Cisco Preferred Architecture for Midmarket Voice 11.*x*

Design Overview

June 14, 2016

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Preface

Cisco Preferred Architectures provide recommended deployment models for specific market segments based on common organizational use cases. They incorporate a subset of products from the total 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.

Getting Started with 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 sales processes.
- Cisco Validated Design (CVD) guides provide detailed steps for deploying components within the Cisco Preferred Architectures. These guides support planning, deployment, and implementation (PDI).
- Cisco Collaboration 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.

About This Guide

This *Cisco Preferred Architecture for Midmarket Voice Design Overview* is for:

- Sales teams that design and sell voice communications environments
- Customers and sales teams who want to understand the overall Cisco Voice 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 Voice portfolio that are built for the midmarket and that provide appropriate feature sets for this market
- Detailing a midmarket voice architecture and identifying general best practices for deploying in midmarket organizations

For detailed information about planning, deploying, and implementing this architecture, see the following CVDs:

- Unified Communications Using Cisco Business Edition 6000 CVD
- Help Desk Using Cisco UCCX CVD
- Collaboration Edge Using Cisco Business Edition 6000 CVD

Introduction

The business need for telephony service is not new, and many would even suggest that using Internet Protocol (IP) telephony instead of Time-Division Multiplexing (TDM) has entered mainstream as well. Positioning a telephony solution that interoperates seamlessly with other communication and collaboration tools can bring great value to an organization.

The increase in workplace mobility and changes in end user devices now require organizations to extend telephony to where the workforce is located, often extending beyond the walls of the organization's physical buildings. Additionally, new smart phones, social media, and personal communication applications have been developed and widely adopted by individuals in their personal lives.

With advancements in collaboration applications and adoption of new technology in their personal lives, individuals are not only willing to adopt new collaboration tools in their workplaces, but also starting to expect them. Organizations can now feel comfortable providing communication applications that employees will quickly adopt and that provide maximum value. These new tools enhance an organization's overall business process, make its employees more productive, and open the door to new and innovative ways for communicating with business partners and customers.

Technology Use Cases

Organizations want to streamline their business processes, optimize employee productivity, and integrate voice services with business processes. The Cisco Preferred Architecture (PA) for Midmarket Voice delivers capabilities that enable organizations to realize immediate gains in productivity and add value to their current voice deployments. Additionally, the following technology use cases offer organizations enhanced capabilities for mobile and remote workers, while delivering even more value in these areas:

- Support Teleworkers and Branch Offices — Let employees work from multiple locations, whether satellite offices, home offices, or when traveling.
- Consolidate Communications Infrastructure — Bring together voice and data into a single IP network to simplify management and support effective communications.

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

Architectural Overview

The Cisco PA for Midmarket Voice provides an end-to-end voice communications deployment for up to 1,000 users. This architecture ensures high availability for critical applications and uses products developed and priced for the midmarket. The consistent user experience provided by the overall architecture facilitates quick user adoption, enabling an organization to recognize immediate value for its investment. Additionally, the architecture supports an advanced set of services that extend to mobile workers, partners, and customers through the following key services:

- Voice communications
- Instant messaging and presence
- Voice conferencing
- Enablement of mobile and remote workers
- Unified voice messaging
- Customer care

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 voice calls. In general, it is a best practice to ensure a voice solution is deployed with proper Quality of Service (QoS) configured throughout the network. Voice IP traffic should be classified and prioritized to preserve the user experience and avoid negative effects such as delay, loss, and jitter. For more information about LAN and WAN QoS, see the latest version of the Cisco Collaboration SRND.

The Cisco PA for Midmarket Voice, shown in Figure 1, provides highly available and secure centralized services. These services extend easily to remote offices and mobile workers, ensuring availability of critical services even if communication to headquarters is lost. Centralizing services also simplifies management and administration of an organization's voice deployment.

Figure 1 Cisco Preferred Architecture for Midmarket Voice

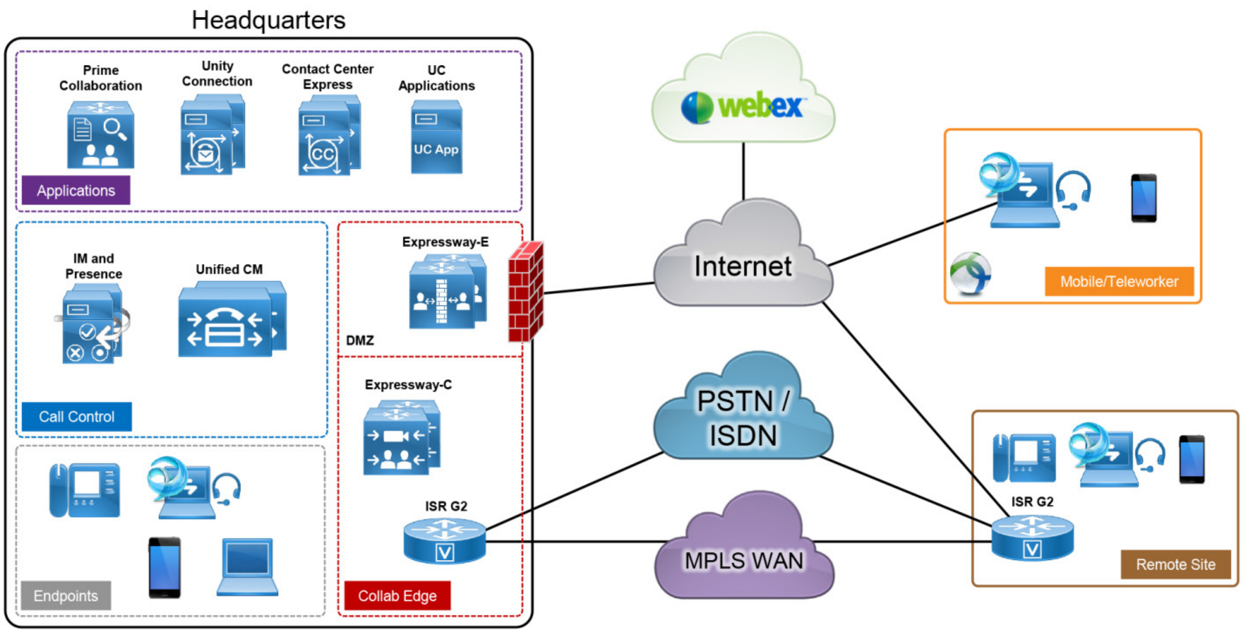


Table 1 lists the products in this architecture. For simplicity, products are grouped into modules to help categorize and define their roles. The content in this guide is organized in the same modules.

Table 1 Components for the Cisco Preferred Architecture for Midmarket Voice

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 Unified IP Phones and Cisco Jabber	Enable real-time voice, video, and instant messaging communications for users
Conferencing	Cisco ISR	Provides voice conferencing resources
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
	Cisco ISR	Provides either public switched telephone network (PSTN) or Cisco Unified Border Element (CUBE) connectivity
Applications	Cisco Unity Connection	Provides unified messaging and voicemail services
	Cisco Unified Contact Center Express (Unified CCX)	Provides customer interaction management services
	Cisco Prime Collaboration Provisioning Standard	Provides administrative functions (provisioning) for Cisco Unified Communications applications

Cisco Business Edition 6000

The Cisco Business Edition (BE) 6000 is a package system designed specifically for organizations with up to 1,000 users, and it is the foundation for this architecture. Cisco BE6000 is built on a virtualized Cisco Unified Computing System (UCS) that is prepared and ready for use with a preinstalled virtualization hypervisor and application installation files. The Cisco BE6000 solution offers premium voice, video, messaging, instant messaging and presence, and contact center features on a single integrated platform. For these reasons the BE6000 is an ideal platform for the Cisco PA for Midmarket Voice.

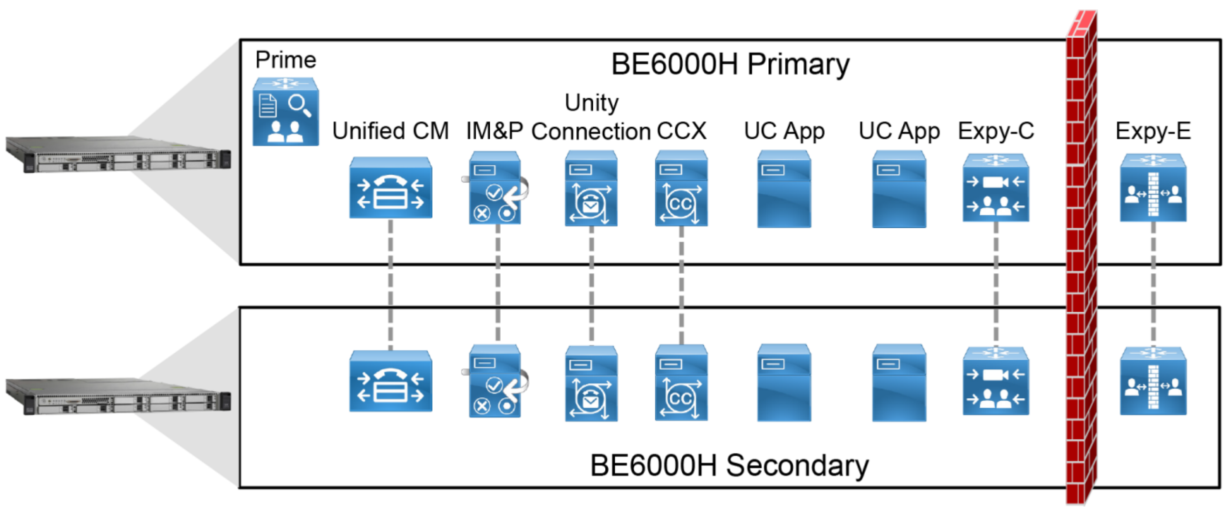
Cisco Business Edition 6000S can be used for entry-level centralized Unified Communications solutions for up to 150 users. BE6000S is a single box solution built on the Cisco 2921 Integrated Services Router with Cisco Unified Computing System (UCS) E-Series blade supporting five Unified Communications applications.

For more information about Cisco BE6000, consult the data sheet.

Core Applications

The Cisco PA for Midmarket Voice is built on two Cisco BE6000H high-density servers to provide high availability for applications within the architecture (Figure 2). Virtualizing multiple applications on a single server 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.

Figure 2 Cisco Preferred Architecture for Midmarket Voice Deployed on Cisco BE6000H



In this architecture, the following applications and Cisco Prime Collaboration Provisioning Standard are deployed on one Cisco BE6000H server, while a second instance of most applications is deployed on a second Cisco BE6000H server, providing hardware and software redundancy for:

- 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 Unified Contact Center Express

BE6000S does not support Cisco Expressway or Cisco Unified Contact Center Express applications.

Cisco recommends always deploying redundant configurations to provide the highest availability for critical business applications; however, a non-redundant Cisco BE6000H server configuration may be deployed for organizations that do not require full redundancy.

Note: Space is available on the Cisco BE6000H for additional Cisco Unified Communications applications. Use the Collaboration Virtual Machine Placement Tool to determine virtual machine placement on the BE6000H servers.

High Availability

The Cisco PA for Midmarket Voice 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, 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 Midmarket Voice provides high availability through the use of redundant power supplies, network connectivity, and disk arrays.

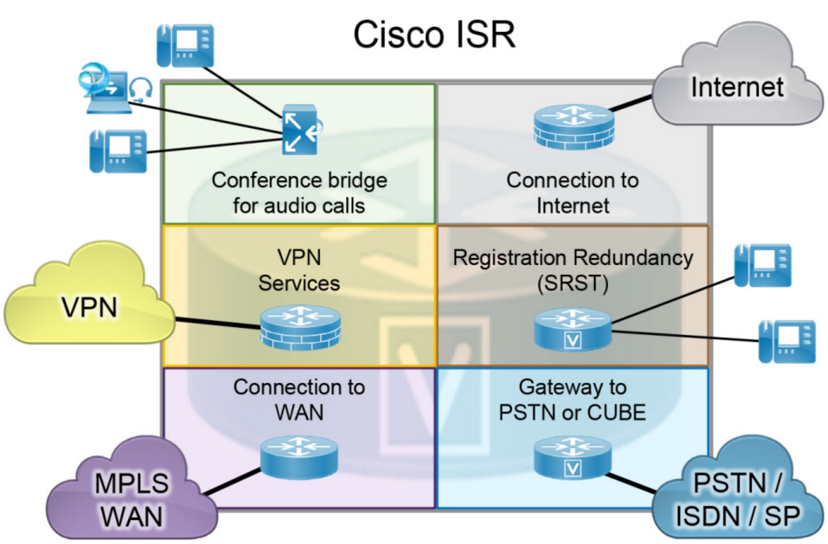
To provide high availability with BE6000S, deploy SRST at the branch site.

Cisco Integrated Services Router

Cisco Integrated Services Routers (ISRs) provide Wide Area Network (WAN) and Cisco Unified Communications services in a single platform. In the Cisco PA for Midmarket Voice, the Cisco ISR can provide the following functions (Figure 3):

- Audio conference bridge for Cisco Unified CM
- External connectivity to the Internet
- IP routing and network services such as DHCP, DNS, NTP, and others
- Cisco Unified Survivable Remote Site Telephony (SRST) to provide 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 an organization's network
- Virtual Private Network (VPN) client to establish secure tunnels to a VPN concentrator

Figure 3 Cisco ISR Functions



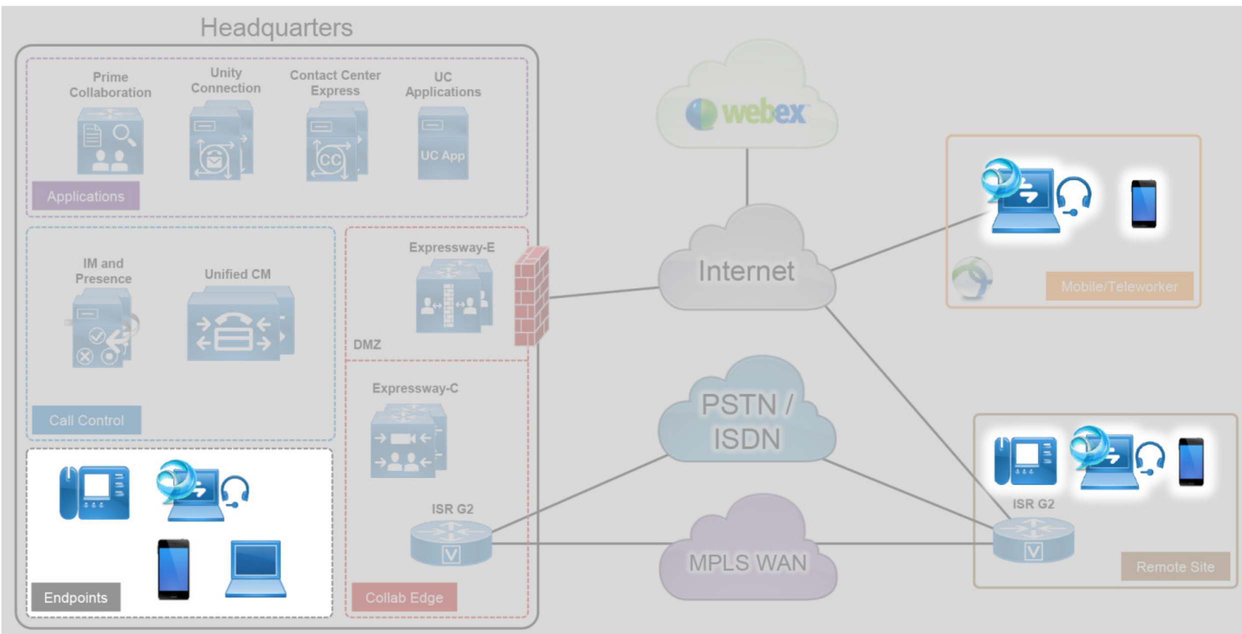
The Cisco ISR has additional slots that support add-on modules such as wireless controllers and VMware ESXi servers. Deployments can use various Cisco ISR models to support different features, to scale, and to accommodate additional services. The modular design enables the Cisco ISR to be deployed at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

Endpoints

Cisco Collaboration endpoints provide a wide range of features, functionality, and user experiences. Cisco endpoints range from low-cost, single-line phones and soft clients to high-end, multiline phones with color displays, allowing an organization to 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
- Conferencing
- Voicemail
- Presence
- Instant messages (Cisco Jabber)

Figure 4 Architecture for Endpoints



Recommended Deployment

Cisco Unified CM is the call control server for the Cisco PA for Midmarket Voice. Cisco IP Phones and Jabber clients 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.

Cisco recommends 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 Unified IP Phone 6901 or Cisco IP Phone 7821	Public space phone
Cisco IP Phone 8800 Series	General office use, multiple-line phone
Cisco Unified IP Conference Phone 8831	IP conference phone

Table 3 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, voicemail, instant messaging, and presence functionality for mobile devices and personal computers

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 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 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 and presence.

Figure 5 Architecture for Call Control

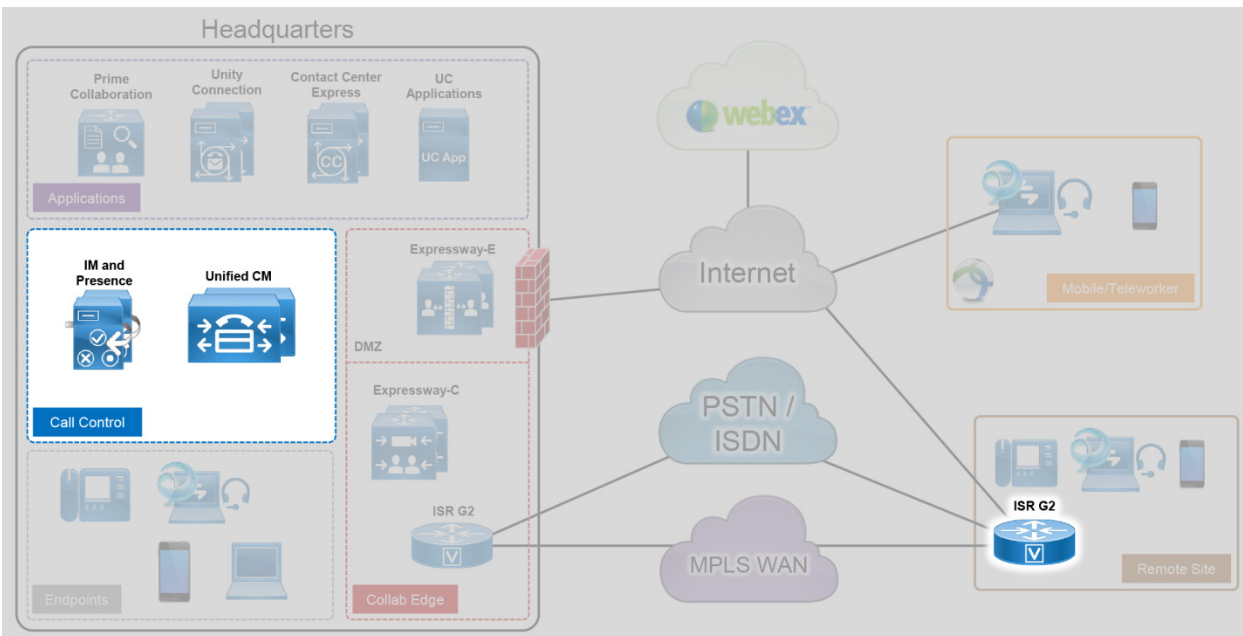


Table 4 lists the roles of the call control components in this architecture and the services they provide.

Table 4 Components for Call Control

Module	Component	Description
Call Control	Cisco Unified Communications Manager (Unified CM)	Provides call routing and services, dial plan, bandwidth management, device-based presence, and enables Cisco Jabber desk phone control
	Cisco Unified Communications Manager IM and Presence Service	Provides Cisco Jabber support for instant messaging and 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 two Cisco Unified CM servers in a cluster configuration that includes a publisher node and a subscriber node for redundancy.
- Deploy two IM and Presence Service servers in a cluster configuration that includes a publisher node and a subscriber node for redundancy.
- Enable Cisco SRST on the Cisco ISR as a backup service at remote sites to provide high availability. Cisco SRST is recommended for BE6000S deployments for high availability.

Note: If full redundancy is not required, a single server may be deployed without loss of functionality.

Benefits

This deployment provides the following benefits:

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

Deployment Best Practices

Cisco Unified Communications Manager and IM and Presence Service

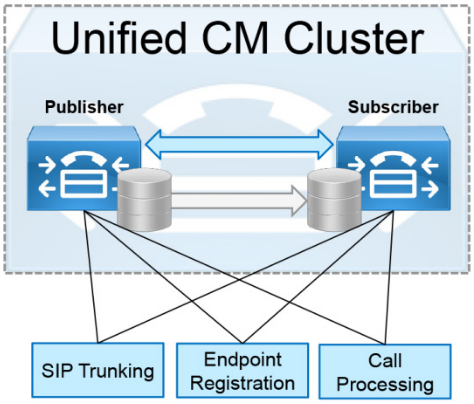
Publisher-Subscriber Deployment Model

A Cisco Unified CM cluster or an IM and Presence Service cluster consists of one publisher node and one subscriber node (Figure 6).

- The publisher node is the server that is installed first. This server contains the cluster's configuration database. Cluster-wide configuration is written to the publisher's database and replicated on the subscriber.
- The subscriber node is the server that is installed second. It contains a replica of the publisher's database. The subscriber is updated automatically whenever the publisher's configuration changes.

Clustering provides an automatic redundancy mechanism for endpoints and for Cisco Unified CM services, such as the ability to receive and process incoming calls. Cisco recommends configuring the Unified CM cluster with the subscriber node as the primary call-processing server and the publisher node as the backup call-processing server. This recommended configuration also applies to the IM and Presence Service cluster. If the IM and Presence Service subscriber node goes down, then IM and presence capabilities will still be available for Cisco Jabber clients.

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 Midmarket Voice, 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.

Configure a SIP trunk from the Cisco Unified CM cluster to external components in the deployment, such as the IM and Presence Service. Specify each server for the external component as a destination in the SIP trunk configuration. This configuration provides continuation of services if a node goes down.

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

Dial Plan

A structured, well-designed dial plan is essential for 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. 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 a 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 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 format 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. 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.

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 recommended because it allows a single location for directory management. Additionally, LDAP directory integration 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 service supported by a dial plan depends on the granularity and complexity of the classes. For more information about classes of service and details on dial plan design, see the latest version of the Cisco Collaboration SRND.

Admission Control

Call admission control (CAC) mitigates congestion on WAN links due to excessive call traffic. In cases where the administrator needs to control how many media calls flow over various links in the WAN topology, Cisco Unified CM Enhanced Location Call Admission Control (ELCAC) provides a solution. ELCAC supports various WAN topologies and gives the administrator the ability to statically model the WAN network topology to support admission control for calls. Cisco Unified CM uses locations and links to model how the WAN network topology routes media for calls between groups of endpoints within a location.

Conferencing

The ability for three or more people to communicate in real time is a core component of voice deployments. Conferencing builds upon existing infrastructure in place for point-to-point calls, offering an organization's users a consistent experience regardless of how many participants are involved (Figure 7).

Figure 7 Architecture for Conferencing

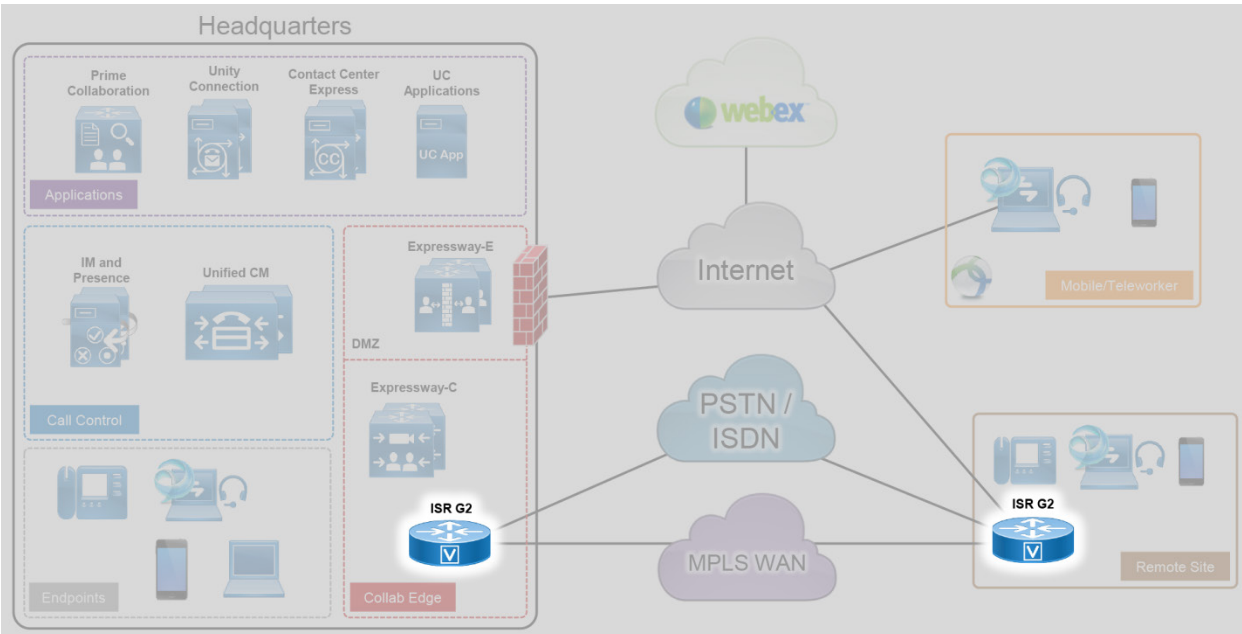


Table 5 lists the roles of the conferencing components in this architecture and the services they provide.

Table 5 Components for Conferencing

Module	Component	Description
Call Control	Cisco Unified CM	Manages and allocates audio conferencing resources from Cisco ISR packet voice/data modules (PVDMs)
Conferencing	Cisco ISR	Provides voice conferencing resources

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.

Note: The components required for scheduled conferences are not part of this architecture. However, Cisco WebEx Cloud hosted services can be used for scheduled audio conferencing. See the product data sheets for more information.

Recommended Deployment

Audio Conferencing

Deploy packet voice digital signal processor module (PVDM) resources on Cisco ISR routers for instant conferences and permanent conferences.

Benefits

This deployment provides the following benefits:

- There is a consistent user experience for launching and joining conferences.
- Dedicated resources (PVDMs) enable greater scale and redundancy.

Deployment Best Practices

Audio Conferencing

For instant and permanent audio conferences, use dedicated packet voice digital signal processor module (PVDM) resources as the audio conference bridges. A Cisco ISR with PVDM(s) is recommended. The Cisco ISR requires a PVDM to support audio conferences, voice interfaces (T1, E1, FXO, FXS), and audio transcoding.

Using Cisco ISRs for a variety of functions such as voice gateway, SRST, conferencing, and WAN connectivity, and combining these voice services into a single platform offers a significant cost savings over individual components. For additional deployment flexibility, PVDMs are available in different densities and support a range of codecs of different complexities.

Sizing Considerations

The decision to integrate conferencing resources into an existing router depends on the voice capacity and overall performance of that router. A standalone gateway is recommended if an organization's existing router:

- Consistently runs above 40% CPU — A standalone gateway avoids voice traffic processing delays.
- Has limited slots available for voice interface cards or digital signal processors — A standalone gateway ensures that additional capacity is available when needed.

Table 6 lists the recommended Cisco ISR platforms and number of audio conference ports to support instant and permanent audio conferences. Use this information as the starting point for deployment planning.

Table 6 Audio Conference Port Recommendations

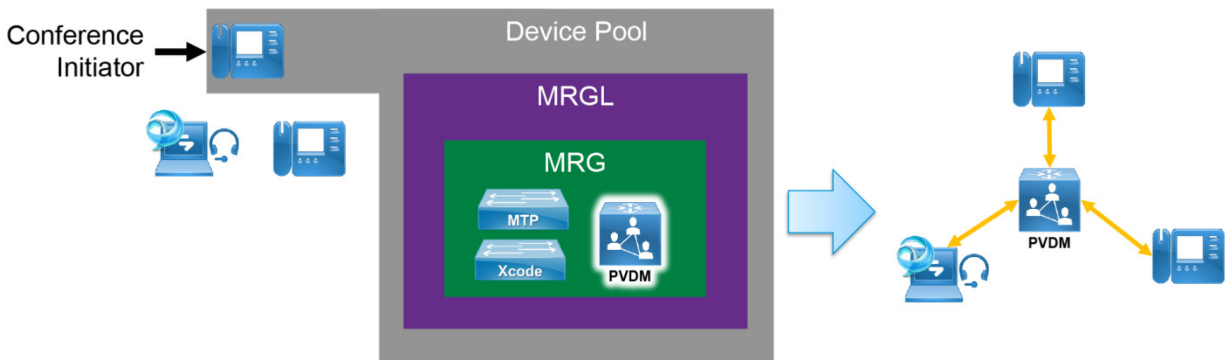
Site Size	Cisco ISR	Audio Conference Ports
50 users	Cisco 2911	25
100 users	Cisco 2921	50
250 users	Cisco 2951	75
750 users	Cisco 3925	100
1,000 users	Cisco 3945	150

Architecture

Permanent audio conferences rely on Cisco Unified CM's Meet-Me feature. This feature requires a set of directory numbers (DNs) to be allocated exclusively for permanent audio conferences. Users invoke the feature by pressing the Meet-Me softkey on their audio endpoints and then dialing DNs within the predetermined range. Subsequent attendees dial the predetermined number directly to join the conference. The existing dial plan is used for controlling access to these DNs. Permanent audio conferences are hosted on the same Cisco ISR PVDM resources as instant audio conferences.

Instant audio conference resources register with Cisco Unified CM and are controlled by media resource group lists (MRGLs) and media resource groups (MRGs). Endpoints invoke these resources if their assigned device pool has access to the appropriate MRGL. Configuring MRGLs is recommended so that conference resources that are local to the initiating endpoint are preferred over other resources (Figure 8).

Figure 8 Media Resource Group List (MRGL) Example



Collaboration Edge

In addition to connectivity to the public telephone system, organizations are looking for ways to enable mobile and remote workers. Providing secure connectivity for mobile and remote workers enables employees to be more productive away from the office. The Cisco PA for Midmarket Voice addresses these needs with the Collaboration Edge architecture illustrated in Figure 9.

Figure 9 Architecture for Collaboration Edge

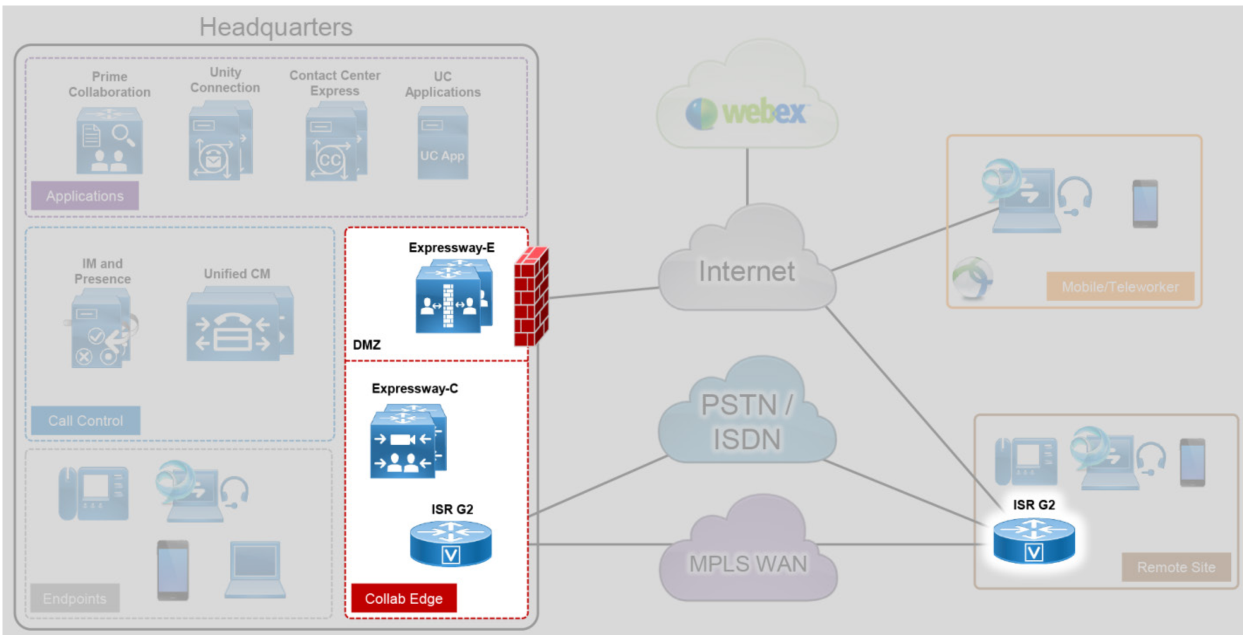


Table 7 lists the roles of the Collaboration Edge components in this architecture and the services they provide.

Table 7 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 and Jabber clients. The traversal server resides in the DMZ.
	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.
	Cisco ISR	Enables local PSTN connectivity.
		Enables connectivity from an organization's network to the service provider network for SIP trunks via Cisco Unified Border Element (CUBE).

Recommended Deployment

Headquarters

- Deploy Cisco ISR as the PSTN gateway, or enable Cisco Unified Border Element (CUBE) functionality on the Cisco ISR for voice connectivity from the organization's network to the service provider network through a SIP trunk.
- Deploy Cisco Expressway-C and Expressway-E servers to enable remote Jabber endpoint registrations, voice calling, IM and presence services, voicemail services, and directory services. Expressway-C and Expressway-E servers should be deployed in separate clustered pairs for redundancy. A BE6000S deployment does not support the Collaboration Edge.

Note: If full redundancy is not required, a single Expressway-C server and a single Expressway-E server may be deployed without loss of functionality.

Remote Sites

- Deploy Cisco ISR as the PSTN gateway.

Teleworker Sites

- Deploy Cisco IP Phone 8800 Series phones with an existing Cisco Adaptive Security Appliance (ASA) or Cisco IOS-based VPN.

Benefits

This deployment provides the following benefits:

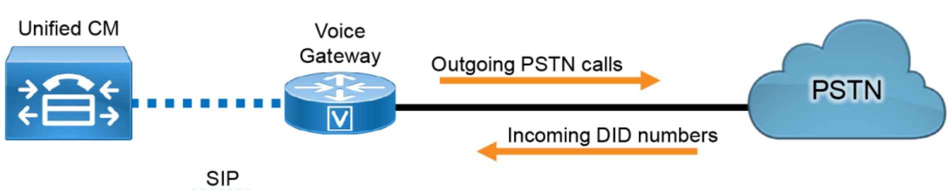
- The Cisco ISR supports standards-based interfaces and various PSTN types, so it can be deployed globally.
- Instead of traditional PSTN interfaces, CUBE functionality can be enabled on the Cisco ISR if a SIP trunk is used.
- The Cisco ISR can also be used for WAN connectivity.
- Cisco Expressway provides calling, presence, instant messaging, voicemail, and corporate directory services for Cisco Jabber.

Deployment Best Practices

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 10).

Figure 10 PSTN Connectivity



Use Cisco ISR with a time-division multiplexing (TDM) module as the PSTN gateway at headquarters. This configuration enables the gateway to route calls to and from the PSTN. Use this gateway for number transformation to PSTN requirements. See the Call Control section for dial plan recommendations.

At remote sites, deploy a Cisco ISR for local PSTN connectivity using voice modules. Table 8 lists the recommended Cisco ISR series by deployment location. For information about Cisco ISR, see the data sheet.

Table 8 Recommended PSTN Gateways

Location	PSTN Gateway
Headquarters	ISR G2 3900 series
Remote Sites	ISR G2 2900 series

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.

Virtual Private Network

A virtual private network (VPN) enables an organization's network to be extended to remote and teleworker sites, allowing those sites full access to corporate services. Use the organization's existing VPN to extend collaboration services to teleworkers, or use Cisco IP phones with VPN clients. The Cisco IP Phone 8800 Series phones have a built-in VPN client that can connect to the Cisco ASA or Cisco ISR over the Internet and establish a secure connection to access the organization's network. No additional device is needed for this connectivity, and these Cisco IP phones are recommended for teleworker sites.

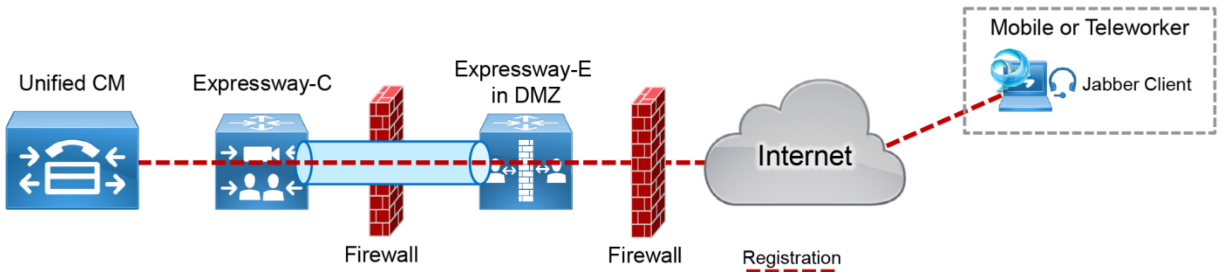
Cisco Expressway

Cisco Expressway provides secure firewall and NAT traversal for mobile Cisco Jabber endpoints (Figure 11). 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 appliance (for example, Cisco Expressway CE500 or CE1000) can be deployed based on the company's security requirements.

Figure 11 Traversal for Registrations 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
- Enable mobile and remote access capabilities and call signaling for Cisco Jabber endpoints, directing them to Cisco Unified CM for SIP registration and/or the IM and Presence Service as well as voicemail and directory services.

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
- 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

Licensing

Cisco Expressway can be used for mobile and remote access with no additional licensing requirement.

Mobile and Remote Access

The mobile and remote access feature enables Jabber clients to register securely to Cisco Unified CM through Expressway-E and Expressway-C without a VPN. A Jabber client can send and receive several types of collaboration flows (audio, instant messaging, presence, voicemail, and directory). When multiple edges are deployed, Cisco recommends using Geo-DNS services to provide the best network option based on assigning the closest Expressway in the DNS response.

Applications

In addition to the call processing and media resource components, the Cisco PA for Midmarket Voice includes the following Cisco applications to enhance usability, functionality, and management (Figure 12):

- Unity Connection to provide messaging (voicemail)
- Unified Contact Center Express (CCX) to provide sophistication to customer care for the organization in an easy-to-deploy fashion
- Prime Collaboration Provisioning Standard for user and device provisioning

Figure 12 Architecture for Applications

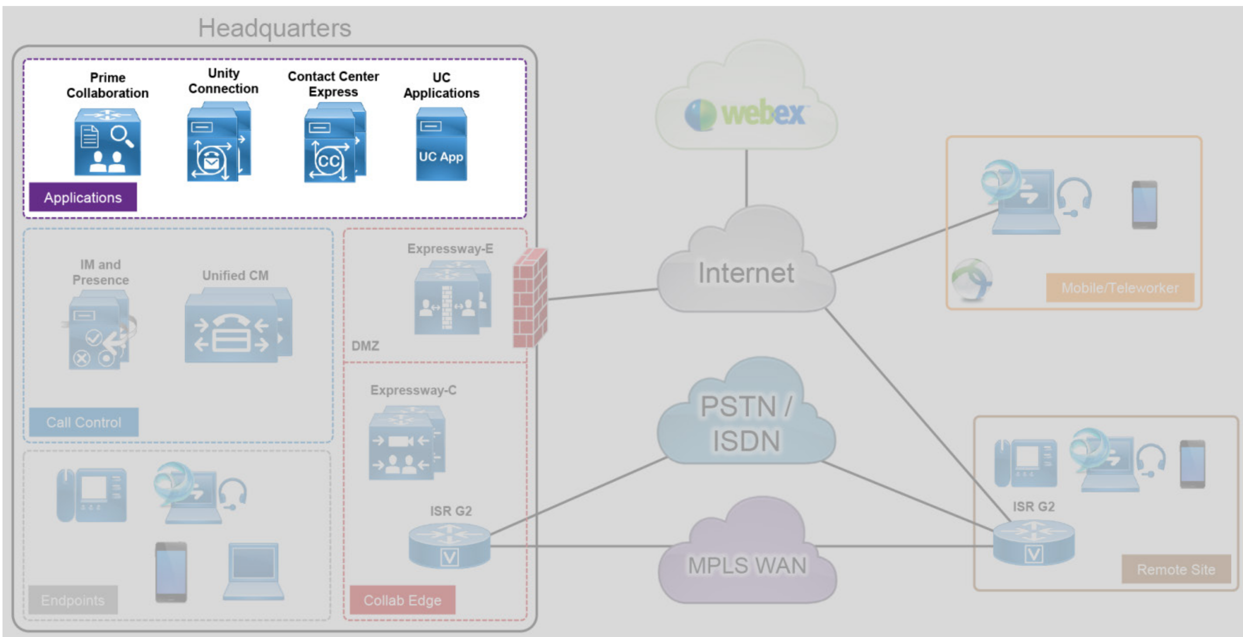


Table 9 lists the roles of the application components in this architecture and the services they provide.

Table 9 Components for Applications

Module	Component	Description
Applications	Cisco Unity Connection	Provides voicemail services
	Cisco Unified Contact Center Express (CCX)	Provides customer interaction and interactive voice response (IVR) services
	Cisco Prime Collaboration Provisioning Standard	Provides user and service provisioning

Cisco Unity Connection

Cisco Unity Connection enables users to access and manage voice messages in a variety of ways, such as by email inbox, web browser, Cisco Jabber, Cisco IP Phone, smartphone, tablet, and many more. Users can interact with Unity Connection through the visual voicemail feature, phone keypad keys, or voice commands that they speak into the phone handset, headset, or speakerphone.

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

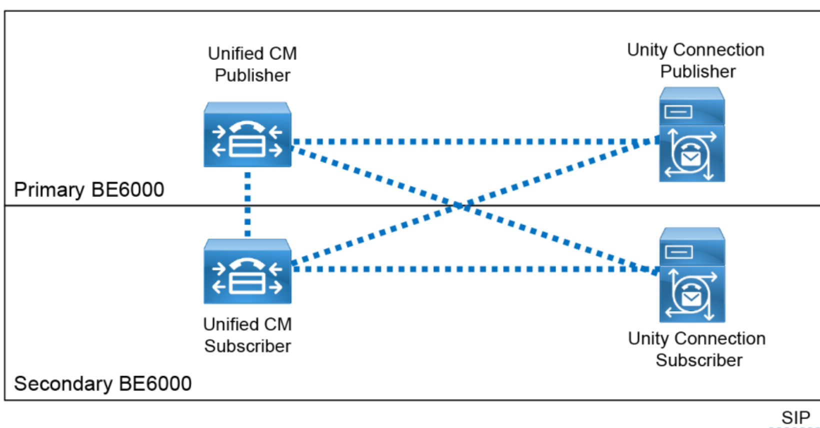
- Users can access the voicemail system and retrieve their voice messages by using their IP phones, mobile devices, and various email client applications with a dialed number, a SIP URI, or the visual voicemail feature.
- Cisco Unity Connection allows users to customize personal settings from a web browser.
- 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 commands.

Note: If full redundancy is not required, a single server may be deployed without loss of functionality.

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 13). If one Unity Connection node 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 13 Cisco Unity Connection Cluster



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 for each individual user separately.

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 user to access voicemail using the graphical interface on the IP phone or Jabber client. 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 including 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.

Cisco Unified Contact Center Express

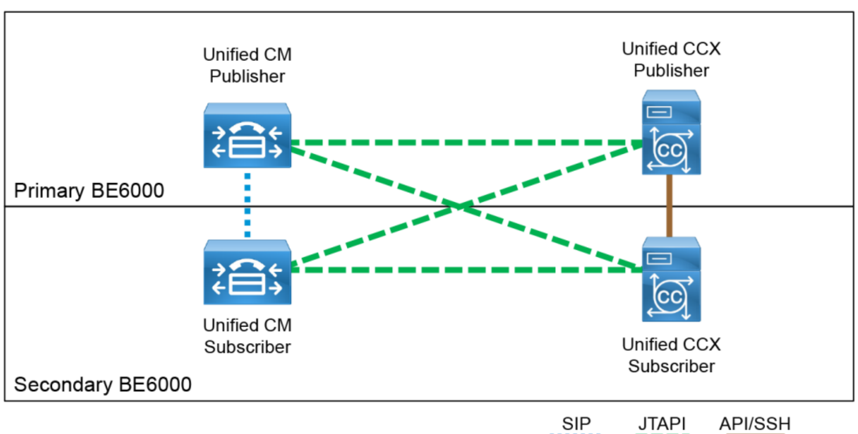
Cisco Unified Contact Center Express (CCX) enables organizations to provide powerful agent queuing and interactive voice response (IVR) services to internal and external customers. These services enable customers to connect easily with the right employees in an organization for sales inquiries or product support.

Recommended Deployment

Deploy two Unified CCX servers for high availability, with one active node and one standby node to handle failover in case of component failure (Figure 14). Also configure a primary and a backup Cisco BE6000 server for the JTAPI interface of the Telephony and Resource Manager-Contact Manager (RmCm) subsystems in Unified CCX. Cisco BE6000S does not support Unified CCX deployments.

Note: If full redundancy is not required, a single server may be deployed without loss of functionality.

Figure 14 Cisco Unified Contact Center Express Cluster



For contact center deployments, use Cisco Finesse as the agent and supervisor desktop. Cisco Finesse is a browser-based application implemented through a Web 2.0 interface with no client-side installation required, and it is highly customizable. In addition, Cisco Finesse supports +E.164, which adheres to the dial plan design recommendations discussed in the Call Control section.

Benefits

This deployment provides the following benefits:

- Recorded greetings and customized prompts provide sophisticated call handling.
- Unified CCX supports external customer interaction.
- Unified CCX facilitates internal company communications for activities such as help desk.

Deployment Best Practices

As with the other components in the Cisco PA for Midmarket Voice, Unified CCX should be deployed with high availability that includes active and standby nodes. Unified CCX downloads the end-user information from Unified CM that is synchronized with the organization's LDAP directory. This minimal configuration enables external callers to dial a single number into the organization and then use simple dial-by-name or dial-by-extension functionality without the need for telephone operators to connect external calls. Depending on the organization's structure and business model, Unified CCX could also be used for the following additional workflow functions:

- Sales
- Customer support
- Internal IT helpdesk
- Human resources

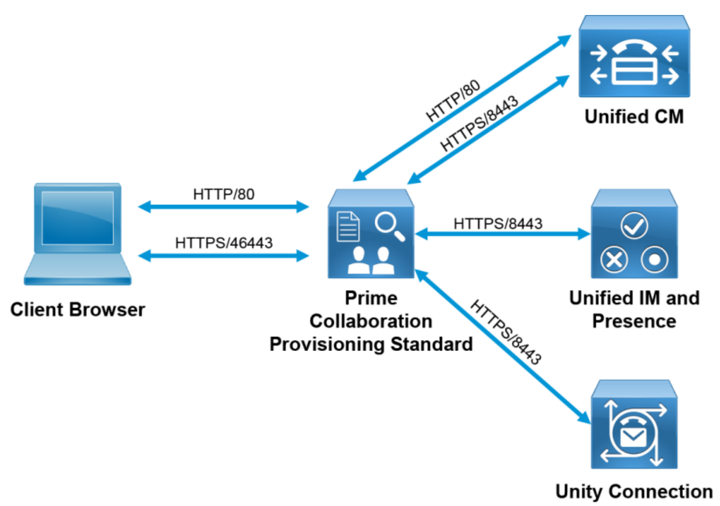
These automated, call-directed workflows provide value to the organization by quickly and easily connecting a person with a need to the appropriate resource within the organization for assistance.

For information about Cisco Unified Contact Center Express, see the latest data sheet.

Cisco Prime Collaboration Provisioning Standard

Cisco Prime Collaboration Provisioning Standard provides a centralized provisioning interface that simplifies administration of day-to-day activities such as moves, adds, changes, and deletions (MACD) of user devices and services in an organization (Figure 15).

Figure 15 Cisco Prime Collaboration Provisioning Standard



Recommended Deployment

Deploy Cisco Prime Collaboration Provisioning Standard on the primary BE6000 server. A single instance of Cisco Prime Collaboration Provisioning Standard is supported per organization.

Benefits

- A consistent, unified approach simplifies the management of multiple Cisco collaboration technologies such as Cisco IP Phones, Cisco Unified CM, and other application servers.
- Features such as bulk-based provisioning, phone MACDs, and consolidated views simplify user and service-related configuration and administration.
- A self-service portal eases support by enabling users to make authorized changes.
- The Getting Started Wizard (GSW) can quickly set up the Unified Communications applications, provision users, and integrate Unified Communications applications.

Appendix

Product List

This product list identifies the Cisco products in the Preferred Architecture for Midmarket Voice, along with their relevant software versions.

Product	Product Description	Software Version
Cisco Unified Communications Manager and IM and Presence Service	Call control, instant messaging, and presence services	11.5(1)
Cisco Unity Connection	Voicemail services	11.5(1)
Cisco Expressway-C and Expressway-E	Mobile and remote access	X8.8
Cisco Contact Center Express	Customer interaction management services	11.5(1)
Cisco Prime Collaboration Provisioning Standard	Provisioning and monitoring services for voice deployments	11.5(1)
Cisco ISR G2	PSTN gateway, SRST, audio conference resource, external connectivity to the Internet	IOS 15.6.2T
Cisco IP Phone 7800 Series	General office use, multiple-line phone	11.0(1)
Cisco IP Phone 8800 Series	General office use with available VPN client	11.0(1.11)
Cisco Unified IP Conference Phone 8831	IP conference phone	10.3(1)sr2
Cisco Jabber[1]	Soft client with integrated voice, voicemail, and instant messaging and presence functionality for mobile devices and personal computers	Jabber for Windows: 11.6 Jabber for Mac: 11.5.2 Jabber for iPhone and iPad: 11.5 Jabber Android: 11.5

1. The minimum Cisco Jabber version required to support Cisco Expressway mobile and remote access capabilities is 9.6 or 9.7, depending on platform. Consult the Jabber product release notes for specific minimum version support.

Ordering Information

For more information on ordering BE6000, see the Cisco Business Edition 6000 Ordering Guide. For pricing guidance on an optimal solution, including the best software licensing, see the Cisco Quick Pricing Tool. (A login account is required to access these resources.)

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