also supports SIMPLE.
3. Cisco Unified Video Advantage does not support SCCP directly, but only through a SCCP based endpoint.
4. VG248 and VG224 supports SCCP. ISR platforms can also register their FXS ports to Cisco Unified Communication Manager through SCCP.

Cisco Unified Communications Application Program Interfaces

Cisco Unified Communications Application Programming Interfaces (APIs) provide you with the flexibility to customize the capabilities of many Cisco Unified Communications components.

Table 4-2 shows the call control signaling APIs that are supported by each component.

Table 4-2 Cisco Unified Communications Application Programming Interfaces

	AXL	CTIQBE	HTTP	IMAP	JTAPI	LDAP	MRCP	SNMP	SOAP	SQL	TAPI	TFTP	VPIM	VXML	XML ¹
Cisco Emergency Responder			◆					◆			◆				
Cisco Unified Communications Manager	◆	◆	◆		◆	◆		◆	◆	◆	◆	◆			
Cisco Unified Communications Manager Express			◆					◆							
Cisco Unified Communications Manager Business Edition	◆	◆	◆	◆	◆	◆		◆	◆	◆	◆	◆	◆		
Cisco Unified Contact Center Enterprise		◆	◆		◆	◆		◆ ²		◆					
Cisco Unified Contact Center Express			◆		◆	◆	◆			◆				◆	
Cisco Unified Customer Voice Portal			◆				◆							◆	
Cisco Unified IP Phones			◆									◆			◆
Cisco Unified MeetingPlace			◆			◆		◆	◆ ³	◆					◆ ⁴
Cisco Unified MeetingPlace Express			◆			◆		◆							
Cisco Unified MeetingPlace Express VT			◆			◆		◆							
Cisco Unified Personal Communicator		◆	◆	◆		◆			◆			◆			

Table 4-2 Cisco Unified Communications Application Programming Interfaces

	AXL	CTIQBE	HTTP	IMAP	JTAPI	LDAP	MRCP	SNMP	SOAP	SQL	TAPI	TFTP	VPIM	VXML	XML ¹
Cisco Unified Presence	◆	◆	◆			◆		◆	◆	◆					
Cisco Unity			◆	◆		◆				◆			◆		
Cisco Unity Connection	◆		◆	◆		◆		◆		◆			◆		
Cisco Unity Express		◆	◆		◆			◆			◆	◆	◆		
Gateways															◆

1. For Cisco IP Phones
2. Supported in Windows platforms
3. Support between Video Integration and Video Admin
4. Cisco Unified Meeting Place supports XML between Video Integration and Video Admin and between Video Admin and MCU



CHAPTER 5

Deployment Methodology

Deploying a Cisco Unified Communications system involves a series of several steps. These steps include analyzing your requirements, designing your system, and implementing the Cisco Unified Communications components. This process will likely involve collaboration between your business and technical personnel and various representatives and experts from Cisco.

This chapter provides a high-level overview of some of the key steps that you will follow when you deploy a Cisco Unified Communications system.

This chapter includes these sections:

- [Step 1: Determine Your Requirements, page 5-1](#)
- [Step 2: Determine the Solution Requirements, page 5-2](#)
- [Step 3: Assess Your Network and Infrastructure Readiness, page 5-2](#)
- [Step 4: Assess Your Operational Readiness, page 5-3](#)
- [Step 5: Develop Site Requirements, page 5-3](#)
- [Step 6: Develop a Detailed Design, page 5-3](#)
- [Step 7: Develop Your Implementation Plan, page 5-4](#)
- [Step 8: Stage and Configure Your Solution, page 5-4](#)
- [Step 9: Install the Solution, page 5-4](#)

Step 1: Determine Your Requirements

The first step in deploying the Cisco Unified Communications system is to determine the requirements for your situation. This step involves:

- Analyzing your business operations to determine features and functions that you need. For example, consider the requirements that are described in [Table 5-1](#).

Table 5-1 Representative System Requirements

Requirement	Example Considerations
Business needs	<ul style="list-style-type: none"> • Do calls to a customer service number need to be distributed among agents at various locations? • Do voice messages need to be recorded? • Is intra-office abbreviated dialing required?
Site requirements	<ul style="list-style-type: none"> • Where are the locations that the system will serve? • How should the system components be distributed across sites?
Availability	<ul style="list-style-type: none"> • Is availability needed 24 hours a day or only during normal business hours? • What is the required recovery time after failure?
Capacity	<ul style="list-style-type: none"> • How many users must be supported? • How many simultaneous calls must be supported?
Integration	<ul style="list-style-type: none"> • Will the Cisco Unified Communications system need to integrate with an existing application or phone system? • Are there legacy components that must be maintained?
Deployment schedule	<ul style="list-style-type: none"> • When must the system be installed and operating? • How will the system be phased in?

- Review options and determine the implications of each alternative.
- Define the components that meet your requirements.

Step 2: Determine the Solution Requirements

After you determine your requirements, you are ready to define the solution that meet your requirements. In this step, you determine the component and component options that meet your business and operational needs.

The solution consists of the Cisco Unified Communications platforms and systems that make up the architecture. It also includes the features, functions, and applications that provide the services that you require.

The solution also may include third-party products.

Step 3: Assess Your Network and Infrastructure Readiness

Network and infrastructure readiness assessment involves the review and audit of all network infrastructure areas that will be affected by the deployment. The assessment should be performed at each site where you will deploy Cisco Unified Communications. Items to assess include:

- Network design (routing and switching network)
- Software

- Hardware
- Power/environment
- Network links
- Network services

Step 4: Assess Your Operational Readiness

Operational readiness assessment involves determining your ability to administer and manage Cisco Unified Communications. Based on this assessment, you will determine whether additional products and services are required, either temporarily or long-term, when you deploy the system.

Operational areas to consider include:

- System configuration
- System monitoring
- System upgrading

Step 5: Develop Site Requirements

When you develop your site requirements, you identify the hardware, software, and physical and environmental needs that relate to the implementation and operation of the Cisco Unified Communications system at each location where you will deploy the solution.

To assist with this process, review the high-level design documents to understand the component requirements for the solution at each location. Consider these issues:

- Solution hardware components
- Solution software levels
- Telephone company circuits
- WAN connectivity
- Solution equipment electrical requirements
- Solution environmental/physical space requirements
- Solution equipment rack and cabinet locations and layouts

Step 6: Develop a Detailed Design

After you develop your site requirements, you are ready to develop a detailed design for the Cisco Unified Communications system based on the requirements that you identified.

The detailed design will address a wide variety of issues regarding each Cisco Unified Communications component that you will implement. These issues include:

- Network infrastructure
- Interoperability requirements
- Call processing system requirements
- Application software requirements

- Customer interaction requirements
- Corporate directory architecture and requirements
- Messaging system architecture and requirements
- Conferencing requirements
- Current utilization of voice messaging system, integration plans and migration strategy
- Enterprise directory and messaging security requirements

Step 7: Develop Your Implementation Plan

Developing an implementation plan involves defining the processes required to carry out the implementation the Cisco Unified Communications system. In this step, make sure to consider:

- Accurate scheduling of any site-specific actions needed prior to implementation
- Equipment delivery and staging
- Resource requirements for network and site-specific implementation, including third-party support requirements
- Project phases and deadlines
- Acceptance criteria for each project phase
- Training

Step 8: Stage and Configure Your Solution

Staging and configuring your Cisco Unified Communications system can help make final installation more efficient. For this step, you can perform some or all of these tasks:

- Assemble the components that will be installed at each site
- Perform basic testing of the hardware and software
- Pre-configure of the devices as appropriate

Step 9: Install the Solution

Installing the Cisco Unified Communications system involves installing and configuring your network infrastructure and installing and setting up the system components. After you verify the readiness of this equipment, you perform the following general steps to install the solution:

- Catalog and inventory the system components
- Install equipment in data racks
- Complete cabling and other physical connectivity
- Place phone units in workspaces

- Verify that all units power up correctly
- Capture Installation-specific information, including:
 - Rack layouts
 - Phone placement
 - Server placements and configurations
 - Key connectivity specifics in routers and switches
 - Port-specific details



APPENDIX **A**

Cisco Unified Communications Architecture Basics

This appendix provides a high-level overview of some of the basic architectural concepts and elements upon which the Cisco Unified Communications System is built.

Additional information regarding Voice over IP technologies is available at:

http://www.cisco.com/en/US/tech/tk652/tk701/tsd_technology_support_protocol_home.html

Overview

The Cisco Unified Communications System provides support for the transmission of voice, video, and data over a single, IP-based network, which enables companies to consolidate and streamline communications. The Cisco Unified Communications System is a key part of the Cisco Unified Communications Solution, which also includes network infrastructure, security, and network management products, wireless connectivity, third-party communications applications, and a lifecycle services approach for preparing, planning, designing, implementing, operating and optimizing (PPDIOO) the system.

The Cisco Unified Communications System leverages an existing IP infrastructure (built on the Open System Interconnection [OSI] reference model) and adds support for voice and video-related devices, features, and applications. Support for major signaling protocols, such as the Session Initiation Protocol (SIP), the Media Gateway Control Protocol (MGCP), and H.323 is provided, as is the ability to integrate with legacy voice and video networks.

[Table A-1](#) shows the relationship between the OSI reference model and the voice and video protocols and functions of the Cisco Unified Communications System.

Table A-1 Voice and Video over IP in the OSI Reference Model

OSI Layer Number	OSI Layer Name	Voice	Video
7	Application	Unified IP Phone, Unified Personal Communicator, etc.	Video endpoint, Unified Video Advantage, etc.
6	Presentation	G.711, G.722, G.723, G.729	H.261, H.263, H.264
5	Session	H.323/MGCP/SIP/SCCP	H.323/SIP/SCCP
4	Transport	RTP/UDP, TCP	

OSI Layer Number	OSI Layer Name	Voice	Video
3	Network	IP	
2	Data Link	Frame Relay, ATM, Ethernet, PPP, MLP, and more	

Following this model:

- **Layer 6**—Digital signal processors (DSPs) compress/encode (decompress/decode) the voice or video signal using the chosen codec. The DSP then segments the compressed/encoded signal into frames and stores them into packets.
- **Layer 5**—The packets are transported in compliance with a signaling protocol, such as Skinny Client Control Protocol (SCCP), H.323, MGCP, or SIP.
- **Layer 4**—Signaling traffic (call setup and teardown) uses TCP as its transport medium. Media streams use Real-time Transport Protocol (RTP) over UDP for the transport protocol. RTP is used because it inserts timestamps and sequence numbers in each packet to enable synchronization at the receiving end. UDP is used because TCP would introduce delays (due to acknowledgements) that are not easily tolerated by real-time traffic.
- **Layer 3**—The IP layer provides routing and network-level addressing.
- **Layer 2**—The data-link layer protocols control and direct the transmission of the information over the physical medium.

Voice over IP

In general, the components of a VoIP network fall into the following categories:

- **Infrastructure**—Provides the foundation for the transmission of voice over an IP network. In addition to routers and switches, this includes the interfaces, devices, and features necessary to integrate VoIP devices, legacy PBX, voicemail, and directory systems, and to connect to other VoIP and legacy telephony networks. Typical products used to build the infrastructure include Cisco voice gateways (non-routing, routing, and integrated), Cisco IOS and Catalyst switches, and Cisco routers, as well as security devices, such as firewalls, virtual private networks (VPNs), and intrusion detection systems. In addition, Quality of Service (QoS), high-availability, and bandwidth provisioning (for WAN devices) should be deployed.
- **Call processing**—Provides signaling and call control services from the time a call is initiated until the time a call is terminated. The call processing component also provides feature services, such as call transfer and forwarding capabilities. In the Cisco Unified Communications System, call processing is performed by Cisco Unified Communications Manager or Communications Manager Express.
- **Applications**—Includes components that supplement the basic call processing to provide users with a complete suite of communications options. Applications in the Cisco Unified Communications System include Cisco Unity for voice messaging products, Cisco Unified MeetingPlace conference scheduling software, Cisco Emergency Responder, and applications that enhance the usability of the system and allow users to be more productive, such as the Cisco Unified Presence.
- **Voice-enabled endpoints**—Includes IP phones, soft phones, wireless IP phones, and analog gateways, which provide access to the PSTN and enable interoperability with legacy telephony devices (such as a Plain Old Telephone System [POTS] phone). For IP phones and softphones, the supported protocols are SCCP, H.323, and SIP. For gateways, the supported protocols are SCCP, H.323, SIP, and MGCP.

For a more in depth discussion of Voice over IP, see [Voice over IP Fundamentals](#) from Cisco Press.

Video over IP

Typical IP videoconferencing components include:

- Gateways—Performs translation between different protocols, audio encoding formats, and video encoding formats that may be used by the various standards. The Cisco Unified Videoconferencing gateways enable conferences using H.323, H.320, SCCP, or SIP endpoints.
- Gatekeepers— Works with the call-processing component to provide management of H.323 endpoints. The gatekeeper handles all Registration, Admission, and Status (RAS) signaling, while the call-processing component handles all of the call signaling and media negotiations.
- Conference bridges—Enables conferencing between three or more participants. Video endpoints are generally point-to-point devices, allowing only two participants per conversation. A conference bridge or multipoint conference unit (MCU) is required to extend a video conference to three or more participants.
- Video-enabled endpoints—Includes stand-alone video terminals, IP phones with integrated video capabilities, and video conferencing software on a PC. These endpoints can be H.323, H.320, SCCP, or SIP.

For additional information about videoconferencing, see the [IP Videoconferencing Solution Reference Network Design](#) guide.

Fax over IP

Fax over IP enables the interworking of standard fax machines over packet-based networks. With fax over IP, the fax image is extracted from the analog signal and converted to digital data for transmission over the IP network.

The components of the Cisco Unified Communications System support three methods for transmitting fax over IP: real-time fax, store-and-forward fax, fax pass-through.

- For real-time fax, Cisco supports Cisco fax relay and T.38 fax relay (from the International Telecommunications Union [ITU-T]). With this method, the DSP breaks down the fax tones from the sending fax machine into their specific frames (demodulation), transmits the information across the IP network using the fax relay protocol, and then converts the bits back into tones at the far side (modulation). The fax machines on either end send and receive tones as they would over the PSTN and are not aware that information is actually going across an IP network.
- For store-and-forward fax, Cisco supports T.37 (from the ITU-T). With this method, the on-ramp gateway receives a fax from a traditional fax device and converts it into a Tagged Image File Format (TIFF) file attachment. The gateway then creates a standard Multipurpose Internet Mail Extension (MIME) e-mail message and attaches the TIFF file to the e-mail. The gateway forwards the e-mail, now called a fax mail, and its attachment to the messaging infrastructure of a designated Simple Mail Transport Protocol (SMTP) server.

Store-and-forward fax allows for fax transmissions to be stored and transmitted across a packet-based network in a bulk fashion, which allows faxes to use least-cost routing and enables faxes to be stored and transmitted when toll charges are more favorable. When using store-and-forward fax, however, the user must be willing to accept fax delays that range from seconds to hours, depending upon the particular method of deployment.

- For fax pass-through, fax data is not demodulated or compressed for its transit through the packet network. With this method, the fax traffic is carried between the two gateways in RTP packets using an uncompressed format resembling the G.711 codec. The gateway does not distinguish fax calls from voice calls.

VoIP Protocols

For signaling and call control, the Cisco Unified Communications System supports the Cisco proprietary VoIP protocol, SCCP, as well as the major industry-standard protocols of H.323, SIP, and MGCP. These protocols can be categorized as using either a client-server or peer-to-peer model.

- The *client-server model* is similar to that used in traditional telephony, in which in which dumb endpoints (telephones) are controlled by centralized switches. With a client-server model, the majority of the of the intelligence resides in the centralized call processing component, which handles the switching logic and call control, and with very little processing is done by the phone itself.

The advantages of the client-server model are that it centralizes management, provisioning, and call control; it simplifies call flows for replicating legacy voice features; it reduces the amount of memory and CPU required on the phone; and it is easier for legacy voice engineers to understand.

MGCP and SCCP are examples of client-server protocols.

- The *peer-to-peer model* allows network *intelligence* to be distributed between the endpoints and call-control components. Intelligence in this instance refers to call state, calling features, call routing, provisioning, billing, or any other aspect of call handling. The endpoints can be VoIP gateways, IP phones, media servers, or any device that can initiate and terminate a VoIP call.

The advantages of the peer-to-peer model are that it is more flexible, more scalable, and more easily understood by engineers who are accustomed to running IP data networks.

SIP and H.323 are examples of peer-to-peer protocols.

Table A-2 *Protocols Supported by Cisco Unified Communications Components*

Protocol	Description
SCCP	<p>A proprietary protocol from Cisco Systems. SCCP uses the client-server model. Call control is provided by the Cisco Unified Communications Manager or Communications Manager Express. Unified IP Phones run a “skinny” client, which requires very little processing to be done by the phone itself.</p> <p>SCCP is supported by all Cisco IP Phones, by Cisco Unified Video Advantage, by many third-party video endpoints, and by select Cisco gateways.</p>
MGCP	<p>The recommendation from the ITU-T for multimedia communications over LANs. MGCP uses the client-server model and is used primarily to communicate with gateways.</p> <p>MGCP provides easier configuration and centralized management. It is supported by most Cisco gateways.</p>

Protocol	Description
SIP	<p>A recommendation from the Internet Engineering Task Force (IETF) for multimedia communications over LANs. SIP uses the peer-to-peer model. Call control is provided through a SIP proxy or redirect server. In Cisco Unified Communications Manager, SIP call control is provided through a built-in back-to-back user agent (B2BUA).</p> <p>SIP uses a simple messaging scheme and is highly scalable. It is supported by an increasing number of Cisco IP phones, by a number of third-party video endpoints, and on the trunk side of many Cisco gateways.</p>
H.323	<p>The recommendation from the ITU-T for multimedia communications over LANs. H.323 uses the peer-to-peer model. It is based on the Integrated Services Digital Network (ISDN) Q.931 protocol. Call control is provided through a gatekeeper.</p> <p>H.323 provides robust support for interfaces and interoperates easily with PSTN and SS7. It is supported by a number of third-party video endpoints and by most Cisco gateways.</p>

Voice and Video Codecs

As previously mentioned, codecs are used to encode and compress analog streams (such as voice or video) into digital signals that can then be sent across an IP network.



Tip

As a general recommendation, if bandwidth permits, it is best use a single codec throughout the campus to minimize the need for transcoding resources, which can add complexity to network design.

Characteristics of a codec are as follows:

- Codecs are either *narrowband* or *wideband*. Narrowband (used by traditional telephony systems) refers to the fact that the audio signals are passed in the range of 300-3500 Hz. With wideband, the audio signals are passed in the range of 50 to 7000 Hz. Therefore, a wideband codec allows for audio with richer tones and better quality.
- The *sampling rate* (or frequency) corresponds to the number of samples taken per second, expressed in Hz or kHz. For digital audio, typical sampling rates are 8 kHz (narrowband), 16 kHz (wideband) and 32 kHz (ultra-wideband). For digital video, typical sampling rates are 50Hz (for Phase-Alternating Line, PAL, used largely in Western Europe) and 59.94 Hz (for National Television System Committee, NTSC, used largely in North America). Both rates are supported by all the video codec listed in [Table A-3](#).
- The *compression ratio* indicates the relative difference between the original size and the compressed size of the audio or video stream. Lower compression ratios yield better quality but require greater bandwidth. In general, low-compression codecs are best suited for voice over LANs and are capable of supporting DTMF and fax. High-compression codecs are better suited for voice over WANs.
- The *complexity* refers to the amount of processing required to perform the compression. Codec complexity affects the call density—the number of calls reconciled on the DSPs. With higher codec complexity, fewer calls can be handled.

The components of the Cisco Unified Communications System support one or more of the audio and video codecs described in [Table A-3](#).

Table A-3 *Codecs Supported by Cisco Unified Communications Components*

Codec	Description
G.711	<p>A narrowband audio codec defined by the ITU-T that provides toll-quality audio at 64 Kbps. It uses pulse code modulation (PCM) and samples audio at 8 kHz. G.711 supports two companding algorithms; mu-law (used in the US and Japan) and a-law (used in Europe and other parts of the world).</p> <p>G.711 is a low-compression, medium-complexity codec.</p>
G.722	<p>A wideband audio codec defined by the ITU-T that provides high-quality audio at 32 to 64 Kbps. It uses Adaptive Differential PCM (ADPCM) and samples audio at 16 kHz.</p> <p>G.722 is similar to G.711 in compression and complexity, but provides higher quality audio.</p>
G.722.1	<p>A wideband audio codec defined by the ITU-T that provides high-quality audio at 24 and 32 Kbps. It uses Modulated Lapped Transform (MLT) and samples audio at 16 kHz.</p> <p>G.722.1 is a high-compression, low-complexity codec. It provides better quality than G.722 at lower bit-rates.</p>
G.723.1	<p>A narrowband audio codec defined by the ITU-T for videoconferencing that provides near toll-quality audio at 6.3 or 5.3 Kbps. It uses Algebraic Code Excited Linear Prediction (ACELP) and Multi Pulse-Maximum Likelihood Quantization (MP-MLQ) and samples audio at 8 kHz.</p> <p>G.723.1 is a high-compression, high-complexity codec. However, the quality is slightly lower than that of G.711.</p>
G.726	<p>A narrowband codec defined by the ITU-T that provides toll-quality audio at 32 Kbps. It uses ADPCM and samples audio at 8 kHz.</p> <p>G.726 is a medium-complexity codec. It requires half the bandwidth of G.711, while providing nearly the same quality. Note that G.726 supersedes G.723, but has no effect on G.723.1.</p>
G.728	<p>A narrowband codec defined by the ITU-T that provides near toll-quality audio at 16 Kbps. It uses Low Delay CELP (LD-CELP) and samples audio at 8 kHz.</p> <p>G.728 is a high-compression, high-complexity codec.</p>
G.729a	<p>A narrowband audio codec defined by the ITU-T that provides toll-quality audio at 8 Kbps. It uses Conjugate-Structure ACELP (CS-ACELP) and samples audio at 8 kHz.</p> <p>G.729a is a high-compression, medium-complexity codec. The quality is lower than that of G.711 and it is not appropriate for DTMF, but it is good for situations where bandwidth is limited.</p>

Codec	Description
iLBC (internet Low Bitrate Codec)	<p>A narrowband audio codec standardized by the IETF that provides better than toll-quality audio at either 13.33 or 15.2 Kbps. It uses block-independent linear-predictive coding (LPC) samples audio at 8 kHz.</p> <p>iLBC provides higher basic quality than G.729 and is royalty free. It enables graceful speech quality degradation in a lossy network. This codec is suitable for real-time communications, streaming audio, archival, and messaging.</p>
AAC (Advanced Audio Codec)	<p>A wideband audio codec standardized by the Moving Pictures Experts Group (as MPEG-4 AAC). It provides high-quality audio at rates of 32 Kbps and above. It uses AAC-LD (low delay) samples audio at 20 kHz.</p>
L16	<p>A wideband audio codec defined by the IETF as a MIME subtype. It provides reasonable quality audio at 256 Kbps. It is based on PMC and samples audio at 16 kHz.</p>
GSM-FR (Global System for Mobile Communications - Full Rate)	<p>An audio codec defined by the European Telecommunications Standards Institute (ETSI). It was originally designed for GSM digital mobile phone systems and provides somewhat less than toll-quality audio at 13 Kbps. It uses Regular Pulse Excitation with Long-Term Prediction (RPE-LTP) and samples audio at 8 kHz.</p> <p>GSM-FR is a medium-complexity codec.</p>
GSM-EFR (Enhanced Full Rate)	<p>An audio codec defined by the ETSI for digital voice that provides toll-quality audio at 12.2 Kbps. It uses ACELP and samples audio at 8 kHz.</p> <p>GSM-EFR is a high-complexity codec and provides better sound quality than GSM-FR.</p>
QCELP (Qualcomm Code Excited Linear Prediction)	<p>An audio codec defined by the Telecommunications Industry Association (TIA) for wideband spread spectrum digital communication systems that provides toll-quality audio at either 8 or 13 Kbps. As indicated by the name, it uses CELP and samples audio at 8 kHz.</p> <p>QCELP is a high-complexity codec.</p>
H.261	<p>One of the first video codecs defined by the ITU-T. It was originally used for video over ISDN. It is designed to support data rates in multiples of 64 Kbps. H.261 supports Common Intermediate Format (CIF - 352 × 288) and QCIF (176 × 144) resolutions.</p> <p>H.261 is similar to MPEG, however, H.261 requires significantly less computing overhead than MPEG for real-time encoding. Because H.261 uses constant bitrate encoding, it is better suited for use with relatively static video.</p>
H.263	<p>A video codec defined by the ITU-T as an improvement to H.261. It is used in H.323, H.320, and SIP networks. In addition to CIF and QCIF, H.263 supports SQCIF (128 × 96), 4CIF (704 × 576), and 16CIF (1408 × 1152) resolutions.</p> <p>H.263 provides lower bitrate communication, better performance, and improved error recovery. It uses half pixel precision and variable bitrate encoding, which makes H.263 better suited to accommodate motion in video.</p>
H.264	<p>The next in the evolution of video codecs. It was defined by the ITU-T in conjunction with the MPEG (as MPEG-4 Part 10) and is designed to provide higher-quality video at lower bit rates.</p> <p>H.264 provides better video quality, compression efficiency, and resilience to packet and data loss than that of H.263. It also makes better use of bandwidth, resulting in the ability to run more channels over existing systems.</p>

Voice- and Video-enabled Infrastructure

By default, an IP data network transmits data based on the concept of “best effort.” Depending on the volume of traffic and the bandwidth available, data networks can often experience delays. However, these delays are typically a matter of seconds (or fractions of seconds) and go unnoticed by users and applications, such as e-mail or file transfers. In the event of significant network congestion or minor route outages, receiving devices can wait and reorder any out-of-sequence packets and sending devices can simply resend any dropped packets.

Voice and video are very time-dependant media, which suffer greatly when subjected to the delays that data applications easily tolerate. In the event of significant congestion or outages, voice applications can only attempt to *conceal* dropped packets, often resulting in poor quality. Therefore, voice and video require an infrastructure that provides for smooth, guaranteed delivery.

A network infrastructure that transmits voice and video, especially that delivered in real-time, requires special mechanisms and technologies to ensure the safety and quality of the media as well as the efficient use of the network resources. In a voice- or video-enabled network, the following must be built into the infrastructure:

- Quality of service
- High availability
- Voice security
- Multicast capabilities

Quality of Service

Quality of Service (QoS) is defined as the measure of performance for a transmission system that reflects its transmission quality and service availability. The transmission quality of the network is determined by the following factors:

- Loss—Also known as packet loss, is a measure of packets faithfully transmitted and received compared to the total number that were transmitted. Loss is expressed as the percentage of packets that were dropped.

Loss is typically a function of availability (see the “[High Availability](#)” section on page A-10). If the network is Highly Available, then loss (during periods of non-congestion) would essentially be zero. During periods of congestion, however, QoS mechanisms can be employed to selectively determine which packets are more suitable to be dropped.

- Delay—Also known as latency, is the finite amount of time it takes a packet to reach the receiving endpoint after being transmitted from the sending endpoint. In the case of voice, this equates to the amount of time it takes for sounds to leave the speaker’s mouth and be heard in the listener’s ear. This time period is termed the “end-to-end delay.”

There are three types of delay:

- Packetization delay—The time required to sample and encode analog voice signals and digitize them into packets.
- Serialization delay—The time required to place the packet bits onto the physical media.
- Propagation delay—The time required to transmit the packet bits across the physical media.

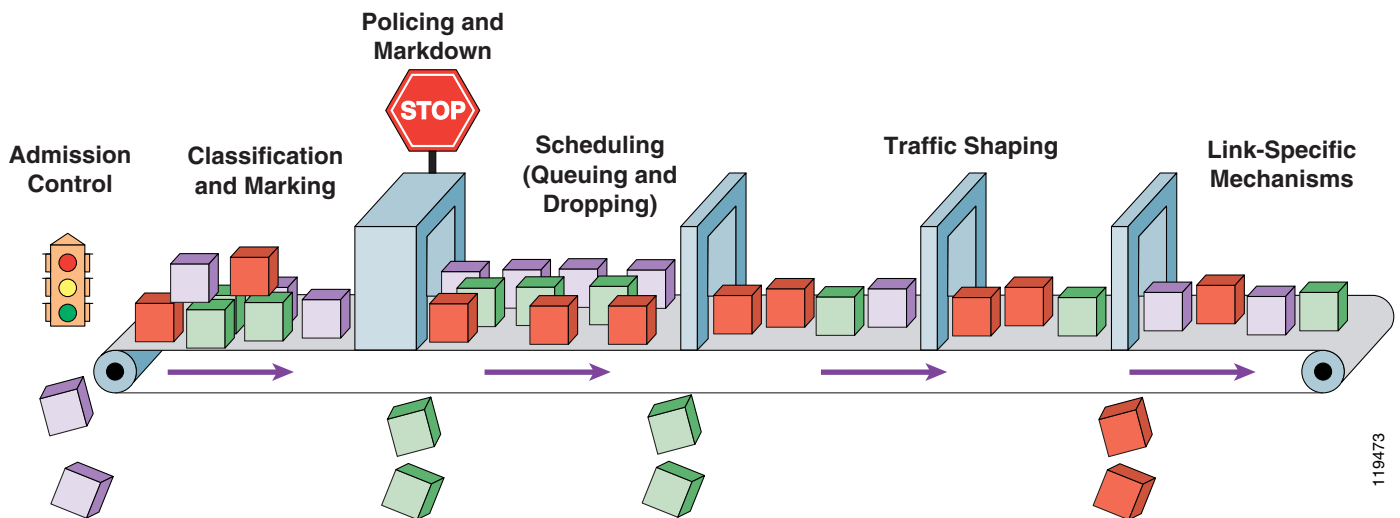
- **Delay Variation**—Also known as interpacket delay, is the difference in the end-to-end delay between packets. For example, if one packet required 100 ms to traverse the network from the source-endpoint to the destination-endpoint and the following packet required 125 ms to make the same trip, then the delay variation would be calculated as 25 ms.

Each end station in a VoIP or Video over IP conversation has a jitter buffer. Jitter buffers are used to smooth out changes in arrival times of data packets containing voice. A jitter buffer is dynamic and adaptive, and can adjust for up to a 30 ms average change in arrival times of packets. If you have instantaneous changes in arrival times of packets that are outside of the capabilities of a jitter buffer's ability to compensate you will have jitter buffer over-runs and under-runs.

- A jitter buffer under-run occurs when the arrival times of packets increases to the point where the jitter buffer has been exhausted and contains no packets to be processed by the DSPs when it is time to play out the next piece of voice or video.
- A jitter buffer over-run occurs when packets containing voice or video arrive faster than the jitter buffer can dynamically resize itself to accommodate. When this happens, packets are dropped when it is time to play out the voice or video samples, resulting in degraded voice quality.

Cisco provides a QoS toolkit that allows network administrators to minimize the effects of loss, delay, and delay variation. These tools (as shown in [Figure A-1](#)) enable the classification, scheduling, policing, and shaping of traffic—the goal being to give preferential treatment to voice and video traffic.

Figure A-1 Cisco QoS Toolkit



- **Classification** tools mark a frame or packet with a specific value. This marking (or remarking) establishes a trust boundary on which the scheduling tools depend.
- **Scheduling** tools determine how a traffic exits a device. Whenever traffic enters a device faster than it can exit it (as with speed mismatches), then a point of congestion develops. Scheduling tools use various buffers to allow higher-priority traffic to exit sooner than lower priority traffic. This behavior is controlled by queuing algorithms, which are activated only when a devices is experiencing congestion and are deactivated when the congestion clears.

- *Policers* and *shapers* are the oldest forms of QoS mechanisms. These tools have the same objectives—to identify and respond to traffic violations. Policers and shapers identify traffic violations in an identical manner; however, they respond differently to these violations. A policer typically drops traffic; a shaper typically delays the excess traffic using a buffer to hold packets and shape the flow when the data rate of the source is higher than expected.

For more information about QoS considerations and tools, see the [Enterprise QoS Solution Reference Network Design Guide](#).

High Availability

The objective of high availability is to prevent or minimize network outages. This is particularly important in networks that carry voice and video. More than a single technology, high availability is an approach to implementing a mixture of policies, technologies, and inter-related tools to ensure end-to-end availability for services, clients, and sessions. High availability heavily on network redundancy and software availability.

Network redundancy depends on redundant hardware, processors, line cards, and links. The network should be designed so that it has no single points of failure for critical hardware (for example, core switches). Hardware elements, such as cards, should be “hot swappable,” meaning they can be replaced without causing disruption to the network. Power supplies and sources should also be redundant.

Software availability depends on reliability-based protocols, such as Spanning Tree and Hot Standby Routing Protocol (HSRP). Spanning Tree, HSRP, and other protocols provide instructions to the network and/or to components of the network on how to behave in the event of a failure. Failure in this case could be a power outage, a hardware failure, or a disconnected cable. These protocols provide rules to reroute packets and reconfigure paths. The speed at which these rules are applied is called convergence. A converged network is one that, from a user standpoint, has recovered from a failure and can now process instructions and/or requests.

For more information about high availability, see [Campus Network for High Availability Design Guide](#).

Security

As with important data traffic, voice (and often video) traffic on an IP network must be secured. In some cases, the same technologies that can be used to secure a data network are employed in a VoIP network. In other cases, unique technologies must be implemented. In both cases, one of the key objectives is to protect the voice or video stream without impacting the quality.

When securing the network, it is important to consider all possible areas of vulnerability. This means protecting the network from internal and external threats, securing internal and remote connectivity, and limiting network access to devices, applications, and users that can be trusted. Comprehensive security is achieved first by securing the network itself, and then by extending that security to endpoints and applications. For voice and video communications, security must protect four critical elements:

- Network infrastructure—The switches, routers, and connecting links comprising the foundation network that carries all IP data, voice, and video traffic. This includes using tools such as:
 - Firewalls
 - Network intrusion detection and prevention systems
 - Voice- and video-enabled VPNs
 - VLAN segmentation
 - Port security

- Access control server/user authentication and authorization
 - Dynamic Address Resolution Protocol (ARP) inspection
 - IP source guard and Dynamic Host Configuration Protocol (DHCP) snooping
 - Wireless security technologies, such as wired equivalent privacy (WEP) and Lightweight Extensible Authentication Protocol (LEAP)
- Call processing systems—Servers and associated equipment for call management, control, and accounting. This includes using tools such as:
 - Digital certificates
 - Signed software images
- Endpoints—IP phones, soft phones, video terminals, and other devices that connect to the IP Communications network. This includes using tools such as:
 - Digital certificates
 - Endpoint authentication
 - Secure RTP stream encryption
 - Switch port security
 - Virus protection and integrated Cisco Security Agent
- Applications—User applications such as unified messaging, conferencing, customer contact, and custom tools that extend the capabilities of IP Communications systems. This includes using tools such as:
 - Secure management
 - Multilevel administration
 - Media encryption
 - Use of H.323 and SIP signaling
 - Hardened platform
 - Virus protection and integrated Cisco Security Agent

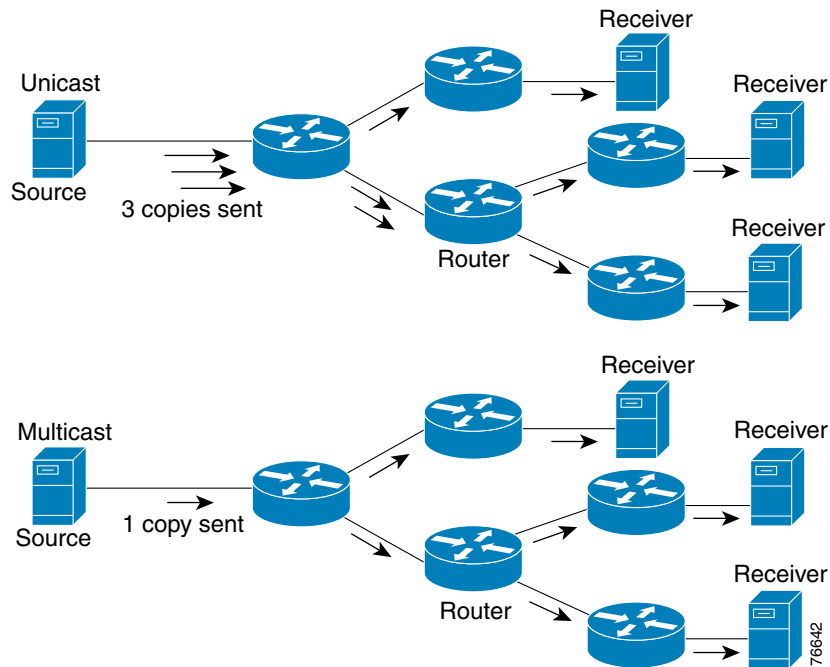
IP Multicast

IP multicast allows for a streamlined approach to delivering data to multiple hosts that need to receive the same data at the same time, such as with distance learning. With IP multicast, an audio or video stream can be sent from a single server to multiple endpoints. For example:

- When configured for IP multicast services, Music-on-Hold (MoH) can stream the same audio file to multiple IP phones without the overhead of duplicating that stream one time for each phone on hold.
- IP/TV allows for the streaming of audio, video, and slides to thousands of receivers simultaneously across the network. High-rate IP/TV streams that would normally congest a low-speed WAN link can be filtered to remain on the local campus network.

In contrast to unicast, which would send individual streams to each of the recipients, IP multicast simultaneously delivers a *single stream* of information to thousands of recipients, thereby reducing bandwidth consumption, as shown in [Figure A-2](#).

Figure A-2 IP Multicast

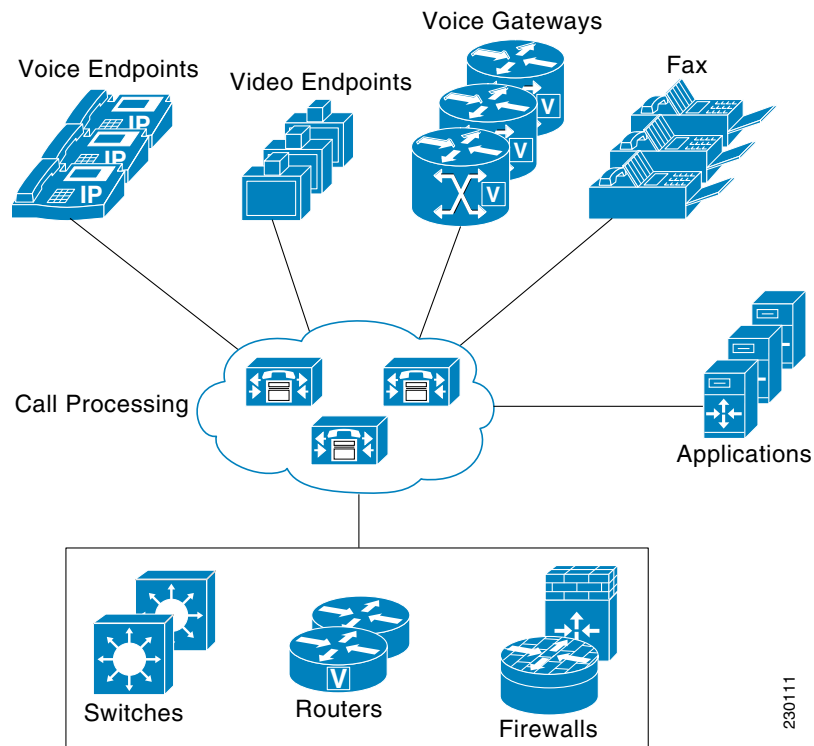


Multicast packets are replicated in the network by Cisco routers and switches enabled with Protocol Independent Multicast (PIM) and other supporting multicast protocols. These routers create “distribution trees,” which control the path that IP Multicast traffic takes through the network in order to deliver traffic to all the receivers.

For more information about IP multicast, see the [Cisco Network Infrastructure IP Multicast Design Guide](#).

Summary

The components and technologies of the Cisco Unified Communications System and the enabling infrastructure work in concert to deliver converged voice, video, and data communications.

Figure A-3 Cisco Unified Communications System

- The components and technologies employed in the infrastructure (such as QoS and IP Multicast) provide a secure, robust, reliable, and efficient foundation.
- Building on the infrastructure, the gateways and call-processing components perform the necessary conversion, integration, and control functions to enable efficient, streamlined communications.
- The applications augment the call processing to provide features and services required by users.
- And the endpoints provide access to the network services and features—enabling users to make the most of their communications system and increase their productivity.



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