Cisco Preferred Architecture for Enterprise Collaboration 11.6

Design Overview

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Preface

Cisco Preferred Architectures provide recommended deployment models for specific market segments based on common use cases. They incorporate a subset of products from the Cisco Collaboration portfolio that is best suited for the targeted market segment and defined use cases. These deployment models are prescriptive, out-of-the-box, and built to scale with an organization as its business needs change. This prescriptive approach simplifies the integration of multiple system-level components and enables an organization to select the deployment model that best addresses its business needs.

Documentation for Cisco Preferred Architectures

- Cisco Preferred Architecture (PA) Design Overview guides help customers and sales teams select the appropriate architecture based on an organization's business requirements; understand the products that are used within the architecture; and obtain general design best practices. These guides support pre-sales processes.
- Preferred Architecture Cisco Validated Design (CVD) guides provide details for deploying components within the Cisco Preferred Architectures. These guides support planning, deployment, and implementation (PDI).
- Preferred Architecture Application Cisco Validated Design (CVD) guides provide an application solution to the foundational Enterprise Preferred Architecture. These guides support planning, deployment, and implementation (PDI).
- Cisco Solution Reference Network Design (SRND) guide provides detailed design options for Cisco Collaboration. This guide should be referenced when design requirements are outside the scope of Cisco Preferred Architectures.

Figure 1 illustrates how to use the guides. As mentioned, this overview is used for the pre-sales process to explain the products and components, while the CVDs are used in the post-sales process for further design, deployment, and implementation. The set of Application CVDs covers optional applications that can be deployed on top of the foundational Preferred Architecture.

Figure 1 Preferred Architecture Documentation Structure

About This Guide

The Cisco Preferred Architecture for Enterprise Collaboration is for:

- Sales teams that design and sell collaboration solutions
- Customers and sales teams who want to understand the overall collaboration architecture, its components, and general design best practices

Readers of this guide should have a general knowledge of Cisco Voice, Video, and Collaboration products and a basic understanding of how to deploy these products.

This guide simplifies the design and sales process by:

- Recommending products in the Cisco Collaboration portfolio that are built for the enterprise and that provide appropriate feature sets for this market
- Detailing a collaboration architecture and identifying general best practices for deploying in enterprise organizations

For detailed information about configuring, deploying, and implementing this architecture, consult the related CVD documents on the Design Zone for Collaboration.

Introduction

In recent years, many new collaborative tools have been introduced to the market, enabling organizations to extend collaboration outside the walls of their businesses. Providing access to collaborative tools for employees outside the office is no longer a luxury; it is mandatory for businesses to stay relevant in today's market. Today's users expect immediate access to these tools from a wide variety of portable and mobile devices. Many of these same tools can be extended to customers and partners, helping strengthen these relationships.

Organizations realize the added value that collaboration applications bring to their businesses through increased employee productivity and enhanced customer relationships. Not long ago, interoperability among collaboration applications was sparse, and applications were difficult to deploy and use. Since then, significant advances have been made in the collaboration space, simplifying deployment, improving interoperability, and enhancing the overall user experience. Additionally, individuals have adopted a wide variety of smart phones, social media, and collaboration applications in their personal lives.

Organizations can now feel comfortable providing collaboration applications that employees will quickly adopt and that provide maximum value. These new collaboration tools enhance an organization's overall business processes, make its employees more productive, and open the door to new and innovative ways for communicating with business partners and customers. Today's collaboration solutions offer organizations the ability to integrate video, audio, and web participants into a single, unified meeting experience.

Technology Use Cases

Organizations want to streamline their business processes, optimize employee productivity, and enhance relationships with partners and customers. The Cisco Preferred Architecture (PA) for Enterprise Collaboration delivers capabilities that enable organizations to realize immediate gains in productivity and enhanced relationships. Additionally, the following technology use cases offer organizations opportunities to develop new, advanced business processes that deliver even more value in these areas:

- **Consolidate Communications Infrastructure —** Bring together voice, video, and data into a single IP network to simplify management and support effective communications.
- **Incorporate Video into Meetings —** Improve communications, relationships, and productivity by making it easier to meet face-to-face over distance.
- **Extend Telephony with Video —** Facilitate face-to-face video communications directly from end-user phones or softphone applications.
- **Support Teleworkers and Branch Offices —** Let employees work from multiple locations, whether satellite offices, home offices, or when traveling.
- **Collaborate with External Organizations —** Easily share information, interact in real time, and communicate using technologies beyond email and telephone.
- **Create Flexible Work Areas and Office Spaces —** Scale office space and create work areas that foster employee inclusiveness, collaboration, innovation, and teamwork.
- **Deploy a Unified Communications Architecture —** Provide the entire global organization with a single communications tool set for all users.

Information about Cisco Collaboration Technologies and use cases is available on Cisco.com.

Architectural Overview

The Cisco PA for Enterprise Collaboration provides end-to-end collaboration targeted for deployments larger than 1,000 users. This architecture incorporates high availability for critical applications. The consistent user experience provided by the overall architecture facilitates quick user adoption. Additionally, the architecture supports an advanced set of collaboration services that extend to mobile workers, partners, and customers through the following key services:

- Voice communications
- Instant messaging and presence
- High-definition video and content sharing
- Rich media conferencing
- Enablement of mobile and remote workers
- Business-to-business voice and video communications
- Unified voice messaging

Because of the adaptable nature of Cisco endpoints and their support for IP networks, this architecture enables an organization to use its current data network to support both voice and video calls. The preferred architecture provides a holistic approach to bandwidth management, incorporating an end-to-end QoS architecture, call admission control, and video rate adaptation and resiliency mechanisms to ensure the best possible user experience for deploying pervasive video over managed and unmanaged networks.

The Cisco PA for Enterprise Collaboration, shown in Figure 2, provides highly available and secure centralized services. These services extend easily to remote offices and mobile workers, providing availability of critical services even if communication to headquarters is lost. Centralized services also simplify management and administration of an organization's collaboration deployment.

Figure 2 Cisco Preferred Architecture for Enterprise Collaboration

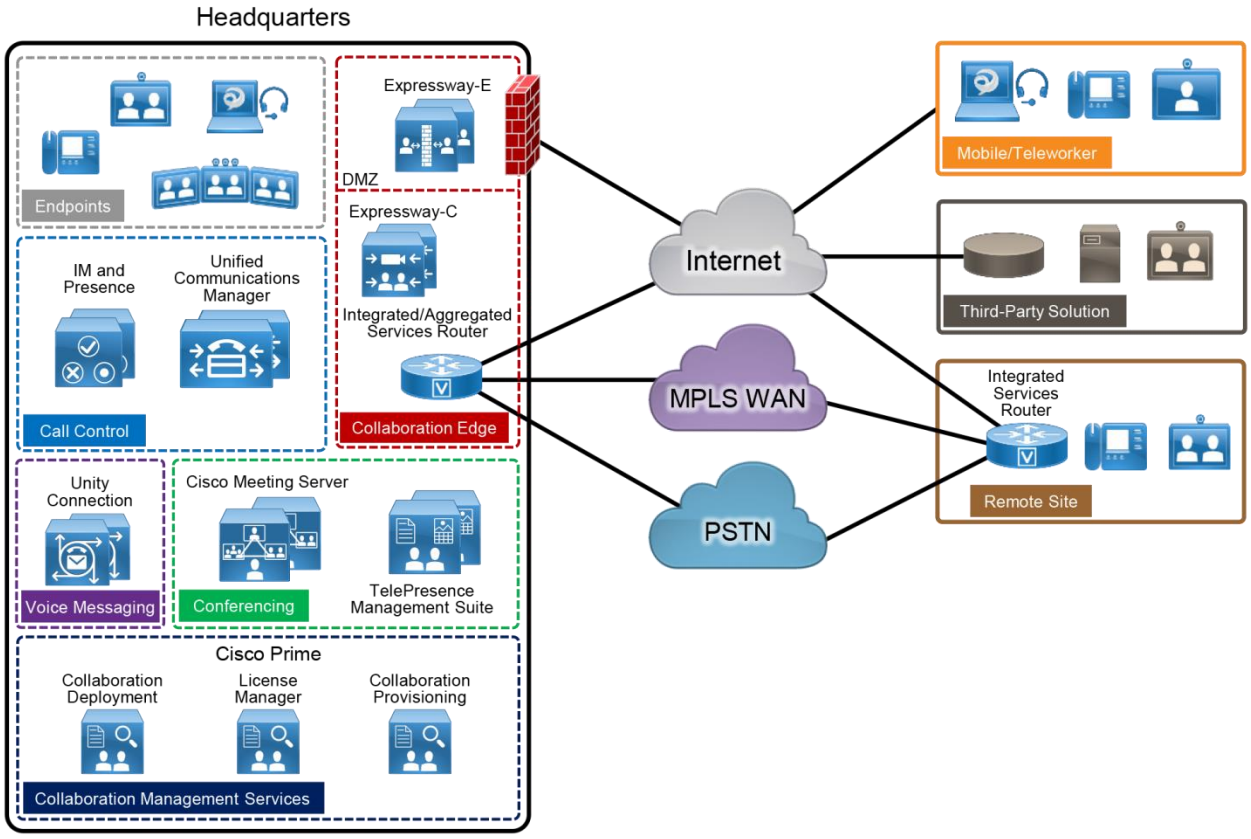


Table 1 lists the products in this architecture. For simplicity, products are grouped into modules to help categorize and define their roles. The content in this guide is organized in the same modules.

Table 1 Components of the Cisco Preferred Architecture for Enterprise Collaboration

Module	Component	Description
Call Control	Cisco Unified Communications Manager (Unified CM)	Provides endpoint registration, call processing, and media resource management
	Cisco Unified Communications Manager IM and Presence Service	Provides instant messaging and presence services
	Cisco Integrated Services Router (ISR)	Provides Survivable Remote Site Telephony (SRST) functionality
Endpoints	Cisco IP Phones, Cisco TelePresence video endpoints, and Cisco Jabber	Enable real-time voice, video, and instant messaging communications for users
Conferencing	Cisco Meeting Server	Provides audio and video conferencing capabilities as well as conference resource management
	Cisco TelePresence Management Suite and Extensions	Provides scheduling, web conferencing integration, and other advanced video features
Collaboration Edge	Cisco Expressway-C	Enables interoperability with third-party systems and firewall traversal
	Cisco Expressway-E	Supports remote endpoint registration to Cisco Unified CM and enables business-to-business communications
	Cisco ISR and ASR	Provides either public switched telephone network (PSTN) or Cisco Unified Border Element (CUBE) connectivity

Module	Component	Description
Voice Messaging	Cisco Unity Connection	Provides unified messaging and voicemail services
Collaboration Management Services	Cisco Prime Collaboration Deployment	Assists in the management of Unified Communications applications. It allows the user to perform tasks such as migration of older software versions of clusters to new virtual machines, fresh installs, and upgrades on existing clusters.
	Cisco Prime License Manager	Provides simplified, enterprise-wide management of user-based licensing, including license fulfillment.
	Cisco Prime Collaboration Provisioning	Enables rapid configuration of collaboration systems by providing a centralized template-based console for device provisioning and simplified moves, adds, and changes.

Virtualization and Core Applications

Virtualizing multiple applications and consolidating them on physical servers lowers costs, minimizes rack space, lowers power requirements, and simplifies deployment and management. Virtualization also accommodates redeploying hardware and scaling software applications as organizational needs change.

Cisco Business Edition 7000

The Cisco Business Edition (BE) 7000 serves organizations with 1,000 or more users, and it is the foundation of the Cisco PA for Enterprise Collaboration. The Cisco BE7000 is built on a Cisco Unified Computing System (UCS) that ships ready-for-use with a preinstalled virtualization hypervisor and application installation files. The Cisco BE7000 solution offers premium voice, video, messaging, instant messaging and presence, and contact center features on a single, integrated platform. For more information about the Cisco BE7000, see the data sheet.

Core Applications

In the Cisco PA for Enterprise Collaboration, the following applications are deployed on multiple UCS servers to provide hardware and software redundancy:

- Cisco Unified Communications Manager
- Cisco Unified Communications Manager IM and Presence Service
- Cisco Unity Connection
- Cisco Expressway, consisting of Expressway-C and Expressway-E
- Cisco Meeting Server
- Cisco TelePresence Management Suite and Extensions
- Cisco Prime Collaboration Deployment
- Cisco Prime License Manager
- Cisco Prime Collaboration Provisioning

We recommend always deploying redundant components and configurations to provide the highest availability for critical business applications.

High Availability

The Cisco PA for Enterprise Collaboration provides high availability for all deployed applications by means of the underlying clustering mechanism present in all Cisco Unified Communications applications.

Clustering replicates the administration and configuration of deployed applications to backup instances of those applications. If an instance of an application fails, Cisco Unified Communications services – such as endpoint registration, call processing, messaging, business-to-business communication, and many others – continue to operate on the remaining instance(s) of the application. This failover process is transparent to the users. In addition to clustering, the Cisco PA for Enterprise Collaboration provides high availability through the use of redundant power supplies, network connectivity, and disk arrays.

Sizing Considerations

Sizing a deployment can become complex for large enterprises with sophisticated requirements. The Preferred Architecture for Enterprise Collaboration Cisco Validated Design (CVD) Guide presents some examples that simplify the sizing process. In addition, Cisco provides several tools to assist with sizing a deployment. The sizing tools are available to Cisco certified partners at http://cucst.cloudapps.cisco.com. If you do not have access to the sizing tools, contact your Cisco account representative or Cisco certified partner to obtain system sizing information.

Licensing

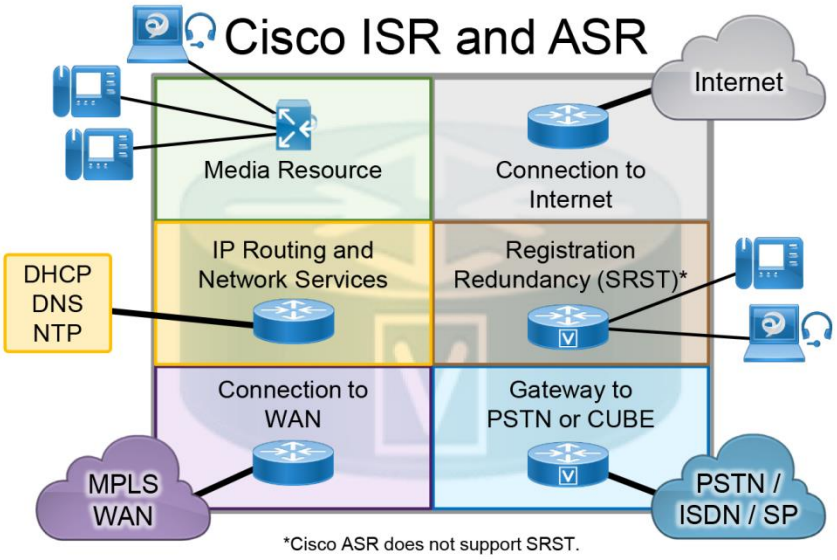
Details about the individual licenses for the endpoints and infrastructure components in the Cisco Preferred Architecture for Enterprise Collaboration are beyond the scope of this document. Information about Cisco Unified Communications licensing is available on the License Administration Portal at https://tools.cisco.com/SWIFT/LicensingUI/Home. The License Administration Portal also provides instructions and tools to assist with license administration.

Cisco Integrated Services and Aggregation Services Routers

The Cisco Integrated Services Router (ISR) and Aggregation Services Router (ASR) provide Wide Area Network (WAN) and Cisco Unified Communications services in a single platform. In the Cisco PA for Enterprise Collaboration, the Cisco ISR and ASR can provide the following functions (Figure 3):

- External connectivity to the Internet
- IP routing and remote-site network services such as DHCP, DNS, NTP, and others
- Cisco Unified Survivable Remote Site Telephony (SRST) to service calls during WAN failures
- Voice gateway to the Public Switched Telephone Network (PSTN), or Cisco Unified Border Element (CUBE) for Session Initiation Protocol (SIP) trunks
- Integrated data and voice connectivity to service providers
- Multiprotocol Label Switching (MPLS) WAN connectivity for the organization’s network
- Media resources for Cisco Unified Communications Manager

Figure 3 Cisco ISR and ASR Functions



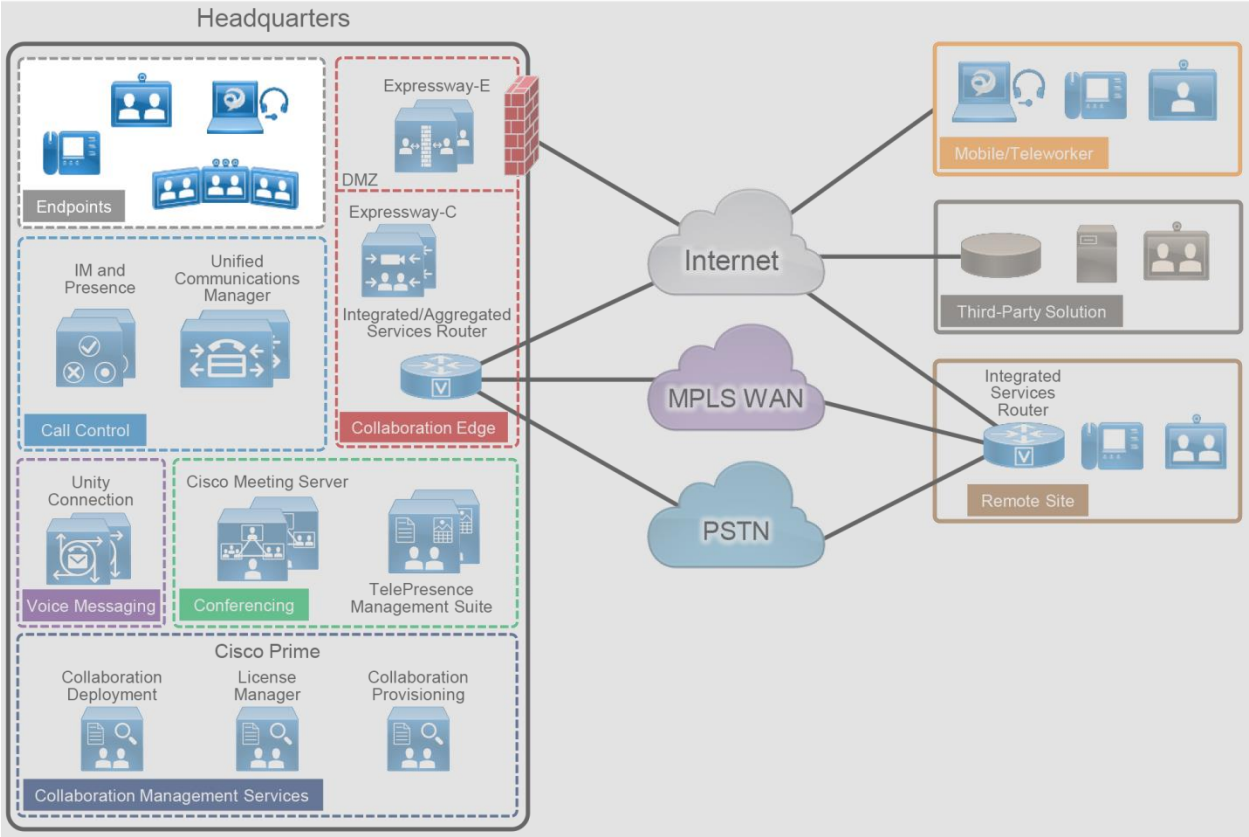
The Cisco ISR and ASR have additional slots that support add-on modules such as wireless controllers and VMware ESXi servers. Deployments can use various Cisco ISR and ASR models to support different features, to scale, and to accommodate additional services. Their modular design enables the Cisco ISR and ASR to be deployed at headquarters, remote locations, or branch locations. For more information about these routers, see the Cisco ISR and Cisco ASR data sheets.

Endpoints

Cisco Collaboration endpoints provide a wide range of features, functionality, and user experiences. Because Cisco endpoints range from low-cost, single-line phones and soft clients to three-screen Cisco TelePresence endpoints, an organization can deploy the right variety of endpoints to meet users' needs (Figure 4). Additionally, these devices enable users to access multiple communication services such as:

- Voice calls
- Video calls
- Conferencing
- Voicemail
- Presence
- Instant messages
- Desktop sharing

Figure 4 Architecture for Endpoints



Recommended Deployment

Cisco Unified Communications Manager (Unified CM) is the call control server for the Cisco PA for Enterprise Collaboration. Cisco IP Phones, Jabber clients, and TelePresence video endpoints use SIP to register directly to Cisco Unified CM. The Unified CM cluster's failover mechanism provides endpoint registration redundancy. If a WAN failure occurs and endpoints at remote locations cannot register to Unified CM, they use SRST functionality for local and PSTN calls, but some services such as voicemail and presence might not be available.

We recommend the endpoints listed in the following tables because they provide optimal features for this design. Cisco has a range of endpoints with various features and functionality that an organization can also use to address its business needs.

Table 2 Cisco IP Phones

Product	Description
Cisco IP Phone 7811	Public space, single-line phone
Cisco IP Phone 8800 Series	General office use, multiple-line phone
Cisco IP Phone 8831	IP conference phone

Table 3 Cisco TelePresence and Video Endpoints

Product	Description
Cisco DX70 and DX80[1]	Personal TelePresence endpoint for the desktop
Cisco MX Series	TelePresence multipurpose room endpoint
Cisco SX Series	Integrator series TelePresence endpoint
Cisco IX Series	Immersive TelePresence room system

1. DX70 and DX80 endpoints run CE firmware.

Table 4 Cisco Jabber

Product	Description
Mobile: Jabber for Android Jabber for iPhone and iPad **Desktop:** Jabber for Mac Jabber for Windows	Soft client with integrated voice, video, voicemail, instant messaging, and presence functionality for mobile devices and personal computers

Table 5 Comparison of Endpoint Features and Capabilities

Product(s)	Audio	Video	Content Sharing	Unified CM High Availability	Mobile and Remote Access	Audio SRST
IP Phone 7811	Y	N	N	Y	Y	Y
IP Phone 8800 Series	Y	Y[2]	N	Y	Y	Y
IP Phone 8831	Y	N	N	Y	N	Y
DX70 and DX80	Y	Y	Y	Y	Y	N
MX Series	Y	Y	Y	Y	Y	N
SX Series	Y	Y	Y	Y	Y	N
IX Series	Y	Y	Y	Y	N	N
Jabber Mobile	Y	Y	N	Y	Y	Y
Jabber Desktop	Y	Y	Y	Y	Y	Y

2. Only the IP Phones 8845 and 8865 support video.

Call Control

Call control is the core element for any communications deployment. It provides endpoint registration, call processing, and call admission control. Call control design considerations include the enterprise dial plan, endpoint addressing scheme, calling party presentation, call admission control, codec selection, PSTN connectivity, and general trunking requirements, as well as other factors.

Cisco Unified Communications Manager (Unified CM) provides a common call control platform for all Cisco Collaboration deployments (Figure 5). Having a highly available and common call control component for a communications infrastructure is crucial to provide consistent services for all devices and communication types and to preserve a uniform dial plan and a consistent feature set across the enterprise.

Adding the IM and Presence Service to a Cisco Unified CM deployment provides instant messaging, network-based presence, and federation for third-party chat servers, and it enables the use of Cisco Jabber for instant messaging, presence, and audio and video communications.

Figure 5 Architecture for Call Control

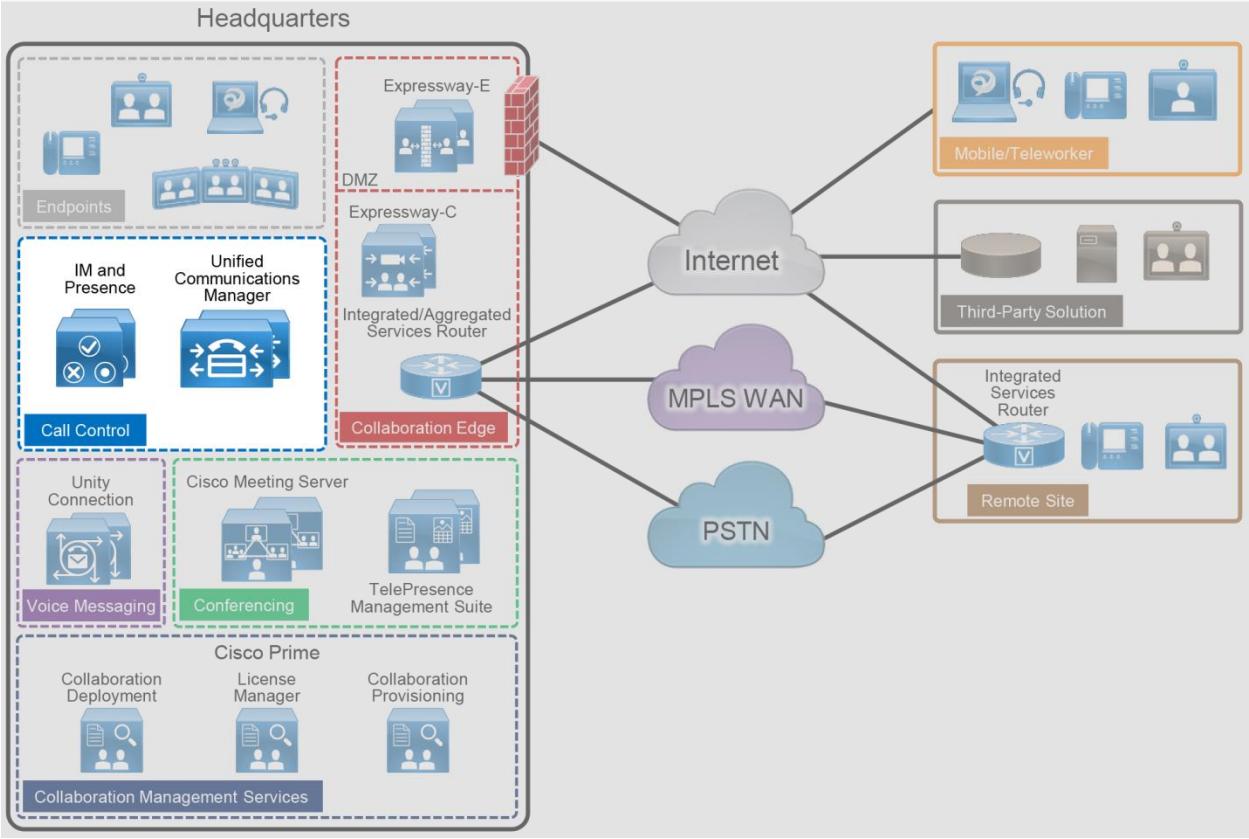


Table 6 lists the roles of the call control components in this architecture and the services they provide.

Table 6 Components for Call Control

Module	Component	Description
Call Control	Cisco Unified Communications Manager (Unified CM)	Provides call routing and services, dial plan, and bandwidth management; and enables Cisco Jabber desk phone control
	Cisco Unified Communications Manager IM and Presence Service	Provides Cisco Jabber support for instant messaging and user-based presence and third-party federation
	Cisco Integrated Services Router (ISR)	Provides Survivable Remote Site Telephony (SRST) to support call control functions during a WAN outage

Recommended Deployment

- Deploy a single Cisco Unified CM cluster for an enterprise with a central site and remote offices. Deploy call processing subscribers in pairs for scalability and redundancy.
- Add additional Cisco Unified CM clusters for very large sites or for geographic and/or organizational separation. Configure SIP trunks to interconnect individual Cisco Unified CM clusters.
- Deploy a pair of IM and Presence Service servers in a cluster configuration. Add more pairs for scalability.
- Enable Cisco SRST on the Cisco ISR as a backup service at remote sites to provide high availability.

Benefits

This deployment provides the following benefits:

- Call control is centralized at a single location that serves multiple remote sites.
- Common telephony features are available across voice and video endpoints.
- Single call control and a unified dial plan are provided for voice and video endpoints.
- Critical business applications are highly available and redundant.

Deployment Best Practices

Cisco Unified Communications Manager and IM and Presence Service

Cluster Recommendations

Cisco Unified CM and IM and Presence support clustering, which is the grouping of nodes that work together as a single logical entity. The publisher node contains the cluster's configuration database, which is replicated to the call processing subscriber nodes and TFTP nodes in the cluster.

Clustering provides an automatic redundancy mechanism for endpoints and for Cisco Unified CM services, such as the ability to receive and process incoming calls. To provide 1:1 redundancy, always deploy call processing subscribers and TFTP nodes in pairs. Each Unified CM cluster must have at least one pair of call processing subscribers and a pair of TFTP nodes in addition to the publisher node, for a minimum of five Cisco Unified CM nodes in a cluster (Figure 6). While the call processing subscribers provide endpoint registration and call processing capabilities, the pair of TFTP nodes provides configuration and firmware updates to endpoints.

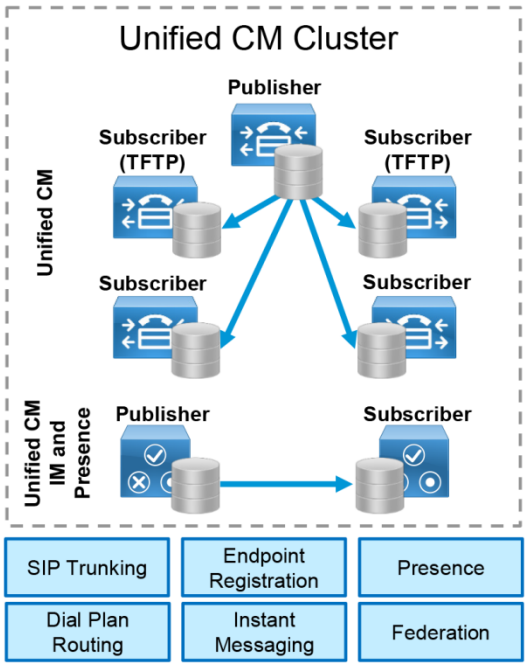
All the TFTP nodes and subscriber nodes periodically receive updates of the configuration database from the dedicated publisher node. These database updates enable all the subscriber nodes to operate in a consistent configuration state.

To provide load balancing of call processing services across the subscribers and to reduce failover response times, deploy each call processing subscriber pair in an active/active redundancy scheme.

For IM and Presence, we recommend deploying a minimum of one IM and Presence publisher and one subscriber. The IM and Presence publisher is not a dedicated node, and the publisher and subscriber provide redundancy for each other. (Figure 6)

Add more pairs of IM and Presence subscribers or Unified CM call processing nodes to accommodate more users.

Figure 6 Cisco Unified CM Cluster



SIP Trunk Recommendations

Use SIP trunks from Cisco Unified CM to communicate with all the components in the Cisco PA for Enterprise Collaboration, including external entities such as third-party systems. SIP trunks offer the following benefits:

- SIP trunks provide a standards-based environment that reduces operations and maintenance complexity of the end-to-end solution.
- SIP trunks are enhanced with presence information.
- SIP trunks are recommended for video communications.

Cisco Unified Survivable Remote Site Telephony

The Cisco Survivable Remote Site Telephony (SRST) feature is critical for remote sites that require continuation of voice services during WAN outages. SRST runs on the same Cisco ISR that provides WAN and PSTN connectivity for the remote site.

Deploy SRST on the Cisco ISR in the following cases:

- The remote site has local PSTN connectivity.
- The remote site does not have local PSTN connectivity but has more than 25 users.

To avoid interruption of external voice services if a WAN outage occurs, provide local PSTN connectivity at the remote site. SRST is required only if the remote site's WAN reliability does not match that site's required service level for voice service availability.

If a WAN failure occurs at a site with SRST and local PSTN access, the following services will be available:

- Internal point-to-point voice calls
- External voice calls through the PSTN
- Call hold, transfer, and conference
- Music on hold

Note: SRST is not available for Cisco DX, IX, MX, or SX Series endpoints. See Table 5 for information about endpoints that support SRST.

Dial Plan

A structured, well-designed dial plan is essential to successful deployment of any call control system. When designing a dial plan, consider the following main factors:

- Dialing habits
- Endpoint addressing
- Routing
- Directory integration
- Classes of service

Dialing Habits

Dialing habits describe what end users can dial to reach various types of destinations. Dialing habits can first be classified as numeric dialing (for example, 914085550123) or alphanumeric dialing (for example, bob@ent-pa.com).

Typically, different types of destinations require support for different dialing habits. For example:

- PSTN toll call: for example, in North America, 91-<*10 digits*>
- PSTN international call: for example, in North America, 9011-<*country code + national significant number*>
- Abbreviated intra-site dialing: for example, 4*XXX*
- Abbreviated inter-site dialing: for example, 8-<*site code*>-<*intra-site number*>
- +-dialing from directories: “+” followed by a fully qualified global PSTN number as described in ITU recommendation E.164
- URI dialing: for example, bob@ent-pa.com for intra-company and inter-company dialing. Endpoints typically allow omission of the right-hand side (host portion) of the URI and they automatically append the local host portion, so that bob@ent-pa.com can also be abbreviated as bob.

Further dialing habits might have to be defined for services such as call pick-up, voicemail, and others. Also, future growth should be considered so that more users and more sites can be added as needed without redesigning the dial plan.

Some dialing habits, typically PSTN dialing habits in particular, need to follow country-specific requirements or established dialing procedures. For example, in contrast to the trunk access code 9 in the above US-based examples, 0 is used for trunk access in many other countries. The dialing habit for national calls in these cases, in addition to the potential for using 0 as the trunk access code, also needs to reflect the characteristics of the national numbering plan of the respective country.

Identifying dialing habits is most important when defining an enterprise dial plan in order to avoid overlaps between any two dialing habits. For example, a trunk access code of 9 prohibits abbreviated intra-site dialing starting with 9. Avoiding overlaps between dialing habits is crucial to avoid inter-digit timeouts, which lead to bad user experiences.

In migration scenarios, the dialing habits supported by the existing system can be used as a first estimate of the dialing habits required in the new system. On the other hand, migration to a new communications system can also serve as a reason to get rid of outdated customs and practices.

Endpoint Addressing

Each endpoint registered with the enterprise call control must have a unique numeric address. Endpoint addresses in Cisco Unified CM are equivalent to the directory numbers provisioned on the lines of the endpoints. Use fully qualified PSTN numbers (E.164 numbers) with a leading "+" as endpoint addresses. This format is typically referred to as +E.164 format. The benefits of using +E.164 endpoint addresses include:

- Wide use in voice networks
- No need to develop and maintain an enterprise numbering scheme
- Easy creation of correct caller ID presentation for all on-cluster and off-cluster call flows
- Easy implementation of directory lookups
- Simplified alternate routing to the PSTN in cases of WAN failure or bandwidth constraints

For endpoints without assigned PSTN-based direct inward dial (DID) numbers (no E.164 number representation exists), create enterprise-wide unique endpoint addresses outside of the default +E.164 domain. These endpoint addresses should be in line with the internal dialing habit defined to reach these endpoints. If, for example, the abbreviated inter-site dialing habit to reach a set of non-DID endpoints in a given site is 84915XXX, then these non-DID endpoints should use this numbering scheme for their endpoint addresses.

In addition to the primary numeric endpoint addresses, administrators can provision alphanumeric URIs (for example, bob@ent-pa.com) in Cisco Unified CM to serve as aliases for the primary addresses, and users can enter the URI as an alternate way to dial the destination endpoint.

Routing

The routing portion of the dial plan enables users to reach the correct destinations when they use the defined dialing habits.

The primary numeric routing is based on +E.164 numbers. External routes to other transport networks such as the PSTN also use the +E.164 scheme. Endpoint addresses in +E.164 provide +E.164 on-net dialing without any further configuration. All other numeric dialing habits, such as abbreviated inter-site and intra-site dialing, are implemented as overlays by adding the appropriate translation patterns to the dial plan to map from the implemented dialing habit to the +E.164 global routing address format. This allows users to reach the same endpoint by means of different dialing habits, depending on user preference.

Alpha-numeric URIs, as aliases for numeric addresses, provide an alternative means of reaching endpoints. The benefits of URI dialing and routing include:

- Conformity with the native dialing habit on most video systems
- Easier business-to-business connectivity
- Direct mapping from instant messaging identifiers to addresses (easier escalation of business-to-business IM sessions to voice and/or video), although technically IM identifiers and SIP URIs are not necessarily identical

If an endpoint is enabled for business-to-business calls over the Internet, we recommend associating a SIP URI to the device so that the business-to-business routing logic can be based on SIP URIs.

As with numeric routing, if an alias or SIP URI is recognized as an internal destination and is associated with a specific device, then Cisco Unified CM sends the call to that device. However, if the dialed SIP URI does not match any registered endpoint alias, Cisco Unified CM uses SIP route patterns to determine where to send the call. For example, if the dialed alias room1@example.com does not exist internally, Cisco Unified CM uses a SIP route pattern (such as *.com) to send the call to Expressway-C as a business-to-business call.

Directory Integration

To enable users to search contacts and dial from the directory, integrate Cisco Unified CM with the organization's LDAP directory. Although Cisco Unified CM allows the creation of local user contacts, LDAP directory integration is required when using Cisco Jabber because it provides a single location for directory management and enables users to authenticate to Cisco Unified CM and Cisco Jabber by using their LDAP directory credentials.

Cisco Unified CM pulls user and contact information from LDAP directories and synchronizes user parameters – name, surname, username, telephone number, and SIP URI – when changes occur. For example, use the *telephoneNumber* attribute to populate the Telephone Number field in the Cisco Unified CM directory. The format of phone numbers in the corporate directory must be globally significant and must match one of the defined dialing habits. Corporate directories typically should have all phone numbers in +E.164 format (leading "+" followed by the fully qualified global number) as long as a DID exists. Only this format allows the phone number in the corporate directory to be used universally inside and outside the enterprise. Non-DID numbers that are not in +E.164 format could be used to dial uses internally from the directory, but they would have no significance outside the enterprise. Use the *mail* attribute to populate the Directory URI field in the Cisco Unified CM directory if URI dialing is used.

The IM and Presence Service pulls user and contact information from Cisco Unified CM.

Class of Service

Classes of service define which users can access which services, such as allowing only emergency and local calls from lobby phones while allowing unrestricted calls from executive phones. The complexity of the dial plan is directly related to the number of differentiated classes of service it supports.

To define classes of service, configure partitions and calling search spaces in Cisco Unified CM. The number of classes of services supported by a dial plan depends on the granularity and complexity of the classes. For more information about classes of service and details on enterprise dial plan design, see the Cisco Collaboration SRND.

Multi-Cluster Deployment Considerations

Consider deploying more than one Cisco Unified CM cluster if you have any of the following requirements:

- **Administrational separation**

  This includes the need to keep users from different parts of the organization on separate infrastructures, or the requirement to have different departments operate different parts of the communications infrastructure.

- **Geographic footprint**

  Technical limitations such as excessive propagation delay might prohibit endpoint registrations (for example, endpoints in Asia registering to an enterprise call control hosted in the US).

In a multi-cluster deployment, interconnect all the individual Unified CM clusters through SIP trunks. To avoid session traversal through individual clusters, deploy a full mesh of SIP trunks. With four or more clusters, deploy Cisco Unified CM Session Management Edition to centralize the dial plan and trunking and to avoid the complexity of a full-mesh SIP trunk topology.

In multi-cluster deployments, use Global Dial Plan Replication (GDPR) to replicate dial plan information between clusters. GDPR can advertise a +E.164 number, one enterprise significant number (ESN), and up to five alpha-numeric URIs per directory number. An ESN is the abbreviated inter-site dialing equivalent of a directory number. The information advertised and learned through GDPR enables deterministic intercluster routing for these dialing habits:

- +E.164 dialing based on the advertised +E.164 numbers
- Enterprise abbreviated inter-site dialing based on the advertised ESNs
- Alpha-numeric URI dialing based on the advertised URIs
- PSTN dialing based on normalization to +E.164

Conferencing

The ability for three or more people to communicate in real time by using voice and video technologies is a core component of collaboration. Cisco rich media conferencing builds upon existing infrastructure in place for point-to-point calls, offering users a consistent voice and video experience (Figure 7).

Figure 7 Architecture for Conferencing

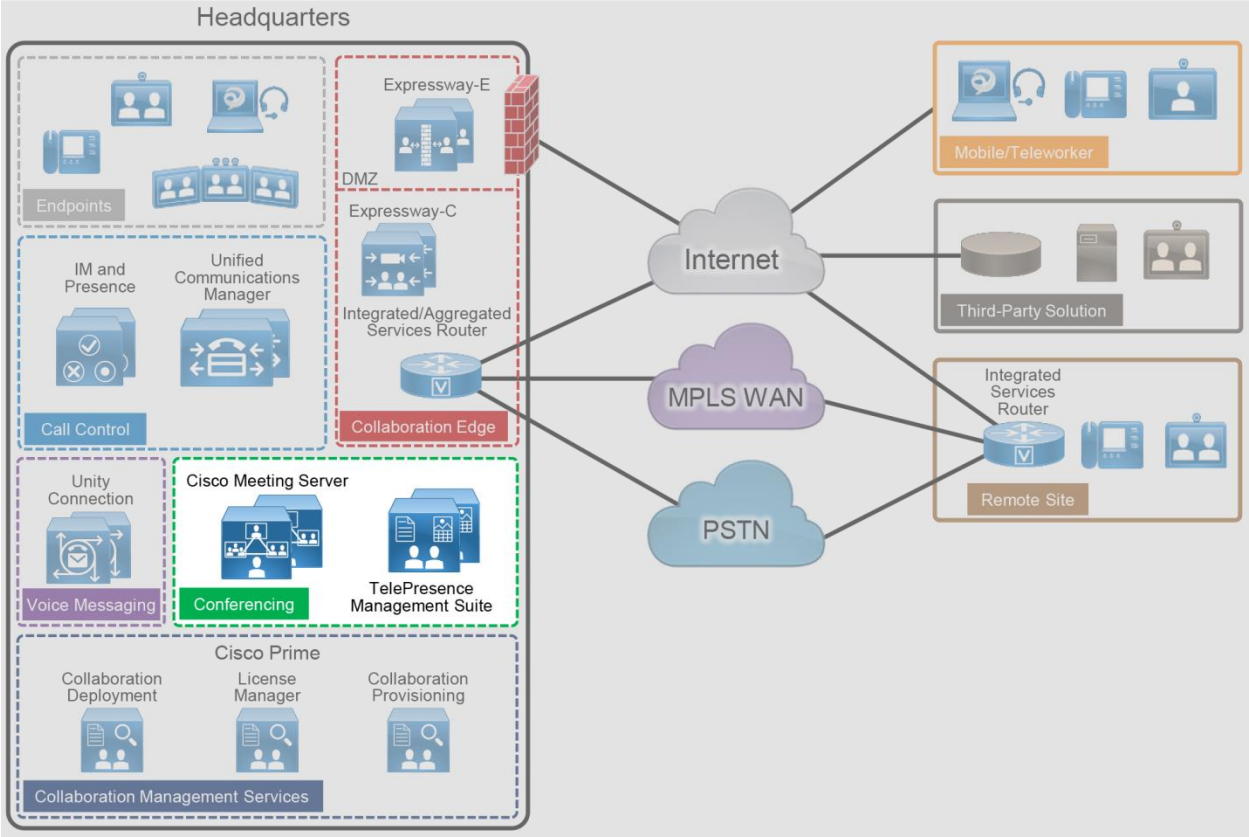


Table 7 lists the roles of the conferencing components in this architecture and the services they provide.

Table 7 Components for Conferencing

Module	Component	Description
Conferencing	Cisco Meeting Server	Provides voice and video conferencing with content sharing Manages and allocates conferencing resources
	Cisco TelePresence Management Suite and Extensions	Provides conference scheduling and monitoring, and device management capabilities Integrates with calendar system to schedule meetings

There are three types of conferences:

- **Instant or ad hoc** — A conference that is not scheduled or organized in advance. For example, a call between two parties who add other parties to the call is an instant conference.
- **Permanent or rendezvous** — A conference that requires callers to dial a predetermined number or URI to reach a shared conferencing resource. Meet-me, static, and rendezvous are other names for this type of conference.
- **Scheduled —** A conference planned in advance with a predetermined start time. Typically, conference resources are guaranteed to be available upon the start of the scheduled conference.

Recommended Deployment

Audio and Video Conferencing

- Deploy Cisco Meeting Server for all conference types.
- Deploy Cisco Meeting Server in a cluster for high availability and increased scale.
- Integrate the Cisco Meeting Server cluster with Cisco Unified CM through SIP trunks and registered media resource conference bridges for instant conferences.
- Integrate the Cisco Meeting Server cluster with Unified CM through SIP trunks and route patterns for permanent and scheduled conferences.
- Deploy Cisco TelePresence Management Suite (TMS) to schedule conferences with Cisco Meeting Server. Deploy Cisco TelePresence Management Suite Extension for Microsoft Exchange (TMSXE) to allow end users to schedule meetings using Microsoft Outlook clients.

Benefits

This deployment provides the following benefits:

- Users have a consistent experience for launching and joining various types of conferences.
- A single conferencing platform provides on-premises audio and video conferencing.
- It provides users with real-time, high-definition video conferencing, including the ability to share content easily over a dedicated presentation channel.
- Cisco TMS provides users with enhanced features such as directories and One Button To Push (OBTP) on controlled endpoints. It enables administrators to import user profiles from Microsoft Active Directory that allow access control to various components and configured systems.

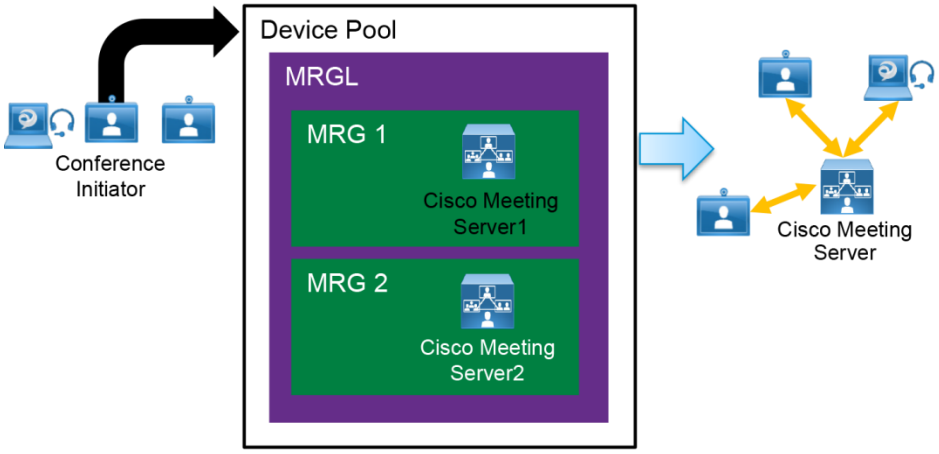
Deployment Best Practices

Audio and Video Instant Conferences

For instant audio and video conferences, use Cisco Meeting Server on-premises as the media resource. Unified CM has the HTTPS and SIP trunk interfaces to Cisco Meeting Server inside the instant conference bridges. HTTPS is used for conference control, while a SIP trunk is used for call signal. These conference bridges are assigned to media resource group lists (MRGLs) and media resource groups (MRGs) in Unified CM. Unified CM uses MRGLs and MRGs to prioritize and allocate media resources such as conference bridges, music on hold sources, annunciators, transcoders, and media termination points (MTPs).

If endpoints have access to the appropriate MRGL, they can request these resources. Resources local to the initiating endpoint are preferred over remote resources (Figure 8).

Figure 8 Media Resource Group List (MRGL) Example



Permanent Conferences with Cisco Meeting Server Spaces

Permanent conferences are deployed using Cisco Meeting Server Spaces. A Meeting Server Space is a virtual persistent meeting room that anyone can join and that has support for video, voice, and content sharing. A Space is automatically created for a user when the user is imported into Cisco Meeting Server from the Active Directory configured in the web administrative interface. Each Space is associated with a few attributes (for example, Username, Space name, and so forth) and can be accessed using a video address URI or numeric alias. These attributes are configured by the administrator through the Field Mapping Expressions. After the Space has been created, the administrator can further customize the Spaces by specifying a default layout or guest access code for each user. The Space owner can log into the Cisco Meeting Application to create a team Space and invite others to join for collaboration.

Scheduled Video Conferences

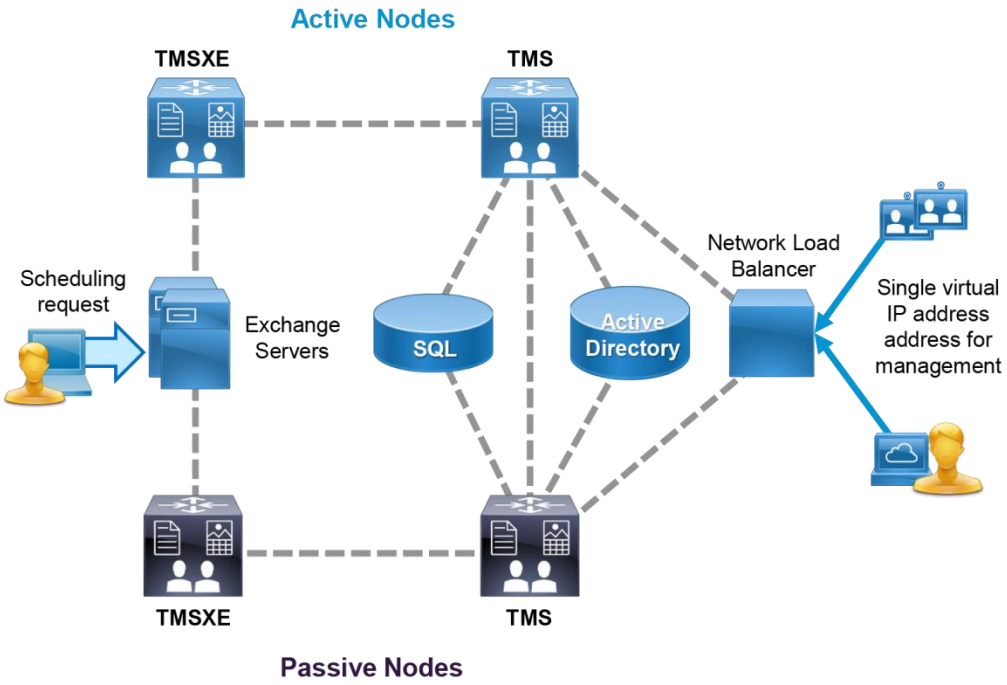
For scheduled video conferences, use the same Cisco Meeting Server as for non-scheduled conferences to provide the conferencing resource. Integrate the Cisco Meeting Server to Cisco Unified CM with SIP trunks, and manage it through Cisco TelePresence Management Suite.

Cisco TelePresence Management Suite (TMS) runs on a Microsoft Windows server and utilizes the Microsoft SQL database to store information about users, controlled devices, and scheduled conferences. User profiles are imported from Microsoft Active Directory, and the permissions model allows for access control to various components and configured systems. Deploy Cisco TMS with Cisco TMSXE to provide Microsoft Exchange integration.

A single deployment of TMS is required for each organization. Leverage the integrated system navigator folder structure to organize all endpoints and infrastructure devices. Even multinational and global organizations can benefit from a single deployment of TMS to facilitate video connections.

Redundancy for TMS and its supporting extensions is different from other components in the Cisco PA for Enterprise Collaboration. TMS and its components operate in an active/passive model instead of clustering. A single instance of TMS consists of a Network Load Balancer, two servers hosting TMS, two servers hosting the TMSXE application, and the SQL database (Figure 9). The licensing for the instance is maintained in the SQL database, so separate licensing is not required for each node. Only one server for each application is active at any moment, with the web pages and services of the passive (inactive) node locked down to refuse all other incoming traffic. All servers must be members of the same domain.

Figure 9 Cisco TMS Redundancy Model



Deploy the Microsoft SQL database separately from the TMS server. The instance of SQL may be shared by other applications within the organization, and it should be a high-availability deployment in accordance with Microsoft's recommendations.

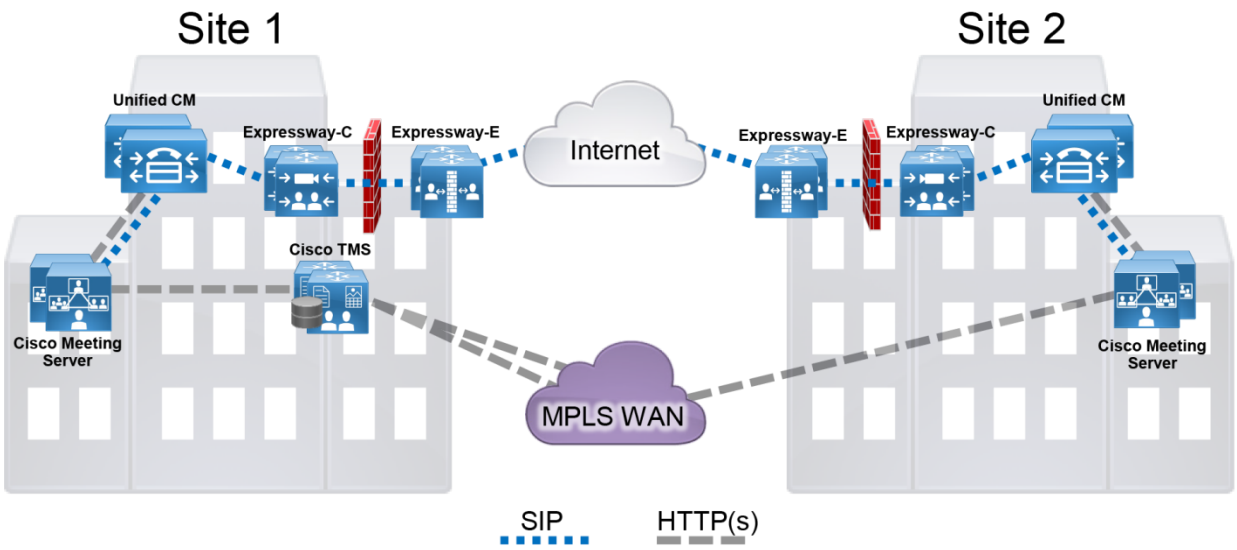
Support for Multiple Call Processing Sites

Organizations may choose to implement more than one Cisco Meeting Server cluster (Figure 10) for any of the following reasons:

- **Administrational separation** — This includes the need to keep users from different parts of the organization on separate infrastructures or to have different departments operate different parts of the communications infrastructure.
- **Geographic footprint** — Physical limitations such as excessive latency between endpoints and conferencing resources could degrade the user experience (for example, US users might not have a productive collaborative meeting if they use conferencing resources located in Europe).

However, when multiple Unified CM clusters are deployed, we recommend deploying a single Cisco Meeting Server cluster with one call bridge group dedicated for each Unified CM cluster. The call bridges within the group should be deployed in the same data center as the corresponding Unified CM cluster. Using a single Cisco Meeting Server cluster enables users to access the same conference using the same video address regardless of which Unified CM cluster they dial from.

Figure 10 Multiple Call Processing Sites with Conferencing



Collaboration Edge

Business demand for connectivity between organizations by leveraging the Internet has increased significantly over the past few years. For many organizations, this connectivity is a fundamental requirement for conducting day-to-day activities. Moreover, securely connecting mobile workers and remote sites to each other and to headquarters is critical functionality that enables organizations to accomplish their business goals. The Cisco PA for Enterprise Collaboration addresses these needs with the Collaboration Edge architecture shown in Figure 11.

Figure 11 Architecture for Collaboration Edge

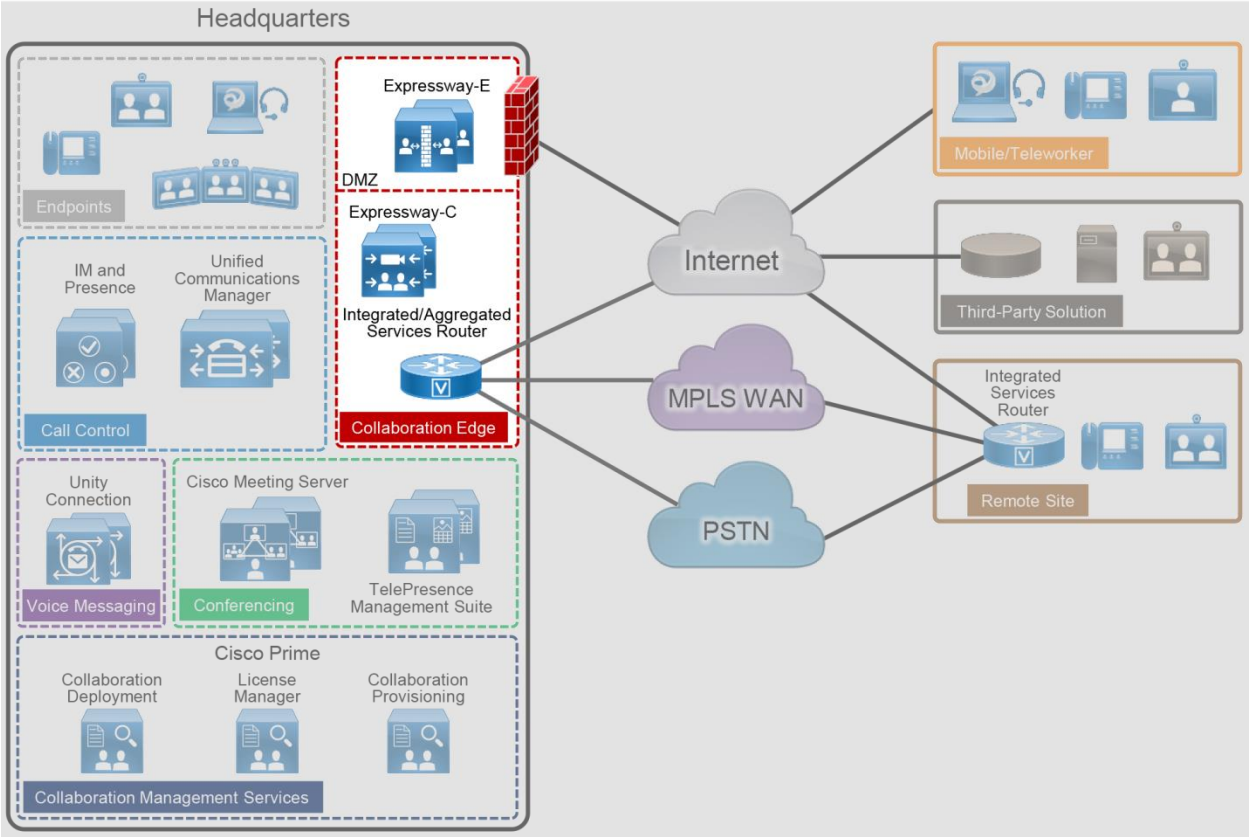


Table 8 lists the roles of the Collaboration Edge components in this architecture and the services they provide.

Table 8 Components for Collaboration Edge

Module	Component	Description
Collaboration Edge	Cisco Expressway-E	The traversal server that enables secure VPN-less mobile and remote access for TelePresence endpoints and Jabber clients. The traversal server resides in the DMZ. The solution also provides business-to-business calling, protocol interworking, and cloud connectivity.
	Cisco Expressway-C	The traversal client that creates a secure, trusted connection through the firewall to Expressway-E. The traversal client resides inside the enterprise network. The solution provides mobile and remote access, business-to-business calling, protocol interworking, and cloud connectivity.
	Cisco Integrated Services Router (ISR) or Aggregation Services Router (ASR) with PSTN interfaces	Enables local PSTN connectivity
	Cisco ISR or ASR with Cisco Unified Border Element (CUBE) software	Enables connectivity from an organization's network to the service provider network for SIP trunks via CUBE

Recommended Deployment

Headquarters

- Deploy a Cisco Expressway-C and Expressway-E server pair to enable remote Jabber and TelePresence video endpoint registrations, and IM and Presence. Deploy a separate Expressway-C and Expressway-E server pair for secure business-to-business connectivity through the firewall. Cluster both Expressway-C and Expressway-E servers in both pairs. If your deployment does not reach or exceed the scalability limit, you can deploy a single Expressway-C and Expressway-E cluster for both business-to-business and mobile and remote access applications.
- Deploy Cisco ISR or ASR as the PSTN gateway, or enable CUBE functionality on the Cisco ISR or ASR for voice connectivity from the organization's network to the service provider network through a SIP trunk.
- If full redundancy is not required, a single server pair (Expressway-C and Expressway-E) may be deployed.

Remote Sites

- Deploy Cisco ISR as the PSTN gateway.
- Deploy Expressway-C and Expressway-E if the remote site has local Internet connectivity and an Internet business-to-business architecture for video calls is required.

Teleworker Sites

- For video-enabled sites, deploy Cisco TelePresence endpoints utilizing the Expressway-C and Expressway-E infrastructure at headquarters or another site.
- In addition, the Cisco Jabber client can be used without the VPN, regardless of the location of the endpoint (internal or external to the organization).
- Legacy audio and video-enabled phones can be deployed with VPN technologies. Depending on the phone type, some of them have an embedded VPN client and may be deployed without a VPN hardware client. For more information on each phone model, refer to the product documentation.

Benefits

This deployment provides the following benefits:

- The Cisco ISR supports standards-based interfaces and various PSTN types, so it can be deployed globally.
- Instead of traditional PSTN interfaces, CUBE functionality can be enabled on the Cisco ISR and ASR if a SIP trunk is used.
- The Cisco ISR and ASR can be used for WAN connectivity.
- Cisco Expressway provides calling, presence, instant messaging, voicemail, and corporate directory services for Cisco Jabber and TelePresence video endpoints.
- Cisco Expressway enables video communications between organizations, partners, and vendors over the Internet.

Deployment Best Practices

Cisco Expressway

Cisco Expressway provides secure firewall and NAT traversal for mobile Cisco Jabber and TelePresence video endpoints (Figure 12) and secure business-to-business communications (Figure 13). Cisco Expressway consists of two applications: Expressway-C and Expressway-E.

Deploy Cisco Expressway-C inside the network, and deploy Expressway-E in the demilitarized zone (DMZ) by connecting separate network ports on Expressway-E to the organization's network and to the DMZ.

Cisco fully supports a virtualized Expressway-E in the DMZ; however, a dedicated server can be deployed based on the company's security requirements.

Figure 12 Traversal for Registrations Through Firewall with Expressway-C and Expressway-E

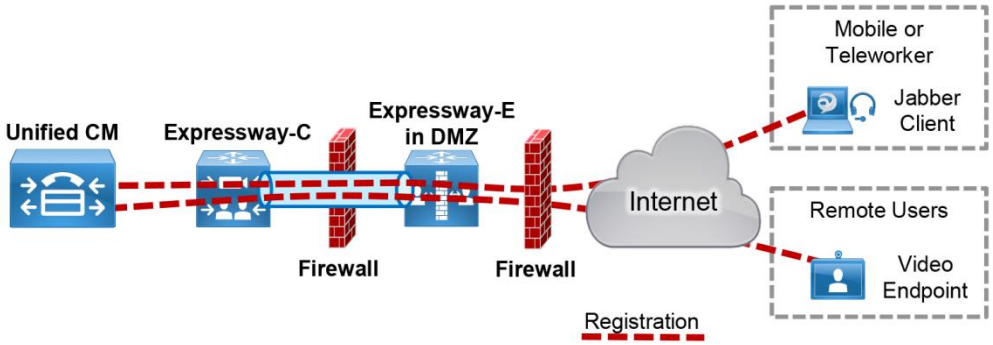
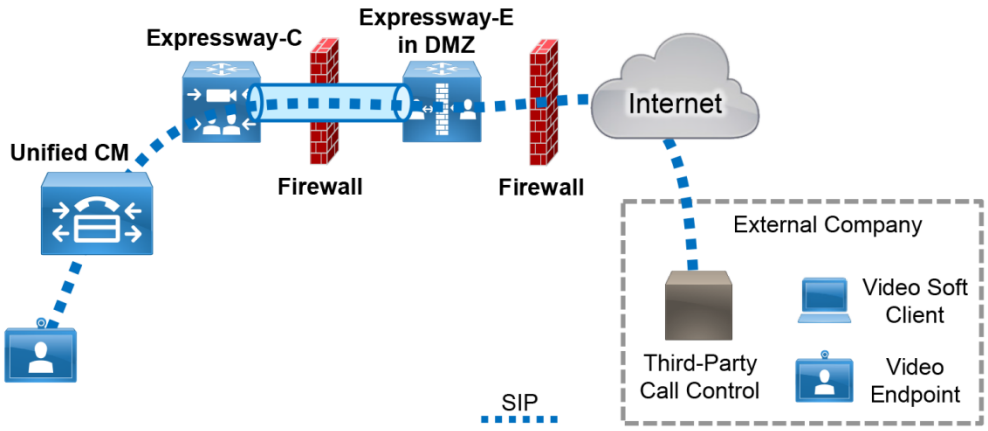


Figure 13 Traversal for Business-to-Business Calls Through Firewall with Expressway-C and Expressway-E



Cisco Expressway-C

Place Expressway-C in the trusted network inside the organization.

Deploy Expressway-C to:

- Function as a traversal client and establish a secure connection to Expressway-E through the firewall
- Establish secure or non-secure connection to Cisco Unified CM
- Integrate with an existing internal video network that uses H.323
- Enable business-to-business calls to external entities that communicate using SIP or H.323
- Provide interworking between H.323 and SIP protocols for H.323 business-to-business communications
- Enable mobile and remote access capabilities and call signaling for Cisco supported endpoints, directing them to Cisco Unified CM for SIP registration and/or the IM and Presence Service (See the Endpoints section for information on which endpoints support mobile and remote access.)

Cisco Expressway-E

Because Expressway-E is reachable directly from the untrusted external network, it should be placed in a DMZ for security. The organization's firewall policies control communications to and from this server.

Deploy Expressway-E to:

- Function as a traversal server and allow secure communications to and from Expressway-C
- Enable audio, video, and IM and Presence connections to other organizations using SIP or H.323 on the Internet
- Provide DNS SRV lookup service to resolve outbound calls and to receive inbound calls over the Internet
- Process registration and IM and presence information from Cisco endpoints on the external network and use secure traversal communications to pass the information to Expressway-C
- Provide interworking between protocols (between SIP and H.323, and between IPv4 and IPv6) for business-to-business communications

Connectivity for Audio and Video over the Internet

URI dialing is the best practice for audio and video dialing over the Internet. We recommend assigning alphanumeric URIs to all devices that will send or receive calls over the Internet. Any device on Cisco Unified CM can be reached over the Internet by dialing the assigned alphanumeric SIP URI or the required directory number (DN) by dialing *<+E.164 number>@domain*. For example, a Jabber user might have a SIP URI set to alice@ent-pa.com and a directory number set to +14085551234. Although a +E.164 addressing scheme can be used over the Internet, we recommend allowing only alphanumeric SIP URIs for security reasons.

Users on Cisco Unified CM have to dial the full SIP URI to reach a user or device from a different organization over the Internet.

The architecture for business-to-business Internet connectivity includes a client/server solution: Expressway-C and Expressway-E. Both servers can be deployed in standalone mode or in a cluster. Deploy the same number of cluster peers for Expressway-C clusters as for Expressway-E clusters.

Considerations for Outbound Business-to-Business Calls

- When multiple Expressway-C and Expressway-E pairs are deployed, Unified CM can redirect an outbound call to the edge server that is nearest to the calling endpoint, thus minimizing WAN traffic.
- For call routing over the Internet, use public DNS service records. DNS SRV records map a domain to an edge system servicing that domain for that protocol. For example, if a remote user dials alice@ent-pa.com, then the remote system uses DNS to query for the host offering the SIP service for the domain ent-pa.com.
- If the remote endpoint supports IP dialing only, Cisco Unified CM users can dial the IP address of the endpoint followed by a string, which will be used to route the call. As an example, the user can dial 10.10.10.10@ip instead of dialing 10.10.10.10. The string "@ip" will be used by Cisco Unified CM to route the call to the Expressway-C. Expressway will send it to the Expressway-E, which will place the call to the IP address specified.

Considerations for Inbound Calls

Once a call reaches an Expressway-E, it is routed to the relevant Unified CM cluster through its corresponding Expressway-C. In deployments with multiple edges (multiple pairs of Expressway-C and Expressway-E), there are two methods to route inbound calls:

- **Call routing based on the calling location**

  In this scenario, a business-to-business call enters the corporate network through the edge that is nearest to the calling endpoint or user. Because the call enters the corporate network after traversing a minimal distance over the Internet, this approach focuses on using the shortest path to the point of entry as the means of providing the best quality experience. In this scenario, use Geo-DNS. Geo-DNS provides unique DNS responses by geographic regions based on the source IP address of the DNS query, and it is thus able to direct an SRV query to the specific edge servicing a particular region. In this way, the calling endpoint is typically directed to the edge nearest to its calling location.

- **Call routing based on the called location**

  It is possible to direct the inbound call to the edge that is nearest to the called endpoint. This approach has the benefit of reducing the video bandwidth consumed over the WAN, but it requires a more complex architecture that has some scaling constraints. For this reason, we do not recommend implementing this architecture when more than two edges are deployed.

Note: Call routing based on the called location applies to room and personal systems or clients that are not moving across sites.

In addition, the Cisco Collaboration Architecture enables IP-based dialing for those endpoints on the Internet that are capable to dial IP addresses only. The Expressway-E external interface IP address can be dialed from the Internet, and

the call can be sent to a multipoint device or to a Cisco Unity Connection system, which will prompt the calling user to specify the destination. Once the destination is entered, the call will be sent to the specified destination.

Mobile and Remote Access

The mobile and remote access feature enables Jabber clients and hardware endpoints (as indicated in Table 5, Mobile and Remote Access column) to register securely to Cisco Unified CM through Expressway-E and Expressway-C without any VPN. A Jabber client can send and receive several types of collaboration flows (audio, video, instant messaging, and presence), while a hardware endpoint can send audio and video streams. When multiple edges are deployed, we recommend using Geo-DNS services to provide the best network option based on assigning the closest edge in the DNS response.

The mobile and remote access functionality also leverages Expressway-C and Expressway-E. Both business-to-business and mobile and remote access services are supported on the same server, but we recommend deploying these services on different Expressway-C and Expressway-E pairs in order to scale.

Instant Messaging and Presence Federation

Instant messaging and presence federation involves allowing users to send XMPP traffic through an organization's external firewall for chat and presence status information to and from users in another organization.

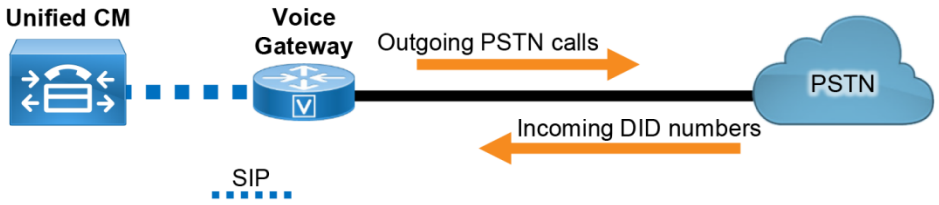
Prior Cisco architectures involved using the Cisco ASA firewall as a TLS proxy and allowing inbound ports to be opened through the external firewall to directly access the IM and Presence servers internally.

This architecture is still valid for SIP federation for IM and Presence only, while Expressway-C and Expressway-E provide XMPP federation with voice and video escalation.

PSTN Gateway

Because landlines and mobile phones use the PSTN for local and international calls, external connectivity to the PSTN from an organization's IP telephony network is a requirement (Figure 14).

Figure 14 PSTN Connectivity



Use Cisco ISR or ASR with a time-division multiplexing (TDM) module as the PSTN gateway at headquarters. This configuration enables the gateway to implement media interworking for the organization's incoming and outgoing PSTN calls.

At remote sites, deploy a Cisco ISR for local PSTN connectivity using voice modules. For more information about Cisco ISR, see the data sheet.

If SIP trunks are used to connect to a service provider for voice calls, enable CUBE functionality on the Cisco ISR that is deployed at headquarters. When deploying Cisco ISR with CUBE functionality, observe the following recommendations:

- Deploy CUBE in the demilitarized zone (DMZ).
- Enable the firewall for NAT to convert the external address to the address of CUBE.
- Enable the firewall to inspect voice calls.

Cisco Unified CM routes calls through SIP trunks to gateways, CUBE, or Cisco Expressway based on the dial plan. For dial plan recommendations, see the Call Control section.

PSTN Connectivity for Voice

Enable PSTN connectivity for voice calls by using either an analog or ISDN interface. A Cisco ISR or ASR with analog or ISDN cards provides these interfaces. Connectivity is usually local, and a site with PSTN interfaces uses its local ISR or ASR as a voice gateway. Follow these recommendations for deploying an ISR or ASR for PSTN connectivity:

- PSTN interface (analog or ISDN)
  - The device providing these interfaces is a Cisco ISR or ASR with analog or ISDN cards.
  - Connectivity is usually local; a site with PSTN interfaces uses its local ISR or ASR as a voice gateway.
  - Redundancy is achieved by deploying multiple ISRs or ASRs. Cisco Unified CM has the ability to route traffic to the closest available router.
- SIP trunks to the service provider and ISR, ASR, or CUBE as a border element
  - This deployment is typically used in a centralized architecture. Remote sites either do not have local connectivity, or they have local connectivity but use it only for backup voice services. In this case the WAN connectivity has to be sized to accommodate PSTN calls traversing the WAN to the central site where CUBE is deployed.
  - Redundancy can be achieved by deploying multiple ISRs or ASRs, sometimes to different voice carriers. Cisco Unified CM has the ability to route traffic to the closest available router.

Voice Messaging

Voice messaging is considered to be a basic requirement and essential service for an Enterprise Collaboration deployment. Cisco Unity Connection enables users to access and manage voice messages from their email inbox, web browser, Cisco Jabber client, Cisco Unified IP Phone, or TelePresence endpoint. The Cisco PA for Enterprise Collaboration includes Cisco Unity Connection to enable voice messaging for the Enterprise Collaboration solution (Figure 15).

Figure 15 Architecture for Voice Messaging

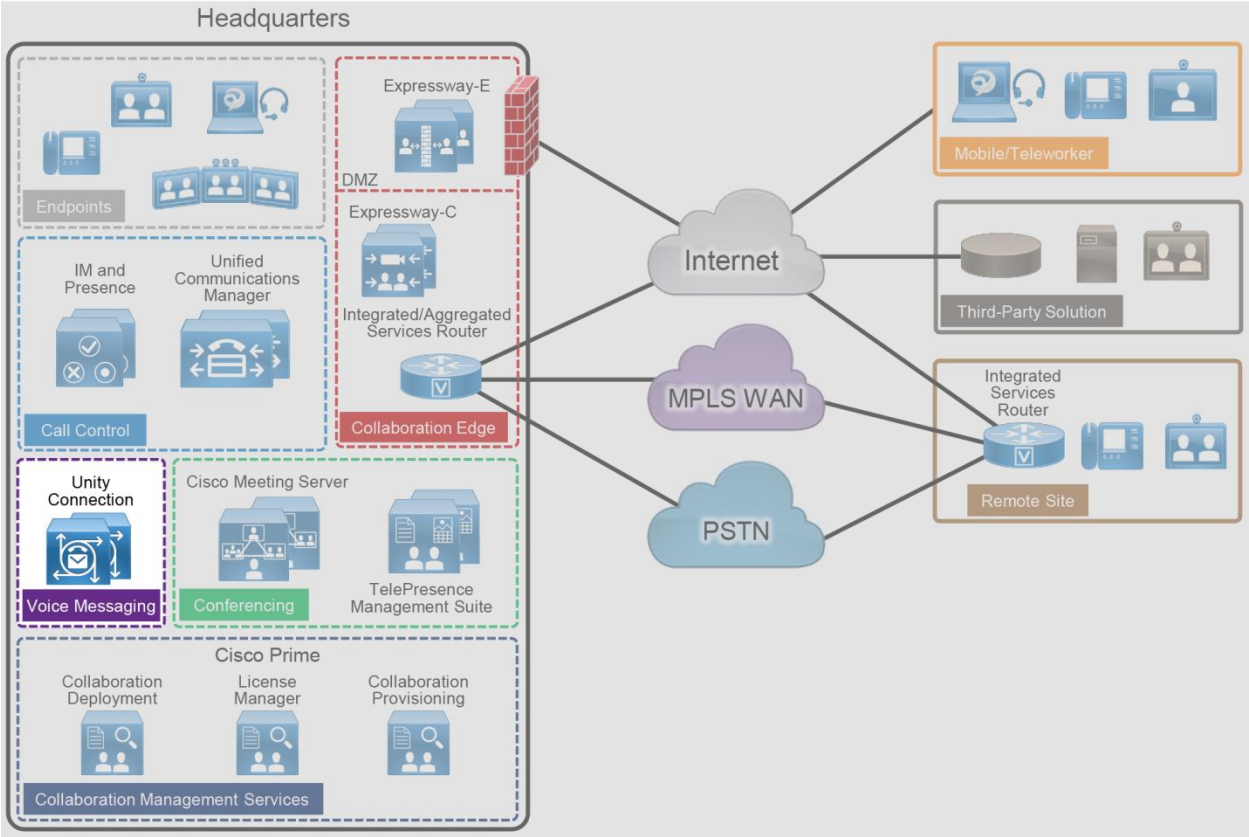


Table 9 lists the roles of the components in this architecture and the services they provide.

Table 9 Components for Applications

Module	Component	Description
Voice Messaging	Cisco Unity Connection	Provides unified messaging and voicemail services

Recommended Deployment

- Deploy two Unity Connection servers for each Cisco Unified CM cluster to provide high availability and redundancy.
- Use SIP trunks to integrate Unity Connection with Unified CM. Configure two SIP trunks, one for each Unity Connection server in a pair.
- Enable the speech-activated voice command interface to maximize productivity of mobile workers.

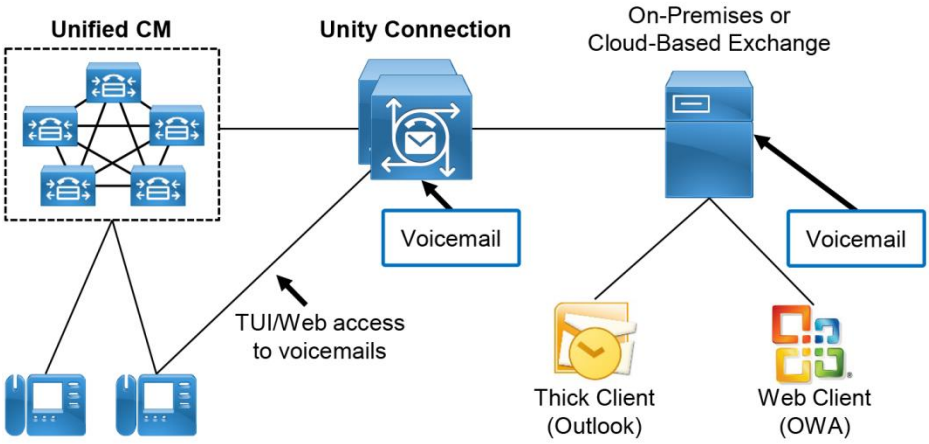
Benefits

- Users can access the voicemail system and retrieve their voice messages by using their IP phones, mobile devices, and various email client applications with either a dialed number or a SIP URI.
- Cisco Unity Connection allows users to customize personal settings from a web browser.
- Cisco Unity Connection offers a natural and robust speech-activated user interface that allows users to browse and manage voice messages using simple and natural speech command.

Deployment Best Practices

Cisco Unity Connection supports a cluster configuration in active/active mode to provide both high availability and redundancy. A Unity Connection cluster consists of a maximum of two nodes, one publisher and one subscriber in an active/active deployment (Figure 16). If one of the Unity Connection nodes fails, the other active node in the cluster handles all the calls, IMAP requests, and HTTP requests for the Unity Connection cluster. Each server in the Unity Connection cluster must have enough voice messaging ports to handle all calls for the cluster.

Figure 16 Unified Messaging Architecture



Single Inbox, one of the unified messaging features in Cisco Unity Connection, synchronizes voice messages in Unity Connection and Microsoft Exchange mailboxes. Unity Connection supports the Single Inbox feature with on-premises Microsoft Exchange, cloud-based Microsoft Exchange, or Microsoft Office 365 server, thereby providing unified messaging for voicemail. All voice messages, including those sent from Cisco Unity Connection ViewMail for Microsoft Outlook, are first stored in Cisco Unity Connection and are immediately replicated to the Microsoft Exchange mailbox of the recipient. This feature can be configured separately for each individual user.

Unity Connection imports the user information from the enterprise LDAP directory. Each mailbox must have a unique voicemail number. Unity Connection supports both E.164 and + E.164 formats for the extension of an end-user account (user with a voice mailbox). Unity Connection also supports alternate extensions per user.

The voicemail pilot number designates the directory number that users dial to access their voice messages. Unified CM automatically dials the voice messaging number when users press the Messages button on their phone. The voicemail pilot number can be an internal extension or a dedicated PSTN number.

Visual Voicemail allows users to access voicemail from the graphical interface on the IP phone. Users can view a list of messages and play messages from the list. Users can also compose, reply to, forward, and delete messages. Each voicemail message displays data that includes the date and time when the message was left, urgency level, and message length.

For more information on Cisco Unity Connection, refer to the product documentation.

Collaboration Management Services

This section focuses on management of the collaboration system environment. The Cisco PA for Enterprise Collaboration includes the following Cisco core management applications that are considered to be a basic requirement and foundational to any Enterprise Collaboration solution (Figure 17):

- Cisco Prime Collaboration Deployment — Assists with installation of applications.
- Cisco Prime License Manager — Assists with license management.
- Cisco Prime Collaboration Provisioning – Assists with provisioning as well as moves, adds, changes, and deletions (MACD) for day-2 management control.

Figure 17 Architecture for Collaboration Management Services

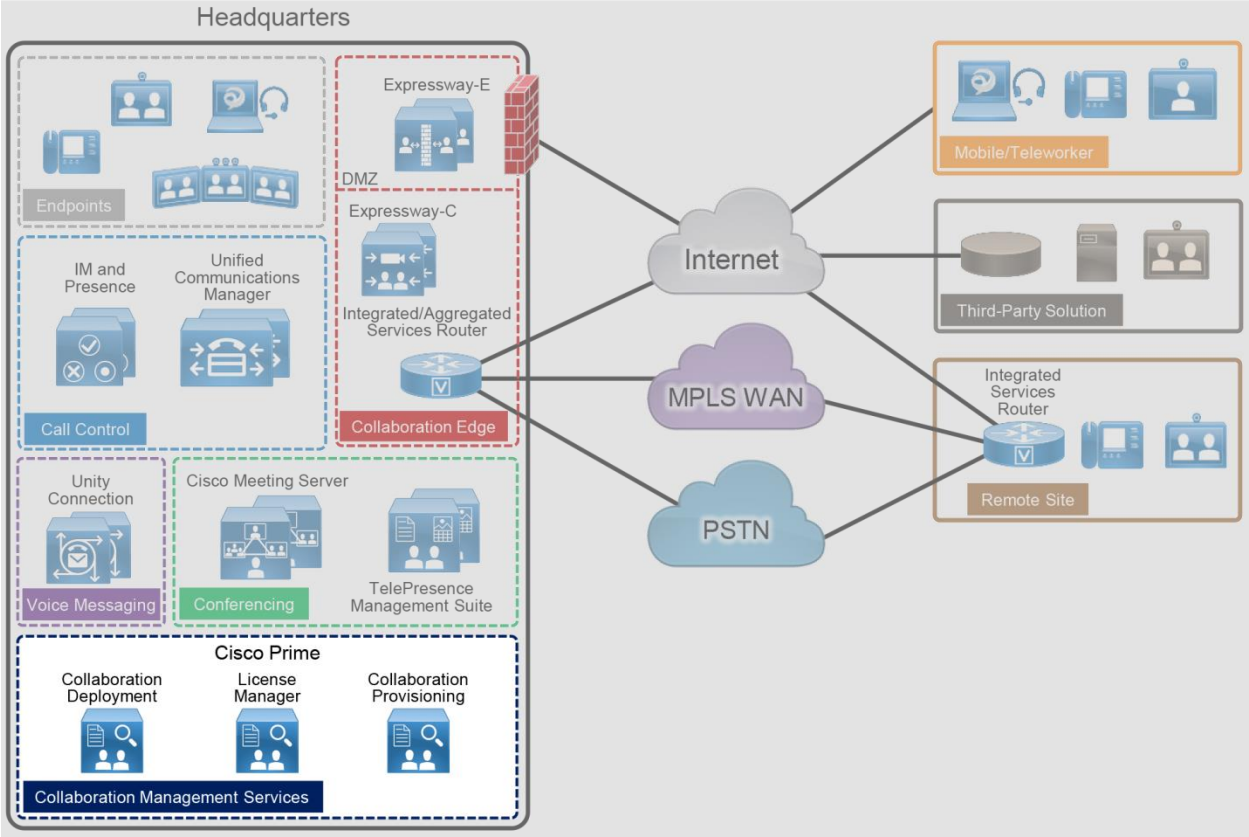


Table 10 lists the roles of the application components in this architecture and the services they provide.

Table 10 Components for Collaboration Management Services

Module	Component	Description
Collaboration Management Services	Cisco Prime Collaboration Deployment	Assists the administrator by automating many of the steps necessary to install a Cisco Unified CM cluster with IM and Presence Service and a Unity Connection cluster.
	Cisco Prime License Manager	Provides the administrator with a single management point for the Cisco Unified CM licenses and Unity Connection licenses used in a deployment.
	Cisco Prime Collaboration Provisioning	Enables rapid deployment of collaboration systems by providing a template-driven console for device provisioning as well as moves, adds, and changes.

Cisco Prime Collaboration Deployment

Cisco Prime Collaboration Deployment assists the administrator by automating many of the steps necessary to preliminarily configure and install Cisco Collaboration applications.

Cisco Prime Collaboration Deployment supports the following Cisco Collaboration Applications in the Enterprise Preferred Architecture:

- Cisco Unified Communications Manager (Unified CM)
- Cisco IM and Presence Service
- Cisco Unity Connection

Recommended Deployment

For the Enterprise PA, we recommend deploying Cisco Prime Collaboration Deployment as a standalone virtual machine (VM) and not co-resident with any other applications.

Benefits

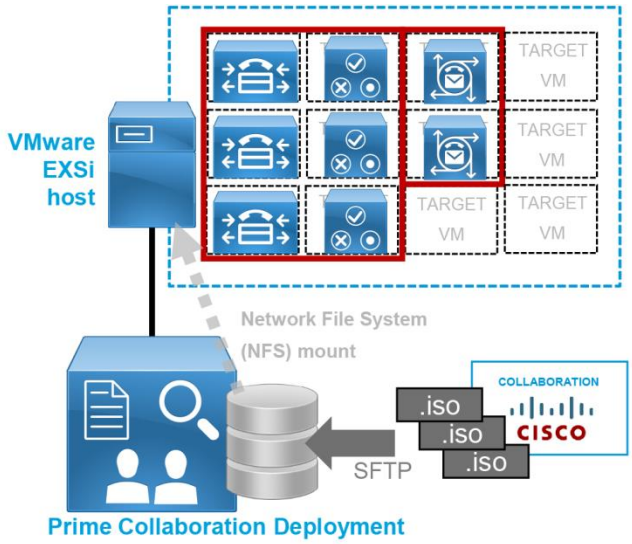
Cisco Prime Collaboration Deployment provides the following benefits:

- Enables automated unattended installation of ESXi-hosted collaboration application virtual machine server nodes
- Facilitates configuration of a common base platform and initial application settings for all collaboration application nodes, including:
  - Network services (time, domain name)
  - Administrative accounts and passwords
  - Base certificate information

Deployment Best Practices

Figure 18 shows the preferred architecture for Prime Collaboration Deployment. Once Cisco Prime Collaboration Deployment is deployed as a standalone VM, the collaboration application installation ISO files (available from Cisco.com) are transferred to the Cisco Prime Collaboration Deployment file store using secure FTP (SFTP). Next the ESXi servers that will host the collaboration application virtual server nodes must be defined within Cisco Prime Collaboration Deployment. The Cisco Prime Collaboration Deployment file store is then mounted automatically via NFS to each ESXi host, thus making the ISO files available for collaboration application installations. After target VMs are created on the ESXi host for the Unified CM and IM and Presence cluster and Unity Connection cluster server nodes, installation of the collaboration application nodes is defined and started from Cisco Prime Collaboration Deployment.

Figure 18 Architecture for Prime Collaboration Deployment



Cisco Prime License Manager

Cisco Prime License Manager provides simplified, enterprise-wide management of user-based licensing, including license fulfillment and reconciliation of licenses across supported products, and it provides enterprise-level reporting of usage and entitlement. Cisco Prime License Manager also supports multiple clusters.

Cisco Prime License Manager supports the following Cisco Collaboration applications in the Enterprise Preferred Architecture:

- Cisco Unified Communications Manager (Unified CM)
- Cisco IM and Presence Service
- Cisco Unity Connection

Recommended Deployment

In the Enterprise PA, we recommend deploying Cisco Prime License Manager as a standalone virtual machine and not co-resident with any other applications.

Benefits

- Centralizes license management for Unified CM, IM and Presence, and Unity Connection
- Provides license pooling
- Minimizes re-hosting of license files
- Eliminates dependency of licenses on versions of Unified Communications applications

Cisco Prime Collaboration Provisioning

Cisco Prime Collaboration Provisioning provides a scalable web-based solution to help administrators manage the provisioning needs of an integrated IP telephony, video, voicemail, and unified messaging environment. Cisco Prime Collaboration Provisioning assists the administrator with user and device provisioning, thereby enabling rapid deployment. After the initial configuration and provisioning, Cisco Prime Collaboration Provisioning simplifies moves, adds, and changes, as well as the configuration and deployment of new features. An intuitive user interface provides a single consolidated view of users and their services.

Recommended Deployment

Cisco Prime Collaboration supports high availability (HA) through the VMware vSphere HA feature. You do not need an additional Cisco Prime Collaboration license to configure HA, and HA is highly recommended to increase uptime in case of a failure of the host on which Prime Collaboration Provisioning resides. For small and medium deployment models, you need one virtual machine for Prime Collaboration Provisioning. For large and very large deployment models, you must configure the Prime Collaboration Provisioning database and application on separate virtual machines.

We highly recommend performing regular backups to an external FTP server and taking periodic VM snapshots because a considerable amount of time and effort is required to configure the system and get it running initially. These processes also help retain the logs and order history for each user and help restore data in case of a catastrophic failure.

Benefits

Cisco Prime Collaboration Provisioning provides the following features and benefits:

- By significantly simplifying moves, adds, changes, and deletions (MACD), Cisco Prime Collaboration Provisioning facilitates delegation of these tasks to helpdesk users, thus allowing organizations to optimize IT resources and further reduce total cost of ownership.
- Cisco Prime Collaboration Provisioning significantly accelerates site rollouts and dramatically reduces the time required for ongoing changes, resulting in exceptional productivity gains and lower operating expenses.
- Wizards in Cisco Prime Collaboration Provisioning expedite provisioning for greenfield deployments.
- Infrastructure and User Services provisioning features for greenfield and brownfield deployments help manage granular Role Based Access Control (RBAC) for different administrators.
- Cisco Prime Collaboration Provisioning provides a single, consolidated view of users across the organization and across all clusters, thereby presenting a single interface for all provisioning needs of an organization.
- Cisco Prime Collaboration Provisioning allows IT administrators to embed policies into Cisco Prime Collaboration Provisioning for managing user services across the Cisco Unified Communications applications. Administrators can configure policies at various levels to determine who can do delegated management, for whom that delegation applies, how business-level services apply to Cisco Collaboration Systems, and which types of users are permitted to order which standard services. With this approach for policy and standards configuration, delegated individuals can provision and activate user services easily. At the same time, the primary administrator retains the overall ability to manage and provide services that use the underlying Cisco Unified Communications applications.
- Cisco Prime Collaboration Provisioning provides template capability, which permits defining standard configurations that can be reused for new sites or additional location deployments.
- Automatic Service Provisioning helps the administrator expedite employee on-boarding and off-boarding processes. This allows the administrator to add a new user and, based on company policies and location, it automatically provisions the new user's common services.
- Batch provisioning permits the rollout of large numbers of users at the same time.

Deployment Best Practices

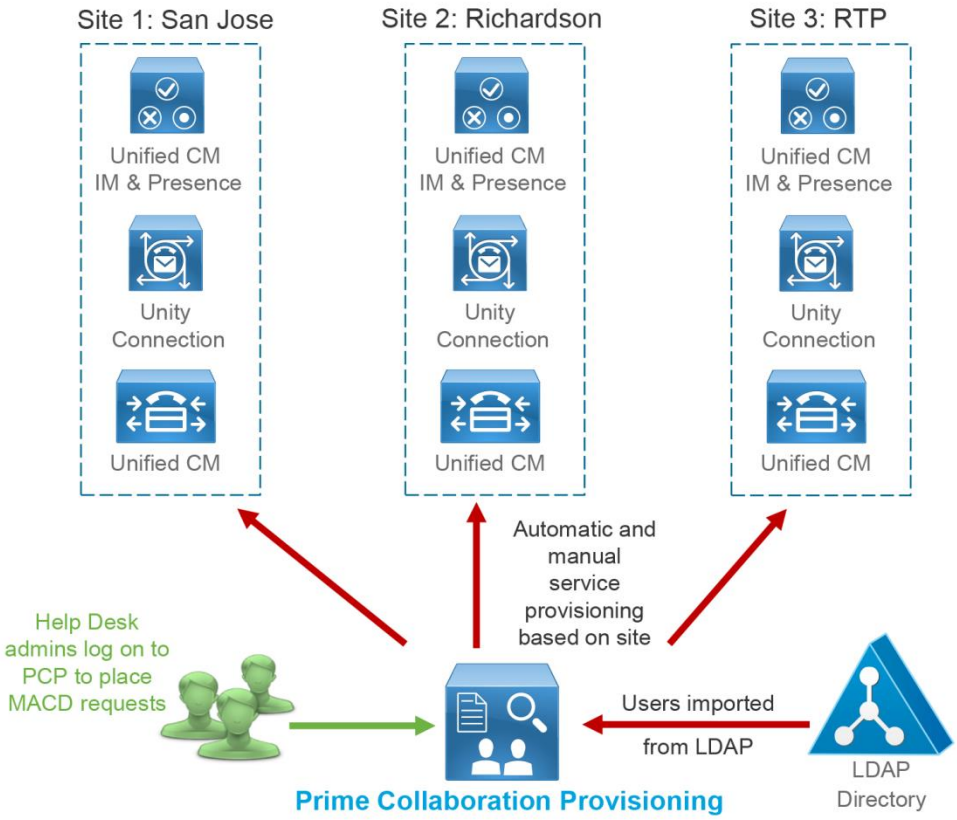
Cisco Prime Collaboration Provisioning is very flexible and can adapt to any enterprise deployment model. For example, users can be brought into Cisco Prime Collaboration Provisioning from an LDAP server directly or from Unified CM as well. Figure 19 shows the recommended architecture, where Cisco Prime Collaboration Provisioning is integrated through LDAP and all users in the organization are brought into Cisco Prime Collaboration Provisioning from LDAP. This architecture allows the synchronization and authentication of users to be decoupled within Unified CM. This setup also allows the administrator to leverage Automatic Service Provisioning, which provisions a bundle of services when a new employee is added into the LDAP server and also de-provisions those services when an employee is deleted from the LDAP server.

Cisco Prime Collaboration Provisioning is also very flexible in terms of how the users are grouped. For example, users can be grouped based on the cluster to which they belong, based on administrative control, or based on the users' site.

Grouping by site is intuitive and also recommended to scale the service areas, user roles, and service templates required within each domain (user group).

With Cisco Prime Collaboration Provisioning as the front end for all provisioning activities in the Unified Communications environment, administrators can also be granted Role Based Access Control (RBAC) to limit access to their respective domains and also infrastructure objects via infrastructure permission profiles.

Figure 19 Cisco Prime Collaboration Provisioning Deployed with LDAP Integration

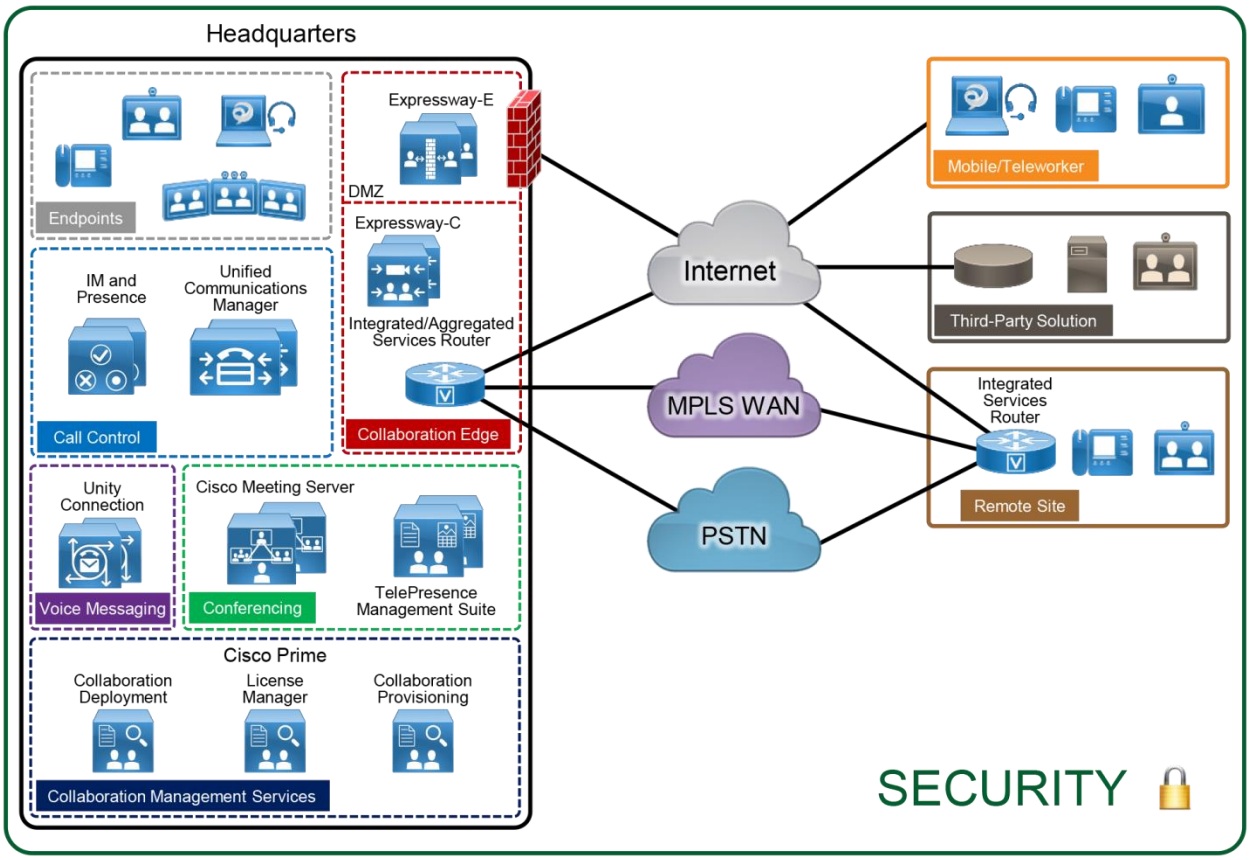


Security

As with almost everything today, it is important to secure your collaboration deployment. A collaboration deployment is subject to threats such as denial of service, unauthorized access, toll fraud, and eavesdropping. It is important to protect your collaboration deployment against these threats. Take a layered security approach by securing various network levels: secure physical access, network infrastructure, collaboration applications, and collaboration endpoints (Figure 20).

Solely following the guidelines in this section does not guarantee a secure environment, nor will it prevent all penetration attacks on a network. You can achieve reasonable security by establishing a good security policy, following that security policy, staying up-to-date on the latest developments in the hacker and security communities, and maintaining and monitoring all systems with sound system administration practices.

Figure 20 Secure All Components of the Enterprise Collaboration Preferred Architecture



Recommended Deployment

- Secure the infrastructure by protecting physical access, and secure the IP network.
- Use hardening techniques to secure all devices, including servers and endpoints.
- Protect your deployment against toll fraud.
- Simplify certificate management by having certain certificates signed by a Certification Authority (CA).
- Do not disable native security features. For example, with Cisco Unified CM, do not disable *Security By Default*.
- Encrypt SIP trunks, HTTP connections, and other server-to-server links.

- To protect sensitive voice and video communications, enable mixed-mode on Cisco Unified CM and enable encrypted signaling and media for endpoints. This is especially important if your network is not entirely trusted and secure.

Benefits

This deployment provides the following benefits:

- Your collaboration deployment is more secure if the physical access is protected and the IP network is secured.
- By protecting network access to servers and phones, you make it more difficult to compromise them and get access to other devices in the deployment.
- By implementing toll fraud protection mechanisms, you can prevent unauthorized access to your telephony system, data network, and PSTN lines.
- By signing certain certificates with a CA, you make it easier to manage the certificates and, more importantly, you increase security by avoiding scenarios where end-users must accept certificates on their computing device, which most end-users do without verifying the authenticity of the certificates.
- Several secure features are implemented by default. For example, with Cisco Unified CM, phone configurations and firmware loads are signed so that it becomes more difficult to compromise the phones by loading malicious configurations or firmware.
- Encryption protects against eavesdropping and protects the privacy of voice and video calls. It also protects against tampering. By encrypting communications between all devices, including the endpoints, you can achieve end-to-end encryption.

Deployment Best Practices

Secure Infrastructure Recommendations

- Secure your infrastructure; it is the foundation of your collaboration deployment.
- Protect physical access to your premises, network, endpoints, and especially the servers.
- Protect your network with firewall and Intrusion Prevention System (IPS) devices.
- Implement security features at Layer 2 and Layer 3 for your network. For example, protect access to your network with 802.1X, and protect your DHCP server with DHCP Snooping and Dynamic ARP Inspection.
- Implement network segmentation by having a separate voice/video VLAN for hardware endpoints and a data VLAN for multipurpose devices such as mobile phones and laptops running Jabber.
- With Cisco Unified Border Element deployed at the network edge, configure the Unified Border Element protection mechanisms against telephony denial of service (TDoS) and configure access control lists (ACLs).

Device Hardening Recommendations

- Protect network access to your devices by using hardening techniques.
- Use secure password policies and do not rely on default passwords.
- Restrict access to your devices.
- Protect not only the servers but also the endpoints.

Toll Fraud Recommendations

On Cisco Unified CM, several mechanisms can be used to prevent toll fraud. Partitions and calling search spaces (CSS) provide segmentation and access control to the directory number that can be called or the device or line that is placing the call. As a best practice, apply the most restrictive class of service possible (for example, no access to PSTN routes for calls coming in from the PSTN) based on partitions and calling search spaces. Other mechanisms can also be used, such as time-of-day routing, enabling the *Block OffNet to OffNet Transfer* service parameter, forced authentication code (FAC), and route filters.

On Cisco Expressway-E, use Call Processing Language (CPL) rules to block fraudulent attempts.

On Cisco Unified Border Element, configure protection mechanisms against toll fraud; for example, configure an IP trust list and explicit incoming and outgoing dial peers.

Certificate Recommendations

Simplify certificate management with Certification Authority (CA) signed certificates. By default, server certificates are self-signed certificates. To establish trust with a service based on a self-signed certificate, the self-signed certificates must be imported into the trust store of all entities requiring secure connections to the service. If the certificates are not imported, the communication can fail or warning messages about the certificates might appear, as with Jabber for example. Importing certificates can be handled if the set of communicating parties is small, but it becomes more difficult for large numbers of communication peers. For this reason, we recommend having some of the certificates signed by a Certification Authority (CA) and extending trust to the CA. This is especially important for certificates such as the Tomcat certificates for Cisco Unified CM with IM and Presence and Cisco Unity Connection as well as the XMPP certificate for IM and Presence.

For Cisco Expressway-E servers, use certificates that are signed by a public CA.

Use multi-server certificates wherever possible, especially for the Cisco Unified CM and Unified CM IM and Presence Tomcat certificates. Multi-server certificates allow the administrator to assign a single certificate for a given service across multiple servers in a cluster in order to further simplify certificate management.

On the endpoints, in general, two types of certificates are available: Manufactured-Installed Certificate (MIC) and Local Significant Certificate (LSC). Endpoint certificates are used for encryption of the signaling and media and for the optional encryption of TFTP phone configuration files. We recommend using LSC certificates instead of MIC certificates.

Encryption Recommendations

Provide encryption for the following:

- SIP trunks
  SIP trunks are used between Cisco Unified CM and other servers such as Cisco Unity Connection, IM and Presence, Cisco Meeting Server, Unified Border Element, business-to-business (B2B) Collaboration Edge, and voice gateways.
- HTTP connections
  Use HTTPS instead of HTTP for all application connections. For example, use HTTPS with Extension Mobility.

With a Cisco Unified CM multi-cluster deployment, also enable encryption for:

- Intercluster Lookup Service (ILS)
- Location Bandwidth Manager (LBM)-to-LBM communication between clusters

To protect sensitive voice and video communications, enable endpoint encryption for signaling and media. This is especially important if your network is not entirely trusted and secure. This requires enabling mixed-mode in Unified CM. With mixed mode, you can select which endpoints are configured to use signaling and media encryption and which are not.

Bandwidth Management

Bandwidth management is about ensuring the best possible user experience end-to-end for all voice and video capable endpoints, clients, and applications in the Collaboration solution. The Cisco Preferred Architecture for Enterprise Collaboration provides a holistic approach to bandwidth management that incorporates an end-to-end Quality of Service (QoS) architecture, call admission control, and video rate adaptation and resiliency mechanisms to ensure the best possible user experience for deploying pervasive video over managed and unmanaged networks.

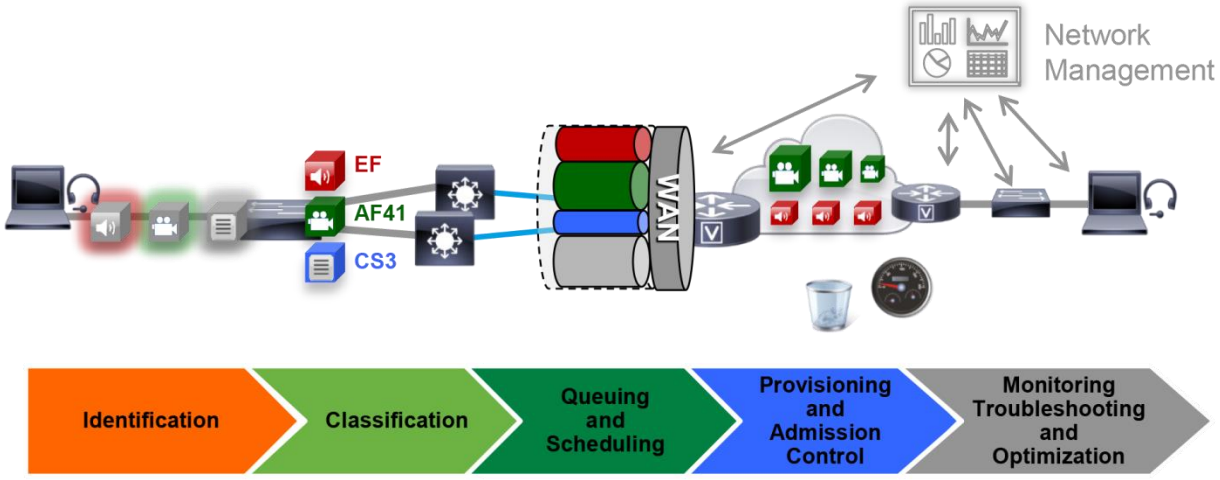
QoS Architecture for Collaboration

QoS ensures reliable, high-quality voice and video by reducing delay, packet loss, and jitter for media endpoints and applications. QoS provides a foundational network infrastructure technology that is required to support the transparent convergence of voice, video, and data networks. With the increasing amount of interactive applications – particularly voice, video, and immersive applications – real-time services are often required from the network. Because these resources are finite, they must be managed efficiently and effectively. If the number of flows contending for such priority resources were not limited, then as those resources become oversubscribed, the quality of all real-time traffic flows would degrade, eventually to the point of becoming useless. “Smart” media techniques, QoS, and admission control ensure that real-time applications and their related media do not over-subscribe the network and the bandwidth provisioned for those applications, thus ensuring efficient use of bandwidth resources. These smart media techniques coupled with QoS and, where needed, admission control can be a powerful set of tools to protect real-time media from non-real-time network traffic and to protect the network from oversubscription and the potential loss of user experience quality for all voice and video applications.

Figure 21 illustrates the approach to QoS used in the Cisco PA for Enterprise Collaboration. This approach consists of the following phases:

- **Identification and classification** — Refers to concepts of trust and techniques for identifying media and signaling for trusted and untrusted endpoints. It also includes the process of mapping the identified traffic to the correct DSCP to provide the media and signaling with the correct per-hop behavior end-to-end across the network for both trusted and untrusted endpoints.
- **Queuing and scheduling** — Consists of general WAN queuing and scheduling, the various types of queues, and recommendations for ensuring that collaboration media and signaling are correctly queued on egress to the WAN.
- **Provisioning and admission control** — Refers to provisioning the bandwidth in the network and determining the maximum bit rate that groups of endpoints will utilize. This is also where call admission control can be implemented in areas of the network where it is required.
- **Monitoring, troubleshooting, and optimization** — Ensures the proper operation and management of voice and video across the network. Cisco Prime Collaboration offers a suite of tools to perform these functions.

Figure 21 Architecture for Bandwidth Management



Recommended Deployment

- Identify traffic based on trusted and untrusted devices.
- Classify and mark traffic at the access switch edge.
  - Mark all audio with Expedited Forwarding class EF (includes all audio of voice-only and video calls).
  - Mark all critical desktop and room system video with an Assured Forwarding class of AF41.
  - Mark all Jabber, Mobile and Remote Access (MRA), and Edge video with an Assured Forwarding class of AF42.
  - Configure QoS on all media originating and terminating applications and MCUs across the solution.
- Apply simplified WAN Edge policies for identifying, classifying, marking, and queuing collaboration traffic:
  - WAN edge ingress re-marking policy
  - WAN edge egress queuing and scheduling policy
- Group video endpoints into classes of maximum video bit-rate to limit bandwidth consumption based on endpoint type and usage in the solution.
- Deploy Enhanced Locations Call Admission Control and limit calling based only in areas of the network where bandwidth resources are restricted.

Benefits

This deployment provides the following benefits:

- Prescriptive recommendations to simplify deployment with a simplified QoS architecture
- Makes more efficient use of network resources
- Supports mobile and multi-media Collaboration devices
- Takes into account “unmanaged” network segments (Internet)
- Is “future-proof” — facilitates introduction of new services, features, and endpoints

Deployment Best Practices

When video is deployed pervasively across the organization, bandwidth constraints typically limit the video resolution that can be achieved during the busiest hour of the day, based on the bandwidth available and the number of video calls during that busy hour. While it is possible to buy more bandwidth in some places of the network, it is not always possible in all places of the network, nor is it cost effective to buy bandwidth that is used only during the busy hour but remains idle during the rest of the operational day.

To address this challenge, we have targeted a group of endpoints and call flows in the Preferred Architecture and have created a strategy of QoS marking and queuing to allow the video endpoints and flows to be more opportunistic in their utilization of video in the network. This concept of opportunistic video is about achieving the best video quality based on the WAN bandwidth resources available at any given time for a determined group of devices and call flows. In the Preferred Architecture all Cisco Jabber clients as well as any Collaboration Edge call flows such as mobile and remote access or business-to-business call flows are “classified” as opportunistic video flows. During the busy hour with higher call volumes, more video flows are expected over the constrained areas of the network. Rather than using call admission control to deny the video calls, we have created a QoS marking and queuing policy that forces opportunistic video flows to rate adapt down when packet loss occurs in that class. Using a single class video queue approach with DSCP-based Weighted Random Early Detection (WRED), we are able to protect prioritized video from packet loss during periods of congestion over opportunistic video. With this single video queue approach, when prioritized video is not using bandwidth in the queue, opportunistic video gains full access to the entire queue bandwidth if and when needed, or it rate adapts down to squeeze more opportunistic video flows into the queue without affecting the prioritized video flows. This is a significant point when deploying pervasive video with Jabber and Collaboration Edge technologies. This frees up more bandwidth for more video flows while still protecting prioritized video flows.

Another consideration taken in the Preferred Architecture is around the QoS of an audio stream of a video call, and how it has traditionally been marked with the same QoS as the video stream of the video call. This approach, however, has two deficiencies:

- The audio stream of a video call can be impacted by packet loss in the video queue.
- Audio stream classification for untrusted devices cannot be distinguished between voice-only calls and video calls.

In the Preferred Architecture we have designed a solution to address these deficiencies by ensuring that all audio is marked with a single value of Expedited Forwarding (EF) across the solution. In this way, whether the audio stream is associated with a voice-only call or a video call, it is always marked to the same value. Thus, the audio stream of a video call will be prioritized above the video and not subject to any packet loss in the video queue. This also solves the identification issue with untrusted devices such as Jabber clients. Because the marking of the client is not trusted by the network access layer, there is no effective way to distinguish the audio stream of a voice-only call from the audio of a video call in the network. Thus, moving to this new model where all audio is marked with a single value simplifies the network prioritization and treatment of the traffic.

Another consideration when the audio and video of a video call are not using the same QoS marking, is the fact that the audio could arrive at the terminating endpoint with less delay than video. This is because audio is prioritized higher and sent into the queues earlier than video traffic. To ensure that there are no lip-sync issues between the audio and video, enabling Real-time Transport Control Protocol (RTCP) on all endpoints is a simple yet strict requirement. RTCP uses timestamps to synchronize audio and video, and thus resolves any lip-sync issues that could arise from a delay variation between audio and video of the same video call.

In some areas of the network, bandwidth is too constrained even for the above strategy, in which case call admission control is the only possible way to ensure that over-subscription of bandwidth resources does not occur in those areas of the network. Enhanced Locations Call Admission Control can be deployed to protect the WAN resources, thus ensuring that any allowed call flows have the bandwidth to proceed without packet loss. If admission control fails a call flow, Unified CM has the ability to find alternative call routing paths if available.

Appendix

Product List

This product list identifies the Cisco products in the Preferred Architecture for Enterprise Collaboration, along with their recommended software versions.

Product	Product Description	Recommended Software Version
Cisco Unified Communications Manager and IM and Presence Service	Call control, instant messaging, and presence services	11.5(1)SU2
Cisco Unity Connection	Voicemail services	11.5(1)SU2
Cisco Expressway-C and Expressway-E	Mobile and remote access and business-to-business communications	X8.9(1)
Cisco Prime License Manager	Single management point for licensing	11.5(1)SU1a
Cisco Prime Collaboration Deployment	Installs Unified CM cluster with IM and Presence Service and Unity Connection cluster	11.6(1)
Cisco Prime Collaboration Provisioning	Configures Unified CM and other applications; provisions users and devices; handles moves, adds, and changes	11.6
Cisco Meeting Server	Audio and video conferencing and resource management	2.1
Cisco ISR and ASR	PSTN gateway, SRST, and external connectivity to the Internet	IOS 15.6(3)M1 for ISR IOS XE 16.3(2) for ASR
Cisco Unified IP Phone 7811	General office use, single-line phone	11.7(1)
Cisco Unified IP Phone 8800 Series	General office use	11.7(1)
Cisco Unified IP Phone 8831	IP conference phone	10.3(1)
Cisco Jabber	Soft client with integrated voice, video, voicemail, and instant messaging and presence functionality for mobile devices and personal computers	Jabber 11.8
Cisco TelePresence DX Series	Personal TelePresence endpoint for the desktop	CE 8.3(1)
Cisco TelePresence MX Series	TelePresence multipurpose room endpoint	CE 8.3(1)
Cisco TelePresence SX Series	Integrator Series TelePresence endpoint	CE 8.3(1)
Cisco TelePresence IX Series	Immersive TelePresence room system	IX 8.2
Cisco TelePresence Management Suite (TMS)	Scheduling, web conferencing integration, and other advanced video features	15.4

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