Cisco Unified Communications System Description

Release 8.6(1)

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Cisco Unified Communications System Description Release 8.6(1), Release 8.6(1)

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Preface

Overview

This document provides an overview of the Unified Communications System Release 8.6(1) system. It describes the Cisco Unified Communications system-level approach, lists main 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	Purpose
Chapter 2, "Cisco Unified Communications Technology Themes"	Provides an overview of the Cisco Unified Communications Technology Themes.
Chapter 1, "Cisco Unified Communications System Overview"	Provides an overview of the Cisco Unified Communications system-level approach and architecture
Chapter 1, "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

New Components

This edition of Cisco Unified Communications System Description includes information on the following new components:

- Cisco Hosted Collaboration Solution
- Cisco Virtualization Experience Infrastructure.
- Cisco TelePresence
- Cisco Cius
- Cisco MediaSense
- Cisco Finesse
- Cisco Design Tools

For more information on these components, see Chapter 3, "Cisco Unified Communications Component Overviews".

Related Documentation

The Cisco Unified Communications Solution provides a suite of interactive documentation that covers details of the system architecture and components, installation and upgrades, troubleshooting, and 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 about 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

Subscribe to the *What's New in Cisco Product Documentation* as an RSS feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service. Cisco currently supports RSS Version 2.0.

CHAPTER 1

Cisco Unified Communications System Overview

The Unified Communications System Release 8.6(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
- Service Offerings
- Career Certifications
- Solution Bundling
- Intelligent Information Network
- Business Productivity Applications
- Customer Interaction Network
- IP Communications
- Security
- Network Management
- 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.

The Cisco Unified Communications system offers the following solutions:

- Enterprise solution for large businesses, with support for up to 30,000 users with Cisco Unified Communications Manager as the call processing component.
- Mid-market solution, with support for up to 1000 users and 1200 phones with Cisco Unified Communications Manager Business Edition 6000 as the call processing component.
- Mid-market solution, with support for up to 500 users and 575 phones with Cisco Unified Communications Manager Business Edition 5000 as the call processing component.
- Mid-market solution, with support for up to 300 users and 400 phones with Cisco Unified Communications Manager Business Edition 3000 as the call processing component.
- Small and medium sized solution, with support for up to 450 users (depending on IOS platform) with Cisco Unified Communications Manager Express as the call processing component.

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.

Cisco Unified Communications 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 Network Professional Voice (formerly 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 CCNP Voice certification can help create a telephony solution that is transparent, scalable, and manageable. A CCNP Voice 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.

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, 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 Unity 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, video, call management, presence, messaging, and web conferencing. The integrated applications include Cisco Unified Communications Manager (Unified CM), Cisco Unified Presence, Cisco Unity, Cisco Unity Connection, Cisco Unified MeetingPlace, and the Lightweight Directory Access Protocol (LDAP) version 3 (v3) server.
- 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), video, 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, Video, 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
  - Agent Greeting
  - Whisper Announcement
  - Mobile agent deployment
  - 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 Cisco Unified Communications deployment 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 7921G, 7925G and 7925-EX. 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.

Network Management

The Cisco Unified Communications system uses the following network management products to monitor the various devices deployed in the Unified Communications system:

- Cisco Unified Operations Manager—With real-time, graphical service-level views of the entire Cisco Unified Communications infrastructure, Cisco Unified Operations Manager facilitates efficient operations. It monitors the network using built-in rules, diagnoses problems, tracks changes to Cisco Unified IP Phone status, and tracks inventory of Unified Communications devices and IP phones. It provides contextual diagnostic tools to facilitate rapid troubleshooting and fault isolation. Service-level, real-time alerting and reporting is built in; in conjunction with Cisco Unified Service Monitor, it also supports alerts and reports on service quality.
- Cisco Unified Provisioning Manager—With automated processes for initial deployment and “day-2” moves, adds, and changes, Cisco Unified Provisioning Manager reduces both initial and ongoing costs of Unified Communications deployments. It is highly scalable, with batch provisioning that facilitates automated rollout to large numbers of subscribers and therefore reducing deployment time. Templates allow administrators define policy-based configurations for both initial deployment and ongoing administration to simplify and reduce the time to configure and change users.
- Cisco Unified Service Statistics Manager—With Cisco Unified Service Statistics Manager, operations personnel can generate call volume, service availability, call quality, and resource utilization reports with trending over time to identify growing operational issues before they become service-affecting. Historical reports improve security by identifying unusual call activity and can help reduce or eliminate costs associated with fraud. Utilization and capacity trend reports enable capacity planners to identify under utilized trunks, gateways and other resources to reduce costs.

- Cisco Unified Service Monitor—By supporting real-time tracking, evaluation, and reports about user experience metrics such as delay, loss, and jitter, Cisco Unified Service Monitor helps service providers achieve high user satisfaction and call quality. It measures service quality and calculates reports associated with active calls on the system. Administrators can establish alert thresholds based on device and codec types. Call quality metrics gathered from Cisco Unified IP Phones, Network Analysis Modules (NAMs) or 1040 sensors are available to Cisco Unified Operations Manager for alerting and to Cisco Unified Service Statistics Manager for capacity planning and trending.

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 (SRST) allows remote branch offices to remain in service even when the WAN access link is lost. Cisco Survivable Remote Site Voicemail (SRSV) provides backup voicemail service in the centralized messaging and centralized call processing deployment.

CHAPTER 1

Deployment Models

With Cisco Unified Communications you can choose from many deployment options, including cloud computing, hybrid, and on-premises. Regardless of the deployment model, benefits include:

- Hardware capacity assurance
- Predictable budgetary planning
- Simplified management
- Industry-leading Cisco IronPort support and corporate stability

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:

http://www.cisco.com/go/designzone

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

To learn more about cloud deployment models, go to:

http://www.cisco.com/en/US/solutions/collateral/ns340/ns517/ns224/ns836/ns976/white_paper_c11-617239.html

This chapter includes the following sections:

Deployment Overview

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

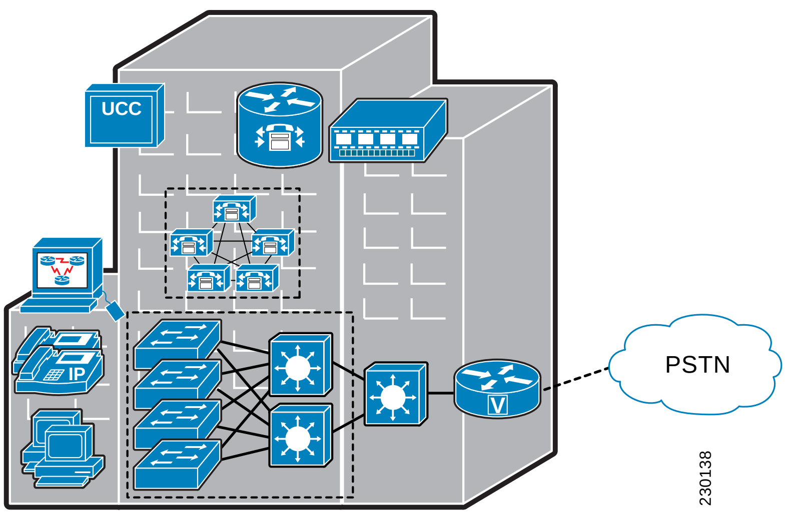
- End-to-end IP communications requirement
- Interoperability between sites
- Administrative requirements (centralized or distributed)
- Messaging requirements
- Conferencing requirements
- Availability requirements
- Mobility requirements
- Scalability requirements
- Customer interaction network requirements

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 1-1 shows an example of this model.

Figure 1-1 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
- 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

IPv6 support

The Cisco Unified Communications System support the deployment of IPv6 in Unified Communications products. The characteristics and benefits of the IPv6 single-site model is the same as those for IPv4 single-site deployments. However, the IPv6 single-site model includes the additional IPv6 and dual-stack product capabilities and features

For more information, see the Cisco Unified Communications Solution Reference Network Design (SRND), available at:

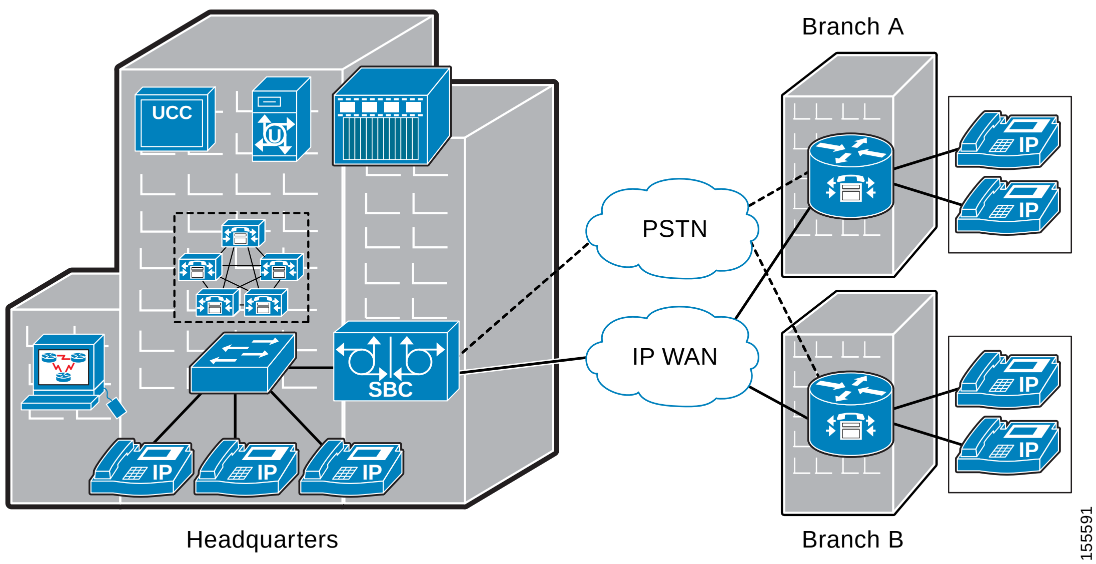
http://www.cisco.com/go/designzone

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 1-2 shows an example of this model.

Figure 1-2 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 1-2. 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)
- 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
  - LAN Management Solution

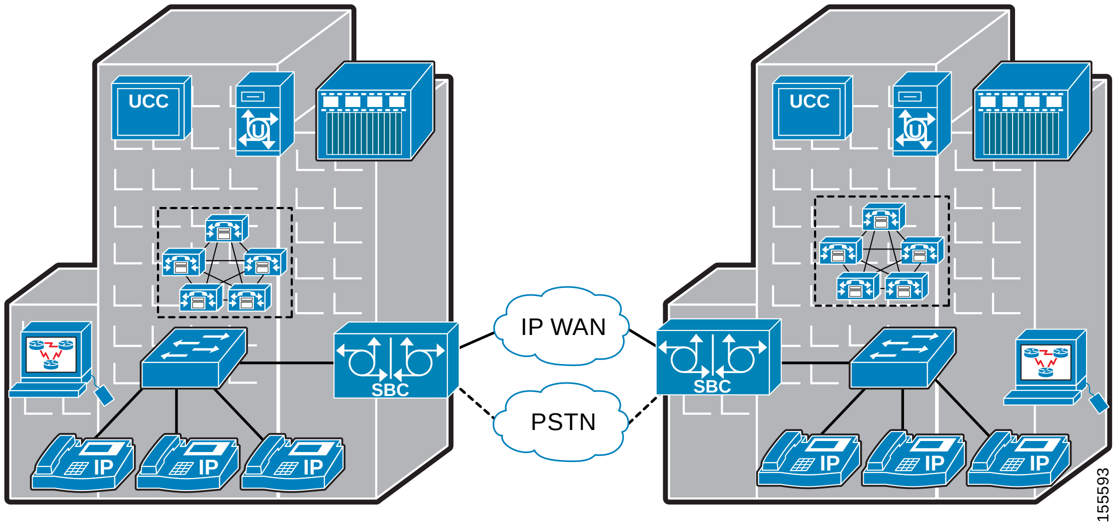
Multisite Distributed Call Processing Model

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 1-2) or Multisite Centralized Call Processing Model (see the "Multisite Centralized Call Processing Model" section on page 1-4) and each has a common dial plan and feature set. Each site has its own call processing agent cluster connected to an IP WAN that carries voice traffic between the distributed sites.

The multisite distributed call processing model supports up to 30,000 SCCP or SIP IP phones or video endpoints per cluster.

Figure 1-3 shows an example of this model.

Figure 1-3 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 1-4.

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 5000 and 6000

  - Cisco Unified Communications Manager Express (smaller sites or deployments)
  - Cisco Unified Communications Manager Session Management Edition
- Contact Center:
  - Cisco Unified Contact Center Enterprise (in one or more locations)
  - Cisco Unified IP IVR (for centralized queueing)
  - 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
- System management:
  - Cisco Unified Operations Manager
  - Cisco Unified Service Monitor
  - Cisco Unified Service Statistics Manager
  - Cisco Unified Provisioning Manager
  - LAN Management Solution

IPv6 support

The Cisco Unified Communications System support the deployment of IPv6 in Unified Communications products. The characteristics and benefits of the IPv6 multi-site WAN deployment with distributed call processing model is the same as those for IPv4 multi-site WAN deployment with distributed call processing deployments. However, the IPv6 multi-site WAN deployment with distributed call processing model includes the additional IPv6 and dual-stack product capabilities and features.

For more information, see the Cisco Unified Communications Solution Reference Network Design (SRND), available at:

http://www.cisco.com/go/designzone

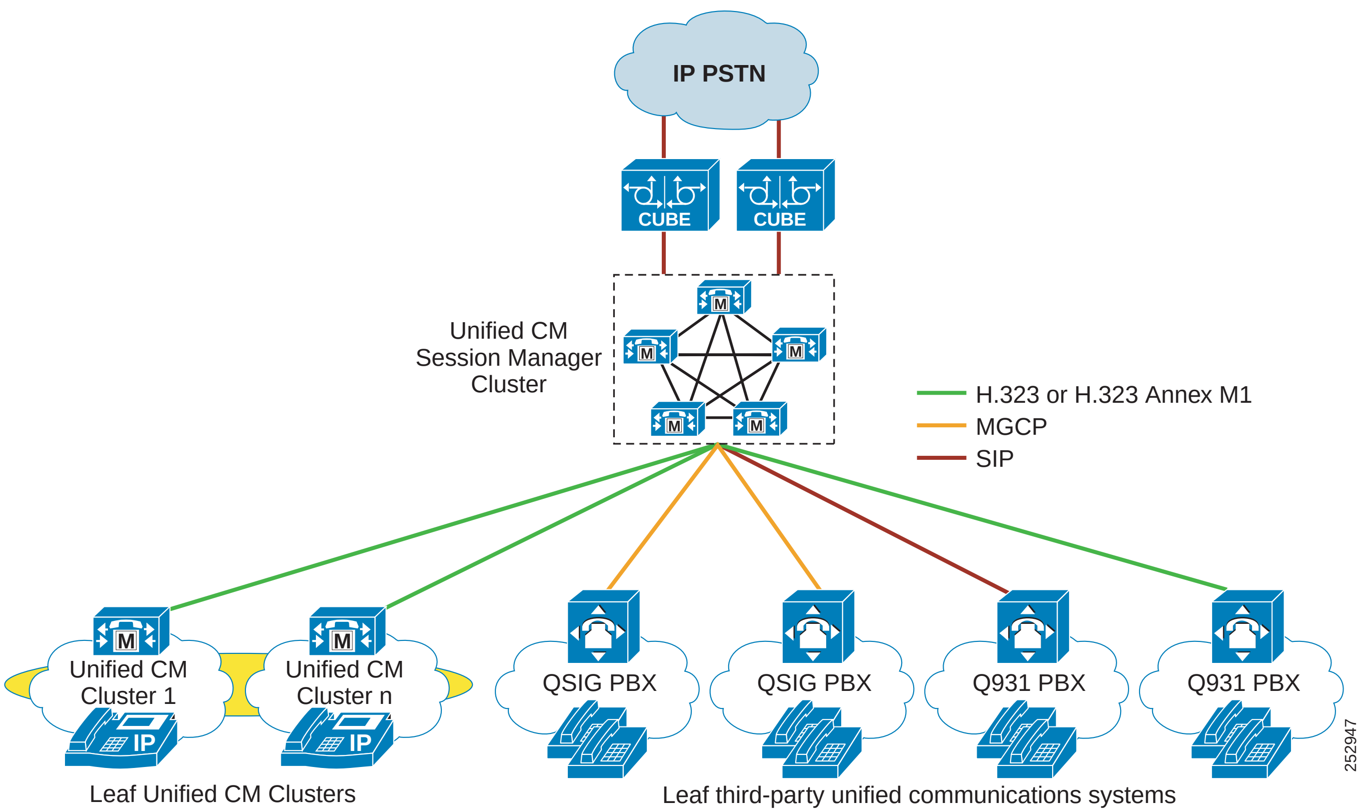
Cisco Unified Communications Manager Session Management Edition

Unified communications deployments using Cisco Unified Communications Manager Session Management Edition is a variation of the multisite distributed call processing deployment model and is typically employed to interconnect large numbers of unified communications systems through a single front-end system, in this case the Unified Communications Manager Session Management Edition.

Unified Communications Manager Session Management Edition may also be used to connect to third-party unified communications systems such as IP PSTN connections, PBXs, and centralized unified communications applications. However, as with any standard Unified Communications Manager cluster, third-party connections to Unified CM Session Management Edition must be tested for interoperability prior to use in a production environment.

Figure 1-4 shows an example of this model.

Figure 1-4 Multisite Deployment with Unified CM Session Management Edition



Unified CM Session Management Edition can be deployed if you want to:

- Create and manage a centralized dial plan—Instead of configuring each unified communications system with a dial plan, Unified CM Session Management Edition allows you to configure the leaf unified communications systems with a simplified dial plan and trunks pointing to the session management.
- Provide centralized PSTN access—Unified CM Session Management Edition can be used to aggregate PSTN access to one or more centralized IP PSTN trunks.
- Aggregate PBXs for migration to Unified Communications System—Unified CM Session Management Edition provides an aggregation point for multiple PBXs for migration from legacy PBXs to Cisco Unified Communications System.

For additional information on the deployment models, refer to the *Cisco Solution Reference Network Design (SRND)* guides and related documents available at:

http://www.cisco.com/go/designzone

Cisco Intercompany Media Engine

Cisco Intercompany Media Engine (Cisco IME) is a variation of a multisite deployment with distributed call processing, but with Cisco IME, the sites are separate enterprise organizations. The Cisco IME will route a call that would normally be sent over a PSTN trunk over the public internet. The solution learns routes in a dynamic, secure manner and provides for secure communications between organizations across the internet. Organizations that work closely together and have high levels of intercompany communications will benefit most from the enhanced communications offered by Cisco IME.

This deployment is configured with two independent Unified CM clusters interconnected via PSTN and public internet, which supports bi-directional calling using Cisco IME as well as simultaneously placing and receiving calls to non-Cisco IME sites over WAN and PSTN.

The Cisco IME deployment is supported on Unified CM 8.x and later versions or Unified CM Session Management Edition 8.x and later.

The Cisco IME components deployed are:

- Cisco Intercompany Media Engine (Cisco IME) Server
- Cisco Unified Communications Manager (Unified CM)
- Cisco Adaptive Security Appliance

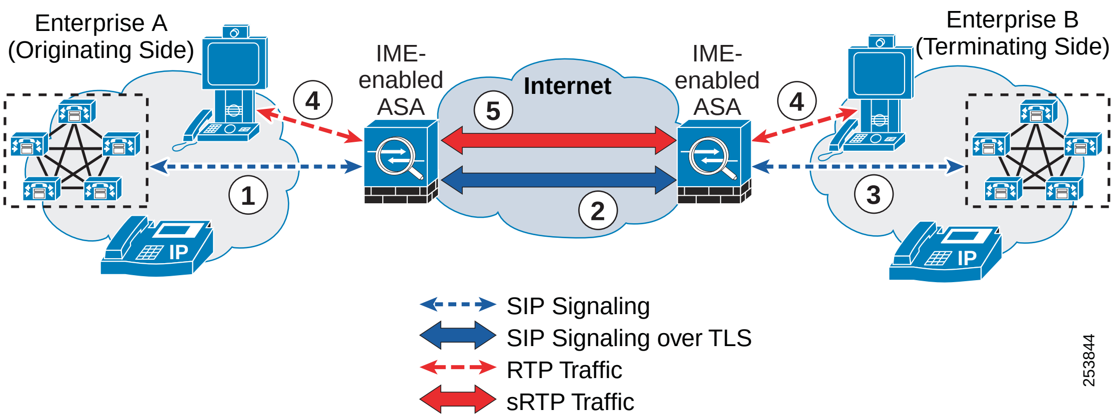
Unified CM communicates with Cisco IME servers to upload the Cisco IME designated directory numbers to the distributed cache ring and sends call records to Cisco IME for PSTN calls made by these directory numbers. Unified CM also receives Cisco IME learned routes that are validated by the Cisco IME servers and initiates dynamic SIP trunk calls to the remote directory numbers in these Cisco IME learned routes. SIP trunk signaling always flows through a Cisco IME-enabled Adaptive Security Appliance (ASA), which provides perimeter security for the solution.

Once the Cisco IME learned route is stored in the Unified CM database, the information in the route is used to set up a Cisco IME call. However, the Cisco IME server is not involved in the call processing phase. To initiate a Cisco IME call, the called number should match the Cisco IME learned route pattern in the database and the directory number of the calling phone should be enrolled in Cisco IME. Then, Unified CM dynamically invoke a Cisco IME SIP trunk to the external IP address or fully qualified domain name of the terminating enterprise.

A Cisco IME enabled ASA serves as a proxy for all Cisco IME communications with remote organizations. The ASA provides network address translation (NAT) and SIP application layer gateway (ALG) functionality to translate addressing inside the SIP messaging itself.

Figure 1-5 provides a high level view of the Cisco IME call processing.

Figure 1-5 Cisco Intercompany Media Engine Call Processing



For additional information on Cisco IME deployment, refer to the *Cisco Intercompany Media Engine* section in the *Unified Communications Deployment Models* chapter of Unified Communications SRND at: http://www.cisco.com/go/designzone

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/designzone

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 1-4.

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 queueing)
  - 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
- System management:
  - Cisco Unified Operations Manager
  - Cisco Unified Service Monitor
  - Cisco Unified Service Statistics Manager
  - Cisco Unified Provisioning Manager
  - LAN Management Solution

Major Components of Deployment Models

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

Table 1-1 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	• Up to 30,000 phones with Cisco Unified Communications Manager • Up to 450 phones (depending on IOS platform) with Cisco Unified Communications Manager Express • Up to 300 users and 400 phones with Cisco Unified Communications Manager Business Edition 3000 • Up to 500 users and 575 phones with Cisco Unified Communications Manager Business Edition 5000 • Up to 1000 users and 1200 phones with Cisco Unified Communications Manager Business Edition 6000	• Up to 30,000 phones and 1,000 sites with Cisco Unified Communications Manager • Up to 450 phones (depending on IOS platform) with Cisco Unified Communications Manager Express for small or branch site or fixed remote teleworker applications • Up to 300 users and 400 phones over a total of 10 sites with Cisco Unified Communications Manager Business Edition 3000 • Up to 500 users and 575 phones over a total of 20 sites with Cisco Unified Communications Manager Business Edition 3000 • Up to 1000 users and 1200 phones over a total of 20 sites with Cisco Unified Communications Manager Business Edition 6000	• Up to 30,000 phones per Cisco Unified Communications Manager instance • Up to 450 phones (depending on IOS platform) per Cisco Unified Communications Manager Express instance.	• Up to 30,000 phones per Cisco Unified Communications Manager instance

Table 1-1 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	• Cisco Unified Communications Manager • Cisco Unified Communications Manager Express • Cisco Unified Communications Manager Business Edition	• 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)	• Cisco Unified Communications Manager (in one or more locations) • Cisco Unified Communications Manager Express (in one or more locations) • Cisco Unified Communications Manager Session Management Edition • Cisco Intercompany Media Engine	• Cisco Unified Communications Manager (subscriber and backup)
Contact Center	• Cisco Unified Contact Center Enterprise • Cisco Unified Contact Center Express • Cisco Unified IP IVR • Cisco Unified Customer Voice Portal	• 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)	• Cisco Unified Contact Center Enterprise (in one or more locations) • Cisco Unified IP IVR • Cisco Unified Customer Voice Portal	• Cisco Unified Contact Center Enterprise • Cisco Unified Customer Voice Portal
Messaging	• Cisco Unity • Cisco Unity Connection • Cisco Unity Express	• Cisco Unity (based in head- quarters) • Cisco Unity Connection • Cisco Unity Express	• Cisco Unity (in one or more locations) • Cisco Unity Connection • Cisco Unity Express	• Cisco Unity • Cisco Unity Connection • Cisco Unity Express
Instant Messaging and Presence	• Cisco Unified Presence	• Cisco Unified Presence (in head- quarters)	• Cisco Unified Presence (in one or more locations)	• Cisco Unified Presence

Table 1-1 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	• Cisco Unified Meeting- Place • Cisco Unified Video- conferencing	• Cisco Unified Meeting- Place (based in head- quarters) • Cisco Unified Video- conferencing	• Cisco Unified Meeting- Place (in one or more locations) • Cisco Unified Video- conferencing	• Cisco Unified MeetingPlace
Network Management	• Cisco Unified Operations Manager • Cisco Unified Service Monitor • Cisco Unified Service Statistics Manager • Cisco Unified Provisioning Manager • LAN Management Solution	Based in head- quarters: • Cisco Unified Operations Manager • Cisco Unified Service Monitor • Cisco Unified Service Statistics Manager • Cisco Unified Provisioning Manager • LAN Management Solution	Distributed: • Cisco Unified Operations Manager • Cisco Unified Service Monitor • Cisco Unified Service Statistics Manager • Cisco Unified Provisioning Manager • LAN Management Solution	• Cisco Unified Operations Manager • Cisco Unified Service Monitor • Cisco Unified Service Statistics Manager • Cisco Unified Provisioning Manager
Off-Premises Calling	• PSTN via gateway	• Site to Site over IP WAN • PSTN as backup for branch sites	• Site to Site over IP WAN • PSTN for off-network calling	• Site to Site over IP WAN

CHAPTER 2

Cisco Unified Communications Technology Themes

The Cisco Unified Communications System Release 8.6(1) introduces the concept of Technology Themes. Technology Themes are solution-focused strategic areas of business support.

These key areas include the following:

- Video Interoperability and Architecture
- Desktop Virtualization/Virtual Experience Initiative
- Solutions for Mobile Unified Communications Deployment
- Hosted Collaboration Solution
- Unified Capabilities Requirements 2008 Certification
- Platform Evolution
- Mid-market Unified Communications Solution
- Ciso Cius
- Increased Operating System Compatibility

Video Interoperability and Architecture

The Video Interoperability and Architecture theme focuses on the following customer benefits:

- Greater collaboration, broader reach, and better productivity
- Native registration of Tandberg endpoints to Unified Communications Manager
- Native interoperability between Unified Communications and Cisco TelePresence System endpoints
- Secure SIP Video Signaling and Media Encryption
- SIP wideband audio codec
- Ad hoc conference bridge support
- More video-enabled Unified Communications endpoints, enabling “Video as the new Voice”

Desktop Virtualization/Virtual Experience Initiative

The Desktop Virtualization/Virtual Experience Initiative theme focuses on the following customer benefits:

- New Cisco VXC 2100 Series endpoints
- New Cisco VXC 2200 Series endpoints
- Cisco virtualized endpoints delivering Cisco Unified Communications experience to any device in any workspace environment
- Simplified IT deployment, security, and scalability
- New management tools
- Reduced operating costs lby everaging Virtualized Datacenter and Virtualized Network, providing an end-to-end virtualization solution

Solutions for Mobile Unified Communications Deployment

The Solutions for Mobile Unified Communications Deployment theme focuses on the following customer benefits:

- Rich mobile collaboration with seamless user experiences regardless of network and transport
- Cisco Mobile enhancements for Android & Nokia clients
- Improved productivity, reduced costs, and improved communication efficiency
- Mobile IM/Presence
- Dial via Office (DVO) enhancements
- Enable/disable Mobile Connect
- Mid-call over cellular
- Seamless Secure Access (SSA)

Hosted Collaboration Solution

The Hosted Collaboration Solution theme focuses on the following customer benefits:

- Optimized service fulfillment and service assurance
- Integration of new applications including Attendant Console and WebEx Connect
- Integration of new clients including CIUS, E20, and EX video endpoints
- Scaled down solution for smaller businesses
- Reduced workload and elapsed time for upgrading HCS deployments
- Integrated HCS solution with pre-deployed IMS networks
- Additional revenue generating features
- Reduced costs for customer activation, with additional automation in Service Fulfillment
- Reduce Mean Time To Repair, resulting in much higher system availability and the ability to offer more stringent SLAs

Unified Capabilities Requirements 2008 Certification

The Unified Capabilities Requirements 2008 Certification theme focuses on the following customer benefits:

- Interoperability and Information Assurance certified products for Department of Defense Components to assist in gaining approval to connect to DoD networks
- Local Session Controller (LSC)
- End Instruments (EI)
- Customer Edge Router (CER)
- Edge Boundary Controller (EBC)

Platform Evolution

The Platform Evolution theme focuses on the following customer benefits:

- Specification-based VMware support for compute, storage, network, and management hardware
- Increased investment leverage for virtualized Unified Communications deployments
- Reduced time for upgrading large multi-server/VM Cisco Unified Communications Manager deployments
- Continued security posture

Mid-market Unified Communications Solution

The Mid-Market Unified Communications Solution theme focuses on the following customer benefits:

- Foundational Unified Communications solution on a single server with integrated gateway.
- Competitive pricing
- Quick setup and easy of use requiring less IT resources
- Flexible deployment options

Ciso Cius

The Cius theme focuses on the following customer benefits:

- Video: 2-way HD video and interoperability with Cisco TelePresence and Cisco video endpoints
- Mobility: Wi-Fi enabled
- Virtual Desktop: Virtual Desktop client for centralized data center business applications
- Form factor: 7 inch multi-touch LCD that is light weight (1.15lb)
- Security: highly secure remote connection with Cisco AnyConnect Security VPN client, remote wipe, encrypted signaling and media
- Applications: Full suite of Cisco Collaboration apps including Cisco Quad, Cisco Show & Share, WebEx, Presence, and IM

- Android Applications: Support for a wide range of applications through Google Android Marketplace

Increased Operating System Compatibility

The Increased Operating System Compatibility theme focuses on the following customer benefits:

- Windows 7 support
- Windows Server 2008 R2 (64-bit) support
- MOC/OCS 14 support
- Windows 2008 Server support
- Office 2010 support

CHAPTER 3

Cisco Unified Communications Component Overviews

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

Cisco Integrated Services Routers

The Cisco 1800, 2800, 3800, 2900, 3900, 3900E series integrated services routers, and the Cisco 4451-X Integrated Services Router (Cisco ISR 4451-X) 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.

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. It includes:

- Cisco Unified Communications Manager Express or Survivable Remote Site Telephony for call processing for up to 8 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

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 2900 and 3900 series integrated services routers (ISRs) offer secure, wire-speed delivery of concurrent data, voice, and video services. The modular design of these routers provides maximum flexibility and allows you to configure the router to meet evolving needs.

The routers support virtual private network (VPN) encryption acceleration, intrusion-protection and firewall functions, and optional integrated call processing and voice mail. A wide variety of legacy network modules and interfaces, service modules (SMs), internal services modules (ISMs), next-generation packet voice/data modules (PVDM3), Services Performance Engines (SPEs), high-density interfaces for a wide range of connectivity requirements, and sufficient performance and slot density for future network expansion requirements and advanced applications are available.

Cisco 2900 and 3900 series integrated services routers with Cisco IOS Release 15.x supports FXS ports, Conferencing and transcoding DSP resources with the following gateways—MGCP 0.1, H.323, SCCP, and SIP. The Cisco 2900 and 3900 Series gateways with the PVDM3 DSPs do not support Cisco fax relay.

For additional information, go to:

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

The Cisco 4400 Series Integrated Services Router (ISR) is a modular router with LAN and WAN connectivity and supports several interface modules, including Cisco Service Modules (SMs), or Enhanced Service Modules (SM-X), and Network Interface Modules (NIMs). The router has slots that support the interface modules and modular Hard Disk Drives (HDD).

The Cisco 4400 Series Integrated Services Routers (ISRs) run on Cisco IOS XE 3.9S and later and extends the support for data, voice, and other applications. This modular architecture increases network resiliency, compared to using fewer modules in standard Cisco IOS software.

The Cisco 4400 Series modular ISRs target the following applications:

- Enterprise applications-Intended as the mid-size aggregation and gateway router typically residing in a regional or large branch office
- Service provider applications-Intended for high-end Enterprise Branch environments.

For additional information about Cisco ISR 4451-X, go to:

http://www.cisco.com/en/US/products/ps12522/tsd_products_support_series_home.html

Cisco Unified Computing System

Cisco Unified Computing System (Cisco UCS) is an architecture that integrates computing resources (CPU, memory, and I/O), IP networking, network-based storage, and virtualization, into a single highly available system. This level of integration provides economies of power and cooling, simplified server connectivity into the network, dynamic application instance repositioning between physical hosts, and pooled disk storage capacity. The architecture uses Unified fabric that provides transport for LAN, storage, and high-performance computing traffic over a single infrastructure with the help of technologies such as Fiber Channel over Ethernet. Cisco's unified fabric technology is built on a 10-Gbps Ethernet foundation that eliminates the need for multiple sets of adapters, cables, and switches for LANs, SANs, and high-performance computing networks.

The Cisco Unified Computing System:

- Streamlines data center resources to reduce total cost of ownership
- Scales service delivery to increase business agility
- Radically reduces the number of devices requiring setup, management, power, cooling, and cabling

For more details on the Cisco Unified Computing System architecture, go to:

http://www.cisco.com/go/ucs

Two types of Cisco Unified Computing System servers are available for a Unified Communications solution:

- **B-Series Blade Servers**—The Cisco UCS B200 M2 Blade Server support production-level virtualization and other mainstream data center workloads. The server is a half-width, 2-socket blade server with substantial throughput and scalability. Up to eight Cisco UCS B200 M2 Blade Servers can be housed in a Cisco UCS 5108 Blade Server Chassis, with a maximum of 320 blade servers per Unified Computing System.
- **C-Series Rack-Mount Servers**—Two models of low-profile, rack-mount C-series servers are available:
  - The Cisco UCS C200 M2 server is a high-density, 2-socket, 1 rack unit (RU) rack-mount server built for production-level network infrastructure, web services, and mainstream data center, branch, and remote-office applications.
  - The Cisco UCS C210 M2 server is a general purpose, 2-socket, 2 rack unit (RU) rack-mount server that balances performance, density, and efficiency for storage-intensive workloads. The system is built for applications such as network file servers and appliances, storage servers, database servers, and content-delivery servers.

Cisco Unified Communications can run virtualized on UCS. For more information go to:

http://www.cisco.com/go/uc-virtualized

Cisco 7800 Series Media Convergence Servers

Cisco Media Convergence Servers (MCS) provide highly available server platforms to host applications within the Cisco Unified Communications system. These platforms address enterprise customer requirements for Cisco Unified Communications Manager installations from two to 30,000 IP phones within a single Cisco Unified Communications Manager cluster.

Cisco Unified Communications Manager is supported on specific Cisco MCS 7800 series servers or on customer-provided servers that have been verified by Cisco to meet the following minimum requirements:

- Processor speed must be 2.0 GHz or greater
- Physical memory size must be 2 GB or greater
- Physical hard disk size must be 72 GB or larger

For a complete list of currently supported hardware configurations, refer to the documentation available at:

www.cisco.com/go/swonly

Note The Cisco MCS 7828 servers support only Unified Communications Manager Business Edition.

For more information about these components, go to:

http://www.cisco.com/en/US/products/hw/voiceapp/ps378/index.html

Cisco Unified IP Phones

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

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

- Business Communications Endpoints: Cisco Unified IP Phones 6900 Series

  The Cisco Unified IP Phones 6900 Series is a innovative portfolio of endpoints, delivering cost-effective business-grade voice communication services to customers worldwide. The Cisco Unified IP Phone 6900 Series offers personalization options, including the choice of two colors and two handset weights. These devices are also energy efficient, consuming less power in support of customer green initiatives. Different Cisco Unified IP Phone 6900 Series models are available with and without displays.

  For more information about the Cisco Unified IP Phones 6900 Series, go to:

  http://www.cisco.com/en/US/products/ps10326/tsd_products_support_series_home.html

- Advanced Business Endpoints: Cisco Unified IP Phones 7900 Series

  The Cisco Unified IP Phones 7900 Series provides IP phones with color liquid crystal display (LCD), including dynamic soft keys for call features and functions. This series also offers support for information services, including Extensible Markup Language (XML) capabilities to extend IP phone systems. The capability to customize XML-based services allows users access a variety of information, such as stock quotes, employee directories, and web content.

  For more information about the Cisco Unified IP Phones 7900 Series, go to:

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

- Advanced Professional Media Endpoints: Cisco Unified IP Phones 8900 Series

  The Cisco Unified IP Phones 8900 Series phones accelerate business success by delivering a high-performance, rich multimedia communications experience. This series offers a broad portfolio of XML and MIDlet applications that can help a company transform its business processes, reduce operating and administration costs, and boost productivity.

For more information about the Cisco Unified IP Phones 8900 Series, go to:

http://www.cisco.com/en/US/products/ps10451/tsd_products_support_series_home.html

- Advanced Collaborative Media Endpoints: Cisco Unified IP Phones 9900 Series

  The Cisco Unified IP Phones 9900 Series supports interactive, high-performance business video, enabled directly from the endpoint, with an optional Cisco Unified Video Camera that supports full-screen, two- and multiparty H.264 standard video.

  For more information about the Cisco Unified IP Phone 9900 Series, go to:

  http://www.cisco.com/en/US/products/ps10453/tsd_products_support_series_home.html

Cisco Unified IP Phone Expansion Modules

The Cisco Unified IP Phone Expansion Modules 7914, 7915, and 7916 are used by administrative assistants and others who need to determine the status of a number of lines beyond the current line capability of the phone.

The Cisco Unified IP Phone Expansion Modules 7914, 7915, and 7916 extend the capability of the Cisco Unified IP Phones 7960G, 7961G, 7961G-GE, 7962G, 7965G, 7970G, 7971G-GE, or 7975G with additional buttons and an LCD. The Cisco Unified IP Phone Expansion Module 7914 provides 14 buttons per module, and the Cisco Unified IP Phone Expansion Modules 7915 and 7916 provide up to 24 buttons per module. Cisco Unified IP Phones 796xG and 797xG can support up to two Cisco Unified IP Phone Expansion Modules. If the IP phone uses Cisco inline power or IEEE802.3af PoE, then the Cisco Unified IP Phone Expansion Modules 7914, 7915, and 7916 require the use of an external power adaptor (CP-PWR-CUBE-3).

Note When two Expansion Modules are used with a single phone, the second module must be the same model as the first one.

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.

The dial plan feature in Unified Communications Manager enable you to:

- Route calls based on the physical location context of the caller.
- Represent calling and called party numbers in a global form such as that described by the International Telecommunications Union's E.164 recommendation.

- Present calls to users in a format based on local dialing habits.
- Present calls to external networks (for example, the PSTN) in a manner compatible with the local requirements for calling party number, called party number, and their respective numbering types.
- Derive the global form of the calling party number on incoming calls from gateways, based on the calling number digits and the numbering type.

For additional information, go to:

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

Cisco Unified Communications Manager Business Edition

The Cisco Unified Communications Manager Business Edition 3000, 5000, and 6000 are the call-processing, mobility, and messaging component of the Cisco Unified Communications system for medium-sized businesses. 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 users in one main and up to five remote locations. It also supports up to 575 Skinny Client Control Protocol (SCCP) or Session Initiation Protocol (SIP) IP phones or video endpoints per Cisco Unified Communications Manager Business Edition autonomous system. Installation is simplified as the applications come pre-loaded onto the server. And management of all applications can be performed through a consolidated interface.

The Cisco Unified Communications Manager Business Edition supports corporate directory synchronization. This feature enables Cisco Unified Communications Manager Business Edition to synchronize directly with an existing corporate directory using LDAP integration. This feature enables administrators to provision users automatically from the corporate directory into the Cisco Unified Communications Manager Business Edition database, thus allowing administrators to maintain a single directory. This method avoids having to add, remove, or modify core user information manually in Cisco Unified Communications Manager Business Edition each time a change occurs in the corporate directory. This feature also helps the end-users authenticate using single sign-on functionality, thus reducing the number of passwords across the network.

For additional information, go to:

http://www.cisco.com/en/US/products/ps7273/tsd_products_support_general_information.html

Cisco Unified Communications Manager Session Management Edition

Cisco Unified Communications Manager Session Management Edition integrates multivendor private branch exchanges into one network and centralizes applications, trunking, dial plan, and policy control. It reduces communication tolls, cuts administrative overhead, and supports easier migration to a full IP telephony environment.

Cisco Unified Communications Manager Session Management Edition extends collaboration applications such as unified messaging, mobility, TelePresence, social networking, and web applications (using Web 2.0 interfaces) to every user on the network. Unified applications are deployed at the network core, so users on multivendor PBXs can use centrally deployed applications.

Cisco Unified Communications Manager Session Management Edition supports the following features:

- H.323 Annex M1 intercluster trunks
- SIP intercluster trunks
- SIP trunks
- H.323 trunks
- MGCP trunks
- Encrypted calls
- Multi vendor SIP and Q.SIG interoperability with Nortel, Siemens, Avaya, and Microsoft
- SIP trunk with Cisco Unified Border Element
- Voice, video, and fax calls

For additional information, go to:

http://www.cisco.com/en/US/products/ps10661/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 450 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 with support for more features than SRST 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 the Quick Configuration Tool (QCT) 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
- Monitor deployments with Cisco Monitor Manager and Cisco Monitor Director
- Use Cisco Configuration Agent (CCA) for configuration tasks

For additional information, go to:

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

Cisco Unified Presence enables the deployment of Session Initiation Protocol (SIP) or eXtensible Messaging and Presence Protocol (XMPP) technology to support unified communication 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. XMPP provides real-time communication of applications including instant messaging, presence, multi-party chat, voice and video calls, and collaboration.

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 that can facilitate aggregate presence information from multiple resources 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.

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 and IBM Lotus, enabling specific functions to work with Cisco Unified IP Phones supported on Cisco Unified Communications Manager.

The SIP/SIMPLE and XMPP interfaces on Cisco Unified Presence make it one of the most open platforms available and can provide value add presence and call control capabilities to any standards based application or service. This native dual protocol support allows for borderless business-to-business communication through the use of federation, which facilitates the exchange of presence and IM with any business that uses one of the major enterprise IM solutions such as Webex Connect, Microsoft or IBM Lotus Sametime, as well as public IM solutions such as GoogleTalk or AOL.

For additional information, go to:

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

Cisco Hosted Collaboration Solution

The Cisco Hosted Collaboration Solution is an architecture for delivering Unified Communications and Collaboration as a hosted service. The Cisco Hosted Collaboration Solution helps Cisco partners to develop and offer broad, differentiated services that increase revenue, and to minimize costs and maximize operational efficiency of services.

For more information about Cisco Hosted Collaboration Solution, go to:
http://www.cisco.com/en/US/netsol/ns1086/index.html

Cisco Virtualization Experience Infrastructure

The Cisco Virtualization Experience Infrastructure (VXI) system integrates virtualized data centers, networks, and endpoints with desktop virtualization services for comprehensive media, security, and performance acceleration. The Cisco Desktop Virtualization solution delivers the following features:

- Unprecedented control and increased security
- Rapid deployment, scaling, and lifecycle management of virtual desktops
- Improved user experience and application responsiveness
- Greater control of desktop total cost of ownership (TCO)

For more information about the Cisco Desktop Virtualization., goto:

http://www.cisco.com/en/US/products/ps11295/Products_Sub_Category_Home.html

Cisco TelePresence

The Cisco TelePresence EX90 for the desktop lets colleagues instantly collaborate face-to-face, whether separated by a hallway, a street, or several time zones. It enables faster decision making, enhances relationships, and improves efficiency. The Cisco TelePresence EX90 includes the following features:

- Full high-definition 24-inch screen with vivid, life-like 1080p30 video
- Simple touch-screen control
- One-touch sharing of high-definition (HD) content
- A built-in document camera feature
- An included wideband handset, with an option to add a headset

For more information about the Cisco TelePresence System EX Series, go to:
http://www.cisco.com/en/US/prod/collateral/ps7060/ps11303/ps11308/ps11327/data_sheet_c78-627494.html

Cisco Cius

Cisco Cius is a business tablet that supports mobile, cloud computing, HD video , business process and collaborative applications.

With an ultra-portable form factor, powerful collaborative capabilities and flexible connectivity, Cisco Cius uniquely addresses the needs of today's workforce. Because it delivers the same rich computing, communications and collaboration experience in the office, around campus and off campus, companies can consolidate the number of devices employees need with a single device.

Support for wired, wireless, and 3G/4G data service means that there are no connectivity restrictions with Cisco Cius. And when it comes to collaboration, there are no compromises. The tablet's 7-inch, high-resolution, touch-target color display offers the perfect balance between the pocket portability of smartphones and the larger display and functionality of a laptop.

For more information about Cisco Cius, go to:
http://www.cisco.com/en/US/prod/collateral/voicesw/ps6789/ps7290/ps11156/solution_overview_c22-608594.html

Cisco Intercompany Media Engine

The Cisco Intercompany Media Engine (Cisco IME) allows you to establish direct IP connectivity between enterprises by combining peer-to-peer technologies with existing PSTN infrastructure. It moves calls from the PSTN to Direct SIP trunks. The term boundary-less Unified Communications is used to describe this technology because it allows for the business-to-business extension of Unified Communications capabilities such as high-fidelity codecs, enhanced caller ID, and video telephony outside the corporate networks. The solution learns routes in a dynamic, secure manner and provides for secure communications between organizations across the internet. Organizations that work closely together and have high levels of intercompany communications will benefit most from the enhanced communications offered by Cisco IME.

Cisco IME provides the following:

- Allows any two enterprises in the world to connect over the public internet as well as support for closed user groups (CUGs) to allow cooperating enterprises to work with each other
- Requires minimal configuration; dial plan restructuring or entry of anyone else's dial plan is not required
- Requires no Service Provider support beyond public IP and basic PSTN
- Cisco IME monitors the QoS of the Real-Time Transport Protocol (RTP) traffic in real time and fallback to PSTN automatically if problems arise.

For additional information, go to:

http://www.aboutcisco.biz/en/US/products/ps10669/tsd_products_support_series_home.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 Unified Attendant Consoles

The three attendant console products supported by Cisco Unified Communications Manger are as follows:

- Cisco Unified Enterprise Attendant Console
- Cisco Unified Business Attendant Console
- Cisco Unified Department Attendant Console

Associated with a Cisco Unified IP Phone, the Cisco Unified Attendant Consoles provide the human attendant console operator with the tools to quickly accept and effectively dispatch incoming calls to individuals across the organization. The applications offer a rich set of features, including a call-queuing engine, endpoint busy status, presence integration, and full Cisco Unified Communications Manager directory search.

For more information about the Cisco Unified Attendant Consoles, go to:

http://www.cisco.com/en/US/products/ps7282/prod_software_versions_home.html

Cisco Unified Border Element

The Cisco Unified Border Element (Enterprise Edition) is Cisco's enterprise optimized Session Border Controller, supported on the Cisco 2900 and 3900 Series Integrated Services Routers (ISR) and the Cisco 1000 Series Aggregation Services Routers (ASR). The Cisco Unified Border Element (CUBE) interconnects Unified Communications networks securely, flexibly and reliably. CUBE enables end-to-end voice, video, and data between independent unified communications networks. Deploying CUBE is essential for routing voice calls beyond the enterprise boundary to Service Providers. With SIP Trunking, CUBE cuts PSTN costs and provides substantial customer savings.

The Cisco Unified Border Element with SIP trunking lowers total communications costs, optimizes network interconnections and enables rich collaboration applications. This session border controller ensures interoperability, security, and service assurance by providing the capabilities that today´s IP networks require, including the following:

- Session management
- Security
- Interworking
- Demarcation

The Cisco Unified Border Element Enterprise Edition with SIP trunking also offers the following:

- Exceptional scalability, with each chassis able to scale up to 16,000 sessions
- Extensive support for digital signal processors (DSPs) in the platform to promote complex media manipulation
- Box-to-box and in-box redundancy so that calls can continue during unscheduled outages

For additional information, go to:

www.cisco.com/go.cube

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 offers benefits such as the following:

- 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 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 Contact Center Enterprise and Cisco Unified Intelligent Contact Management Enterprise Software

Cisco Unified Contact Center Enterprise (UCCE) provides a full-featured distributed contact center infrastructure, which segments customers, provides call treatment and network-to-desktop computer telephony integration (CTI), 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. UCCE is complimented by additional components and products which provide reporting, desktop, IVR, social media, and other functionality.

For more information about Unified Contact Center Enterprise, go to:

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

and:

http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_general_information.html

Cisco Unified Contact Center Express

Cisco Unified Contact Center Express meets the needs of midmarket and enterprise branch-office or departmental companies that need easy-to-deploy, easy-to-use, secure, virtual, highly available, and sophisticated customer interaction management for up to 400 agents. Cisco Unified Contact Center Express support for powerful, agent-based service as well as fully integrated self-service applications results in reduced business costs and improved customer response by providing sophisticated and distributed automatic call distributor (ACD), interactive voice response (IVR), computer telephony integration (CTI), and agent and desktop services in a single-server, contact-center-in-a-box deployment while offering the flexibility to scale to larger, more demanding environments. Cisco Unified Contact Center Express helps ensure your business rules for inbound and outbound voice and email; and customer interaction management helps ensure that each contact is delivered to the right agent the first time.

For additional information, go to:

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

Cisco Hosted Collaboration Solution

The Cisco Hosted Collaboration Solution is an architecture that delivers Unified Communications and Collaboration as a hosted service. The Cisco Hosted Collaboration Solution allows Cisco partners to provide a wide range of Cisco collaboration applications to their customers in a subscriber based model.

This solution is composed of four integrated components:

- Cisco Unified Communications and Collaboration
- Optimized Virtualization Platform
- Centralized Management
- Service Provider System Architecture

For additional information, go to:

http://www.cisco.com/en/US/netsol/ns1086/index.html

Cisco Virtualization Experience Clients

Cisco Virtualization Experience Client (VXC) endpoints allow you to move to desktop virtualization without compromising a rich collaborative user experience.

The Cisco VXC 2100 is a compact device that is physically integrated with Cisco Unified IP Phone 8900 or 9900 Series, optimizing desk real-estate. It supports Power-over-Ethernet and is equipped with two video ports and four USB ports to support a mouse and keyboard or other peripherals in a virtual desktop environment.

The Cisco VXC 2200 is a sleek, stand-alone, small footprint zero client device which also provides users with access to a virtual desktop and business applications running in a virtualized desktop environment. Designed with the green workspace in mind, the VXC 2200 can be powered via Power over Ethernet or an optional power supply, and is equipped with two video ports and four USB ports to support a mouse and keyboard or other peripherals in a virtual desktop environment.

Cisco Virtualization Experience Client endpoints help you to:

- Choose from industry-leading desktop virtualization clients
- Deliver a better user experience with virtualized desktops
- Extend your investment in Power over Ethernet
- Conserve desktop real estate

For additional information, go to:

http://www.cisco.com/en/US/products/ps11295/Products_Sub_Category_Home.html

Cisco Unified Customer Voice Portal

The Cisco Unified Customer 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 Unified 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 Intelligence Suite and Intelligence Center

Cisco Unified Intelligence Center extends the boundaries of traditional contact center reporting by creating a comprehensive information portal where data can be integrated from multiple sources and shared throughout an organization. With this intuitive advanced reporting platform, you can report on relevant business data and web components with ease. Unified Intelligence Center provides a dashboard-based canvas for grouping multiple reporting objects together, offering a comprehensive view of contact center statistics, linking multiple reports, and integrating third-party data including workforce management, quality management, and web content.

For additional information, go to:

http://www.cisco.com/en/US/products/ps9755/index.html

Cisco Computer Telephony Integration

The Cisco Computer Telephony Integration (CTI) Option enables Cisco Unified Intelligent Contact Management (ICM) Enterprise and Cisco Unified Contact Center Enterprise to provide a complete network-to-desktop strategy, including comprehensive functionality at individual workstations.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/custcosw/ps14/index.html

Cisco Agent Desktop

Cisco Agent Desktop is a computer telephony integration (CTI) solution for single- and multisite IP-based contact centers. It is easy to deploy, configure, and manage. Powerful tools help increase agent and supervisor productivity, improve customer satisfaction, and reduce costs. An intuitive GUI decreases IT dependency and simplifies customization, maintenance, and change management. Transparent integration with Cisco Unified Contact Center helps you easily deploy CTI capabilities at new locations as customer contact operations expand.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/custcosw/ps427/index.html

Cisco MediaSense

Cisco MediaSense is a media recording platform that uses Web 2.0 Application Programming Interfaces(APIs) to expose its functionality to third-party customers so they can create custom applications.

The system is comprised of several components. The Capture Server terminates media streams for storage on a local disk; meta data associated with the recording is stored in a database, and exportable to open file format. The Application Management Server provides web services interfaces to enable applications to search for and retrieve recordings and associated call history and meta data.

Cisco MediaSense provides the following features:

- Audio capture
- Video capture
- Media storage and management
- Meta data storage and search
- Scalable and reliable architecture
- Open web-based application interfaces
- Integration with Cisco Unified Communications Manager recording interface

For additional information, go to:

http://www.cisco.com/en/US/products/ps11389/index.html

Cisco Finesse

Cisco Finesse is the next-generation agent and supervisor desktop for Cisco Unified Contact Center Enterprise, providing benefits across a variety of communities that interact with the customer service organization. It is designed to provide a collaborative experience that improves the customer experience by enhancing customer service representative experience.

For IT professionals, Cisco Finesse offers smooth integration with the Cisco Collaboration portfolio. It is standards-compliant, and offers low cost of customization of the agent and supervisor desktops.

Note The first release of Cisco Finesse, Release 8.5(1), is for lab use only. The production release of Cisco Finesse, Release 8.5(2), is scheduled to be available in the second half of CY11. This information is subject to change based on product requirements and schedules.

For more information about Cisco Finesse, go to:

http://www.cisco.com/en/US/products/ps11324/prod_literature.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 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 Personal Communicator

Cisco Unified Personal Communicator 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 (Cisco Unity or Unity Connection), video (Cisco TelePresence), web conferencing (Cisco Unified MeetingPlace), call management (Unified CM), directories (LDAP), and presence and instant messaging (Unified 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.

Cisco Unified Personal Communicator operates in Desk Phone (CTI control of the user's desk phone for Click to Call) and Soft Phone (software client operation) modes, and is supported on Apple Macintosh and Microsoft Windows platforms.

For additional information, go to:

http://www.cisco.com/en/US/products/ps6844/tsd_products_support_series_home.html

Cisco Unified Mobility

Cisco Unified Mobility gives users the ability to redirect incoming IP calls from Cisco Unified Communications Manager to different designated phones, such as cellular phones. Users can also transition active calls between their Cisco desktop and phone without interruption.

Cisco Unified Mobility includes these features:

- Streamlined communications, giving callers one number to dial, and by redirecting incoming calls to multiple phones
- Active calls can move between the Cisco desktop and mobile phone to take advantage of the best available resource
- Simplified message management, by directing unanswered calls to a Cisco Unity or Cisco Unity Connection account
- Personalized access lists that determine which business calls get extended to alternate phone numbers, and at what point that occurs

For more information about Cisco Unified Mobility, go to:

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

Cisco Jabber

Cisco Jabber helps enterprise users consolidate presence, instant messaging, voice and video, voice messaging, desktop sharing, and conferencing. Cisco Jabber provides integration across devices, including PCs, Macs, tablets, and smart phones.

Cisco Jabber client software works in conjunction with Cisco Unified Communications Manager to provide users with a unified client they can deploy across on-premise and cloud-based options.

Cisco Jabber clients include the following:

- Cisco Jabber for Mac

  Cisco Jabber for Mac provides presence, instant messaging, voice, visual voicemail, desktop sharing, and conferencing capabilities in the familiar Mac experience.

- Cisco Jabber for Android

  Cisco Jabber for Android provides Android users with enterprise voice over IP (VoIP) calling and access to the corporate directory while connected to the corporate network over Wi-Fi.

- Cisco Mobile for iPhone

  Cisco Mobile for iPhone provides enterprise voice over IP (VoIP) calling with handoff to GSM/cellular network and extensive mid call features such as Hold, Conference, Transfer and Swap. Business visual voicemail and access to corporate directory are supported on iPhone, iPad, and iPod touch.

- Cisco Mobile for Nokia

  Cisco Mobile for Nokia provides enterprise IM and Presence for on premise or SaaS deployments. It includes enterprise voice over IP (VoIP) calling with automatic handoff to cellular network and extensive mid-call features for Symbian smartphones. Corporate directory access and one click access to voicemail are included.

- Cisco Jabber IM for BlackBerry

  Cisco Jabber IM for BlackBerry provides basic IM functionality for BlackBerry devices while delivering enterprise grade features such as Single Sign On, Proxyless Architecture, and SSL encryption.

For additional information, go to:

http://www.cisco.com/en/US/prod/voicesw/ps6789/jabber_uc_apps.html

Cisco Unified Communications Integration™ for Microsoft Lync

Cisco Unified Communications Integration™ for Microsoft Lync provides seamless collaboration with Cisco Unified Communications and Microsoft instant messaging (IM) and Presence capabilities.

It extends proven Cisco Unified Communications services to Microsoft Lync with a single easy-to-manage communications platform. This provides interoperability with Microsoft Lync Server 2010 and Microsoft Lync. Cisco UC Integration™ for Microsoft Lync uses the Client Services Framework (CSF) and incorporates it into Microsoft Lync. This integration allows for the use of audio telephony of existing Cisco Unified Communications Manager endpoints, acting both as a softphone (softphone mode) and controlling a Cisco Unified IP Phone (desk phone mode).

This integration for Microsoft Lync leverages a common unified client services framework to:

- Increase productivity—Instantly connect with colleagues, partners, and customers from anywhere and have a business-class communication experience with an integrated Cisco IP softphone.
- Streamline communications—View telephony presence status, access corporate voicemail and communications history, or simply click to call through Cisco Unified IP Phone directly from your desktop.
- Enhance collaboration—Initiate multiparty conference calls and quickly add more participants as needed.
- Reduced complexity—Extend proven attributes of Cisco Unified Communications Manager directly to your desktop with an easy-to-deploy integration and benefit from reduced management complexity of a single call control architecture.
- Protect investments—Make an immediate business impact with interoperable Cisco Unified Communications while protecting your investments in existing desktop applications.

For additional information, go to:

http://www.cisco.com/en/US/products/ps10317/index.html

Cisco UC Integration™ for Cisco WebEx Connect

Cisco UC Integration(TM) for Cisco WebEx Connect is a collaborative software-as-a-service (SaaS) platform that enables developers, partners, and customers to create powerful collaborative business solutions that can extend their reach through collaborative solutions. Cisco WebEx Connect provides an open and extensible collaboration platform for enforcing enterprise-class security, scalability, performance, and availability, while delivering transparent communication with the Cisco Unified Communications solution. Cisco WebEx Connect contains two main components, the Cisco WebEx Connect Client and the Cisco WebEx Connect Platform.

For additional information, go to:

http://www.cisco.com/en/US/products/ps10627/tsd_products_support_series_home.html

Cisco Unified Communications Widgets

Cisco Unified Communications Widgets applications deliver a productive and personalized user experience with Cisco Unified Communications applications and Cisco Unified IP Phones. These free-to-download and easy-to-add widgets streamline business communications and provide a tailored and familiar communications experience.

Cisco Unified Communications Widgets include the following:

- The Click to Call Widget is a Cisco Unified Communications application for PCs that lets users quickly place calls from desktop productivity applications or web browsers. Users can simply highlight and click on a phone number to make a call.
- The Visual Voicemail Widget for Cisco Unified IP Phones displays all Cisco Unity and Cisco Unity Connection voice messages on the phone display. Caller name, time of message, message length, and urgency are prominently displayed. Users can view, play, save, respond to, and delete messages without having to dial in to enterprise voicemail.

For additional information, go to:

http://www.cisco.com/en/US/products/ps9829/index.html

Cisco Unified Video Advantage

Cisco Unified Video Advantage brings video telephony functionality to select Cisco Unified IP Phones and to the Cisco IP Communicator softphone application. Users make and receive calls using the familiar phone interface, with the video component displayed on user PCs without additional user action required. Enterprises can leverage their existing IP networks and desktop phones to extend video calling to everyone in the organization.

For additional information, go to:

http://www.cisco.com/en/US/products/sw/voicesw/ps5662/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 Untiy Connection integrates with Cisco Unified CallManger, Cisco Unified CallManger Express, and various legacy PBX models (using the PIMG) to support a variety of deployment models and configurations.

Key features of Cisco Unity Connection include:

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

For additional information, go to:

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

Cisco Survivable Remote Site Voicemail

Cisco Survivable Remote Site Voicemail (SRSV) provides backup voicemail service in the centralized messaging and centralized call processing deployment. SRSV utilizes Cisco Unity Express in the branch location to provide backup voicemail service for Cisco Unity Connection located in the headquarters when the connection between sites is unavailable. (See Figure 23-2.) During normal operation, Cisco

Unified Messaging Gateway in the headquarters retrieves the configurations (for example, SRST phones, user, and mailbox information) from Cisco Unified CM and Cisco Unity Connection to provision and update the mailboxes in Cisco Unity Express SRSV based on a configured schedule. Cisco Unity Express SRSV is active when only SRST is activated, and it remains idle otherwise. When the network connection between sites is restored, Cisco Unity Express SRSV uploads all messages (new, saved, deleted, and so forth) to Cisco Unity Connection.

SRSV uses bandwidth from the WAN link during the following activities:

- Configuration uploads from Unified CM and Cisco Unity Connection to Cisco Unity Express SRSV
- Voice message uploads from Cisco Unity Express SRSV to Cisco Unity Connection when the WAN link is restored.

Survivable Remote Site Telephony (SRST) and Cisco Unity Express SRSV are one logical unit, with Cisco Unity Express SRSV installed in the SRST router.

For additional information, go to:

http://www.cisco.com/en/US/products/ps10769/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 centralized voicemail solution for up to 10 Cisco Unified Communications Manager Express sites and 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:

- Interactive Voice Response (IVR) – integrates your automated attendant into the company database.
- Paging and Announcement system – provides live and scheduled paging to Cisco IP Phones and overhead speakers. Integrates with legacy paging systems.
- TimeCardView – integrated time and attendance management system for the branch office. Syncronize your payroll data to Intuit QuickBooks
- 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 Unified SIP Proxy

Cisco Unified SIP Proxy is a high-performance, highly available Session Initiation Protocol (SIP) proxy server for centralized routing and SIP signaling normalization. By forwarding requests between call-control domains, Cisco Unified SIP Proxy provides the means for routing sessions within enterprise. The Cisco Unified SIP Proxy application is delivered in Network Module and Service Module form factors on Cisco 2900, 3800, 3900, and 3900E Series Integrated Services Routers

The Cisco Unified SIP Proxy brings the following benefits to a network using Unified communications Manager SIP trunks:

- Aggregation and routing—The Unified SIP Proxy is capable of connecting several SIP servers to each other without each of the servers connecting to every other one in a full-mesh configuration
- Scalability—The Unified SIP Proxy can be used to terminate calls to and from the enterprise and IP-PSTN service providers. The proxy, in turn, distributes the calls across a pool of Unified Border Elements. More Unified Border Elements may be added to increase capacity.
- Availability and load balancing—The Unified SIP Proxy distributes calls over the pool of available Unified Border Elements and monitors the status of each Unified Border Element to ensure reliable call completion.
- Message normalization—The Unified SIP Proxy serves to hide differences in SIP protocol messaging by providing the means to manipulate headers and contents of the messages as they pass through the Unified SIP Proxy.

For additional information, go to:

http://www.cisco.com/en/US/products/ps10140/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 includes the following key features:

- Centrally manage a network of branch-office telephony sites and automatically synchronize them with the central call control with Enhanced Survivable Remote Site Telephony (E-SRST)
- Automatically configures SRST routers for use with centralized Cisco Unified Communications Manager.

- Centrally manage a network of survivable branch-office voicemail sites and automatically synchronize the messages with the central Cisco Unity Connection with Survivable Remote Site Voicemail (SRSV).
- 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/prod/collateral/voicesw/ps6789/ps5745/ps8605/product_data_sheet0900aecd806a8a3c.html

Cisco VG200 Series Gateways

The Cisco Unified Communications System supports the following VG200 Series Gateways:

- Cisco VG224 Analog Voice Gateway
- Cisco VG204 Analog Voice Gateway
- Cisco VG202 Analog Voice Gateway

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

The Cisco VG204 Analog Voice Gateway combines granular RJ11 analog interfaces with Cisco IOS Software manageability to deliver a platform designed to maximize the functionality of existing distributed analog equipment in a Cisco Unified Communications system deployment. It offers the following key benefits:

- Low-density four-port gateway for analog phones, fax machines, modems, and speakerphones
- Enhances an enterprise voice system architecture that is based on a Cisco Integrated Services Router, Cisco modular access router or a Cisco VG224 in a Cisco Unified Communications Manager or Cisco Unified Communications Manager Express deployment
- Compact, fanless, desktop form-factor chassis that is wall-mountable

The Cisco VG202 Analog Voice Gateway combines granular RJ11 analog interfaces with Cisco IOS Software manageability to deliver a platform designed to maximize the functionality of existing distributed analog equipment in a Cisco Unified Communications system deployment. It offers the following key benefits:

- Low-density two-port gateway for analog phones, fax machines, modems, and speakerphones
- Enhances an enterprise voice system architecture that is based on a Cisco Integrated Services Router, Cisco modular access router, or a Cisco VG224 in a Cisco Unified Communications Manager or Cisco Unified Communications Manager Express deployment.
- Compact, fanless, desktop form-factor chassis that is wall-mountable

For additional information, go to:

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

Internet Protocol Version 6 (IPv6)

Cisco has taken a leading role in the definition and implementation of the IPv6 architecture within the Internet Engineering Task Force (IETF) and continues to lead the industry in IPv6 development and standardization.

The deployment of IPv6 is primarily driven by IPv4 address space exhaustion. As the worldwide usage of IP networks increases, the number of applications, devices, and services requiring IP addresses is rapidly increasing. Current estimates by the Internet Assigned Numbers Authority (IANA) and Regional Internet Registries (such as ARIN, LACNIC, and APNIC) indicate that their pools of un-allocated IPv4 addresses will be exhausted sometime between Q4 2011 and Q1 2012.

Because the current IPv4 address space is unable to satisfy the potential huge increase in the number of users and the geographical needs of the Internet expansion, many companies are either migrating to or planning their migration to IPv6, which offers a virtually unlimited supply of IP addresses.

The process of transforming the Internet from IPv4 to IPv6 is likely to take several years. During this period, IPv4 will co-exist with and then gradually be replaced by IPv6.

It is recommended that you deploy IPv6 in a dual-stack Cisco Unified Communications Manager (Unified CM) cluster with approved dual-stack devices (phones, gateways, and so forth). This approach is recommended to avoid IPv6-only deployments, which are not currently supported in production environments. Single-site and multiple-site distributed call processing deployments are supported, but multiple-site deployments with centralized call processing are not supported.

An IPv6 address consists of 8 sets of 16-bit hexadecimal values separated by colons (:), totaling 128 bits in length. For example:

2001:0db8:1234:5678:9abc:def0:1234:5678

Leading zeros can be omitted, and consecutive zeros in contiguous blocks can be represented by a double colon (::). Double colons can appear only once in the address. For example:

2001:0db8:0000:130F:0000:0000:087C:140B can be abbreviated as

2001:0db8:0:130F::87C:140B

As with the IPv4 Classless Inter-Domain Routing (CIDR) network prefix representation (such as 10.1.1.0/24), an IPv6 address network prefix is represented the same way:

2001:db8:12::/64

The following Cisco Unified Communications products support IPv6:

- Cisco Unified Communications Manager—All Cisco Media Convergence Server (MCS) platforms
- Cisco IP Phones
  - Third generation Cisco IP Phones running SCCP only—Cisco 7906G, Cisco 7911G, Cisco 7931G, Cisco 7941G, Cisco 7941GE Cisco 7942G, Cisco 7945G, Cisco 7961G, Cisco 7961GE, Cisco 7962G Cisco 7965G, Cisco 7970G, Cisco 7971G-GE, Cisco 7975G
- Gateways
  - SIP gateways (2800, 2900, 3800, 3900, 3900E Series; Cisco AS5400XM; Cisco AS5350XM)
  - Cisco VG224, VG204, VG202 SCCP Analog Gateways
  - SCCP FXS ports on Cisco ISR 2800, 2900, 3800, 3900, 3900E Series Routers

- Cisco IOS MTPs for IPv4-to-IPv6 RTP media conversion
- Cisco Unified CM SIP Trunks
- Applications
  - Cisco Unified CM CTI (IPv6 aware)
  - Cisco Unified CM AXL/SOAP interface (IPv6 aware)
  - Cisco Unified CM SNMP (IPv6 aware)

For more information on the implementation and deployment of IPv6 architecture see the Cisco Unified Communications SRND, available at:

http://www.cisco.com/go/designzone

Cisco Adaptive Security Appliances

Cisco ASA 5500 Series Adaptive Security Appliances provide intelligent threat defense and highly secure communications services. These solutions help organizations lower their deployment and operational costs while delivering comprehensive network security for networks of all sizes.

Cisco Adaptive Security Appliances integrate:

- Stop attacks before they penetrate the network perimeter
- Protect resources and data, as well as voice, video, and multimedia traffic
- Control network and application activity
- Reduce deployment and operational costs
- Adaptable architecture for rapid and customized security services deployment
- Advanced intrusion prevention services that defend against a broad range of threats
- Highly secure remote access and unified communications to enhance mobility, collaboration, and productivity

For more information about these components, go to:

http://www.cisco.com/en/US/products/ps6120/index.html

Management and Serviceability Components

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

- 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 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 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 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 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

Cisco Design Tools

The Cisco Unified Communications Solution includes the following Design Tools components.

Note These tools are available to Cisco and Unified Communications specialized partners only.

- Unified Communications Sizing Tool—a web-based tool that assists users with hardware sizing of large or complex Cisco Unified Communications solutions by calculating the call processing requirements for products that have a major impact on performance and scalability.

  With the Cisco Unified Communications Sizing Tool, system engineers with Cisco Unified Communications solution experience or individuals with equivalent abilities can design and model solutions for existing and prospective customers. The tool requires various types of information to calculate the minimum size and type of devices required for a solution, such as the type and quantity of IP phones, gateways, and media resources. For most device types, the tool also requires the average number of call attempts per hour per device during the busy hour (known as busy hour call average or BHCA) and the average utilization time. The resulting calculations produced by the tool can be saved, copied, and sent to other users.
- Quote Builder—a solutions quoting application for Cisco Unified Communications products.

With Quote Builder, users can build a system quote with design documents to aid in the implementation of the solution. Quote Builder also validates designs for common deployments. Quote Builder generates a bill of materials, a network diagram, and design guides for deployment. To access Quote Builder, go to the following URL

- Solution Expert—a web-based tool that assists in the design, configuration, quoting, and ordering of Cisco Unified Communications products.

  With Solution Expert, users can generate a recommended solution based on their requirements. Users can modify the recommended configuration if desired. Solution Expert validates any changes when it presents the new solution. Solution Expert also generates a bill of materials with list pricing, a Visio diagram, and other design documentation.

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

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 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							◆		
Cisco Unified SIP Proxy							◆		
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 Programing Interfaces

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

1. Supported in Windows platforms
2. Support between Video Integration and Video Admin

3. Cisco Unified Meeting Place supports XML between Video Integration and Video Admin and between Video Admin and MCU
4. Cisco Unity Express is not fully IMAP compliant. IMAP integration is supported only for Outlook, Outlook Express, Lotus Notes and Entourage 2008

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

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	• 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	• Where are the locations that the system will serve? • How should the system components be distributed across sites?
Availability	• Is availability needed 24 hours a day or only during normal business hours? • What is the required recovery time after failure?
Capacity	• How many users must be supported? • How many simultaneous calls must be supported?
Integration	• 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	• 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 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	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. SCCP is supported by all Cisco IP Phones, by Cisco Unified Video Advantage, by many third-party video endpoints, and by select Cisco gateways.
MGCP	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. MGCP provides easier configuration and centralized management. It is supported by most Cisco gateways.

Protocol	Description
SIP	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). 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.
H.323	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. 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.

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	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). G.711 is a low-compression, medium-complexity codec.
G.722	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. G.722 is similar to G.711 in compression and complexity, but provides higher quality audio.
G.722.1	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. G.722.1 is a high-compression, low-complexity codec.It provides better quality than G.722 at lower bit-rates.
G.723.1	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. G.723.1 is a high-compression, high-complexity codec. However, the quality is slightly lower than that of G.711.
G.726	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. 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.
G.728	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. G.728 is a high-compression, high-complexity codec.
G.729a	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. 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.
iLBC (internet Low Bitrate Codec)	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. 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.

Codec	Description
AAC (Advanced Audio Codec)	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.
L16	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.
GSM-FR (Global System for Mobile Communications-Full Rate)	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. GSM-FR is a medium-complexity codec.
GSM-EFR (Enhanced Full Rate)	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. GSM-EFR is a high-complexity codec and provides better sound quality than GSM-FR.
QCELP (Qualcomm Code Excited Linear Prediction)	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. QCELP is a high-complexity codec.
H.261	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. 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.
H.263	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 x 96), 4CIF (704 x 576), and 16CIF (1408 x 1152) resolutions. 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.
H.264	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. 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.

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, guarenteed 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.

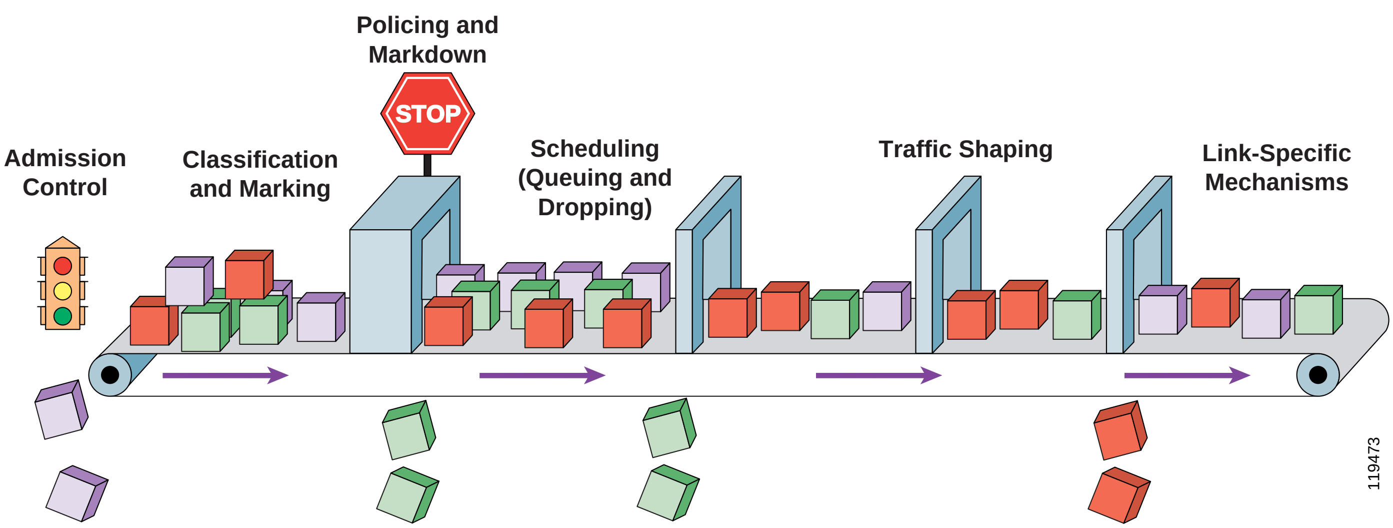
  Each end station in a VoIP or Video over IP conversation has a jitter buffer. Jitter buffers are used to smooth out changes in arrival times of data packets containing voice. A jitter buffer is dynamic and adaptive, and can adjust for up to a 30 ms average change in arrival times of packets. If you have 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 toolset 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 queueing 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

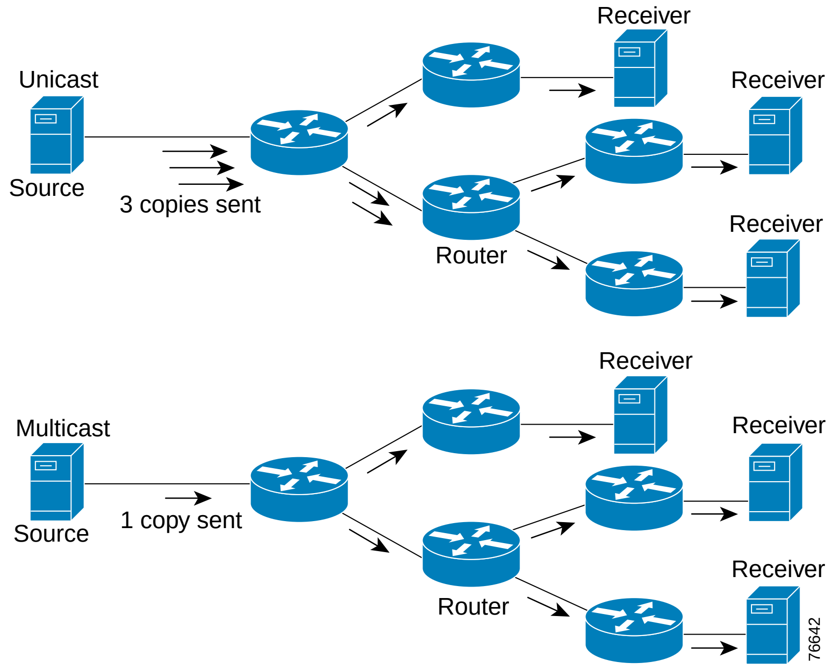
IP Multicast

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



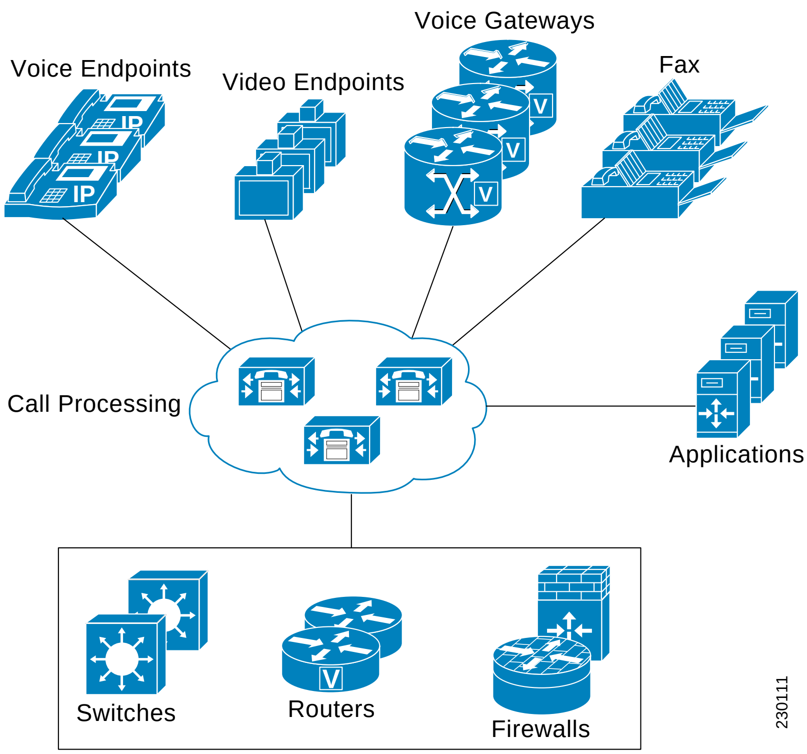
Multicast packets are replicated in the network by Cisco routers and switches enabled with Protocol Independent Multicast (PIM) and other supporting multicast protocols. These routers create "distribution trees," which control the path that IP Multicast traffic takes through the network in order to deliver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

For more information about IP multicast, see the Cisco Avvid 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 an 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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