CHAPTER 14

Cisco Unified MeetingPlace

Last revised on: November 15, 2010

This chapter covers system-level design and implementation of a Cisco Unified MeetingPlace (Unified MP) system in the Cisco Unified Communications environment. This chapter does not cover any hardware requirements or software component configurations of Unified MP that are not related to system design. For information on these topics, refer to the Unified MP product documentation available at http://www.cisco.com.

What's New in This Chapter

Most of the sections in this chapter were reorganized and rewritten for the initial release of Cisco Unified MeetingPlace 7.0. If you are reading this document for the first time since the 7.0 release of Unified MP, it would be best to read the entire chapter. Table 14-1 lists the topics that are new or that have changed significantly since the 7.0 version of this chapter.

Table 14-1 New or Changed Information Since the Previous Release of This Document

New or Revised Topic	Described in:
Support for multisite Unified MP using WebConnect was removed.	
Support for Reservationless Single Number Access (RSNA) is limited to two Unified MP systems.	Design Considerations for RSNA, page 14-7
Support for SIP integration with Cisco Unified Border Element has been removed until a future release of Unified MP.	
Unified MP redundancy information was updated.	Redundancy, page 14-19
Unified MP capacity information was updated.	Capacity and Sizing, page 14-14

Cisco Unified MeetingPlace Components

This section introduces and briefly discusses various components in a Unified MP deployment.

Unified MP Application Server

The Unified MP Application server is installed on the Cisco Media Convergence Server (MCS) 7835H2 or 7845H2 platform running the Linux operating system and the IBM Informix Dynamic Server (IDS) database. The Unified MP Application server acts as the master component and it ties the rest of the components together. The Unified MP Application server provides SIP B2BUA support and it has SIP connections to other call processing devices such as Cisco Unified CM. The Unified MP Application server can inherently integrate with Cisco Webex and external applications such as email using Simple Mail Transfer Protocol (SMTP) or Microsoft Outlook using HTTP/HTTPS. Integration with LDAP directory services is accomplished via a Unified CM that is integrated with the corporate directory.

Unified MP Web Collaboration Server

The Unified MP Web Collaboration server supports multiple functions, depending on licensing and installation options. Web scheduling, web participant lists, and single sign-on come as part of the base bundle. Unified MP Web Collaboration servers can be clustered to provide increased scalability and redundancy. In addition, Unified MP supports Internal (on the intranet) and External (located in a demilitarized zone (DMZ)) Web server clusters in a single deployment. Unified MeetingPlace uses a concurrent licensing model for voice and video conferencing; however, the number of web conferencing licenses can be different than that for the voice and video conferencing. Licenses are not associated with either the internal or external web server but are used across all web servers when multiple web servers are deployed.

The Unified MP Web Collaboration server communicates with the Unified MP Application server using a GWSIM protocol. Unified MP GWSIM is installed automatically when the Unified MP Application server and Unified MP Web Collaboration software are installed.

Unified MP Media Server

Unified MP Media servers are Cisco Unified Videoconferencing 3515 and 3545 systems, which have onboard DSP resources providing voice and video conferencing. The Unified MP Media server is controlled by the Unified MP Application server via SIP and Unified MP Media Control protocols. When the Unified MP Media server is deployed, end users need only one universal number to dial into voice and video conferencing.

Unified MP Notes Gateway

The Unified MP Notes Gateway provides integration with the IBM Lotus Notes Domino server for Unified MP scheduling, notification, and attendance through the user calendar. This gateway functions similar to Unified MP for Outlook. The main difference is that Unified MP for Notes is server-based while Unified MP for Outlook is client-based.

Unified MP LCS Gateway

The Unified MP Live Communications Server (LCS) Gateway provides services for Microsoft Office Communicator users to escalate a text-based instant messaging session into a voice conference hosted by Unified MP.

Unified MP Conference Manager

Unified MP Conference Manager is a system tool performing help-desk tasks on Unified MP. It can connect multiple Unified MP systems to manage scheduled and in-session meetings. Unified MP Conference Manager is a GUI-based client. Communication between the client and Unified MP Application server is over TCP ports 80 and 443.

Unified MP Deployment Models

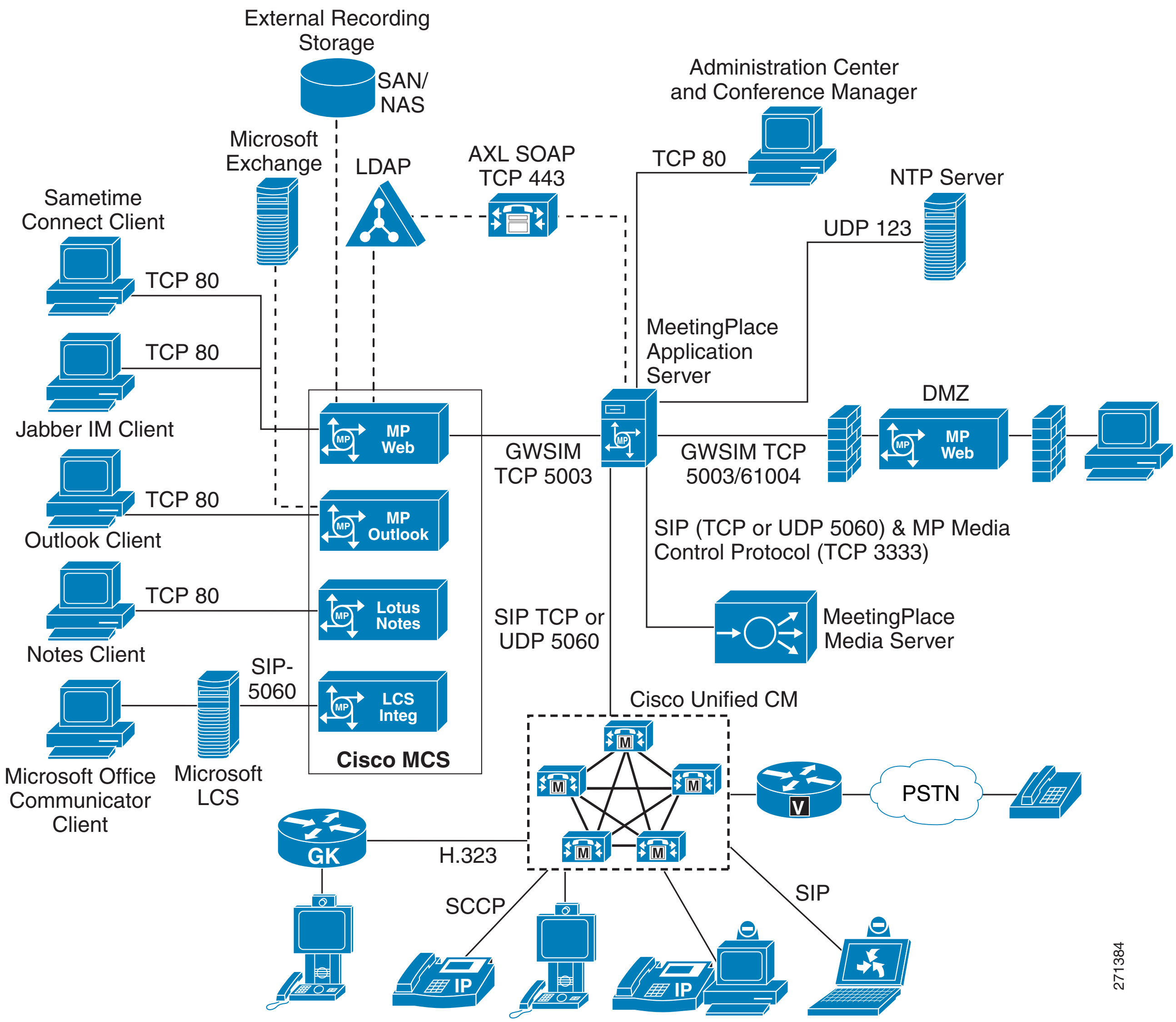
This section discusses design considerations and recommendations for various Unified MP deployment models including single-site, multisite and single-number access deployments. Many variations are possible in the deployments, but the models presented here cover the base implementations without examining every variation.

Single-Site Unified MP Deployment

The single-site deployment model is the base deployment model, with all server components and users located at a single site interconnected by a single LAN (see Figure 14-1). In this model, Unified MP is co-located with the Unified CM cluster and integrated via SIP. Single-site deployments have the following characteristics:

- The Unified MP Media server(s) must be co-located with the active Unified MP Application server.
- The round-trip delay between the active Unified MP Application server and the Unified MP Web Collaboration server(s) must not be greater than 150 ms.
- Single-site Unified MP deployments can include integrations with Microsoft Exchange, Microsoft LCS, IBM Lotus Notes, Directory Services, Jabber Messenger, and Sametime Connect.
- External web conferencing access is made available by configuring Segmented Meeting Access as described in the section on Segmented Meeting Access Option, page 14-7. Within a single site, web conferencing servers can be configured in both internal and external clusters for increased capacity and redundancy.
- Unified MP audio, video, and web recordings and meeting attachments can optionally be stored on an external SAN/NAS storage server.
- Unified MP 7.*x* does not support direct integration with third-party IP PBX endpoints. Cisco Unified CM 6.1(2) or later release must be deployed to front-end Unified MP 7.*x* for third-party IP PBX integration.
- Network Time Protocol (NTP) must be implemented to allow Unified MP components to synchronize their clocks to a network time server or network-capable clock. NTP is a critical network service for Unified MP because it ensures accurate time for scheduling meetings. The external NTP source can be specified during Unified MP Application server installation, and other Unified MP components will synchronize with the application server automatically.
- For redundancy, a single-site Unified MP deployment can be deployed in a single or dual data center environment. This is described further in the section on Unified MP Application Server Redundancy, page 14-19.

Figure 14-1 Single-Site Unified MP Deployment



Note Unified MP 7.*x* Application server has built-in capability to integrate with Microsoft Outlook at the front end. This integration is in the form of an Outlook plug-in that communicates with the Unified MP Application server to schedule meetings. Outlook can also be integrated at the back end to allow users to schedule meetings via the Unified MP Web Collaboration server's web interface, and to have these meetings added to the Outlook calendar. This back-end Outlook integration is optional. In Unified MP 7.0(1), it requires that you install the Unified MP Outlook Application gateway component on an alternate Cisco Media Convergence Server (MCS). In Unified MP 7.0(2) and later releases, the back-end integration with Outlook is incorporated into the Unified MP Application server, and a separate Cisco Media Convergence Server (MCS) is no longer required.

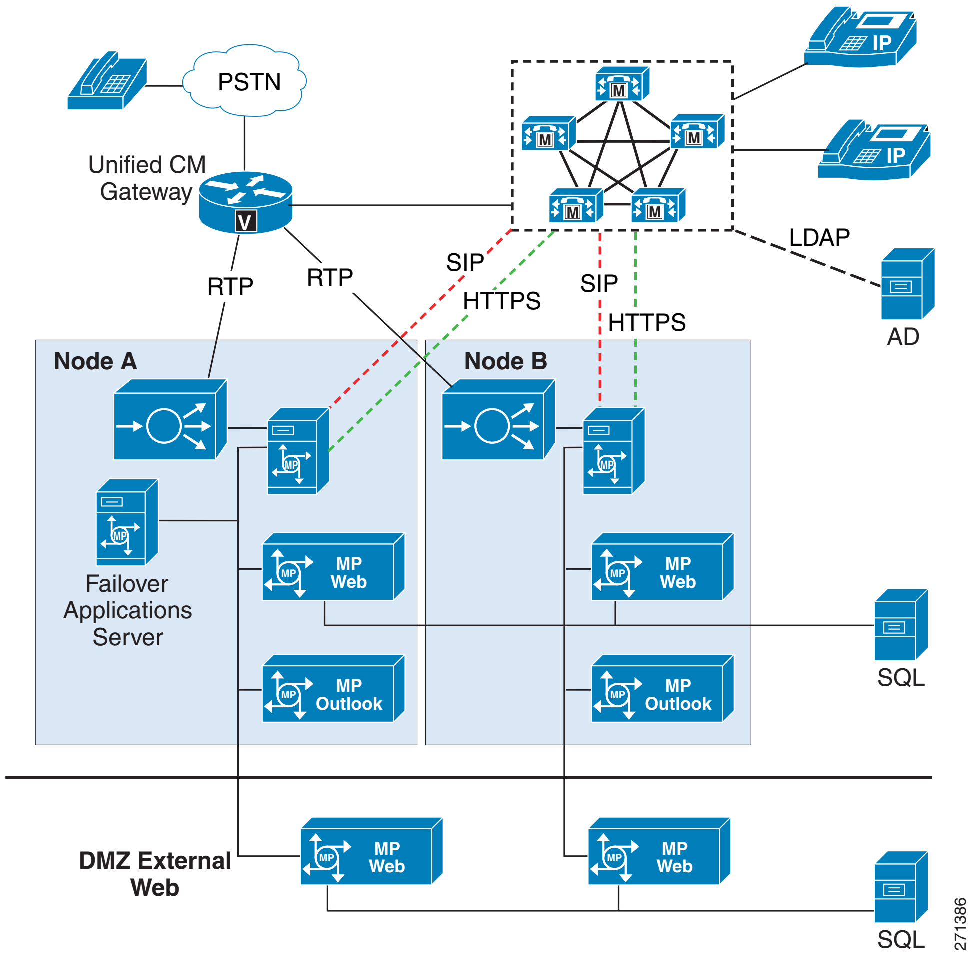
Reservationless Single-Number Access Deployment

Reservationless single-number access (RSNA) is the concept of using one access number to access multiple Unified MP reservationless systems. The RSNA feature enables multiple Unified MP Web Conferencing servers that share the same SQL database to appear as one server to the user community. Regardless of where the end user's profile is located, Unified MP will automatically transfer the end user to the appropriate server after their profile number or meeting ID is entered. With RSNA implemented, the reservation or scheduled meeting option becomes unavailable. RSNA is required under the following conditions:

- Reservationless usage will exceed the capacity of a single Unified MP system.
- The customer requires one access number for all Unified MP systems.
- Failover support is required to handle the situation where a single Unified MP media server goes down.
- Videoconferencing is supported with RSNA deployments in Unified MP 7.*x*.

Figure 14-2 illustrates an RSNA deployment.

Figure 14-2 Reservationless Single-Number Access Deployment



Design Considerations for RSNA

- RSNA requires user profiles to be synchronized on all participating Unified MP systems, which is normally done via the Unified MP Directory Service.
- RSNA relies on the SIP REFER method for inbound calls, which is supported by the Unified MP Application server and Unified CM.
- If H.323 VoIP gateways are used, H.323 and SIP interworking must be implemented to convert H.323 to SIP prior to reaching Unified MP. This can be accomplished by using Unified CM to interwork between the H.323 VoIP gateway and RSNA. Unified CM 6.1(2) or later release is required to perform H.323 and SIP interworking.
- RSNA supports a maximum of two separate Unified MP systems.

Segmented Meeting Access Option

This deployment option may be applied to any of the deployment models discussed previously in this chapter.

Cisco Unified MP supports the placement of a web collaboration server within the demilitarized zone (DMZ) for publicly listed meetings. This is referred to as Segmented Meeting Access (SMA). External participants use this server for web collaboration, while internal participants in internal meetings use an internal web server for web collaboration. When internal participants join an external meeting, they are redirected by the internal web collaboration server to the external web collaboration server. Audio from external participants comes into Unified MP through a VoIP gateway (inter operating with Cisco Unified CM) to the Unified MP Media server. Table 14-2 lists the ports that must be opened on the internal corporate firewall to allow communication between the DMZ web server and the various Unified MP components on the internal network.

Table 14-2 Ports Used by Cisco Unified MeetingPlace

Protocol	Port Type	Ports	Port Usage
HTTP or HTTPS	TCP	80 (1627), 443	Web
RTMP	TCP	1627	Web
GWSIM	TCP	5003, 61004	Cisco Unified MP application server
SQL	TCP	1433	Database

Note In order to allow external users to access meeting recordings that are stored on an internal SAN/NAS server, it is necessary to open the ports that are used by the SAN/NAS server on the internal corporate firewall. Consult your SAN/NAS device product documentation for specific port requirements.

SMA implementations require deployment of the internal web server(s) in addition to the DMZ web server. For more information on SMA, refer to the Cisco Unified MeetingPlace product documentation available at http://www.cisco.com.

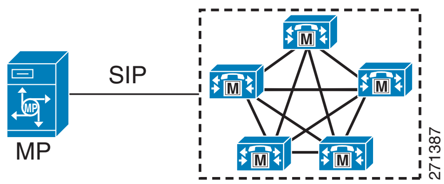
Integrating Unified MP with SIP and H.323 Call Processing Agents

This section discusses design guidance and recommendations for integrating Unified MP with Unified CM via SIP. Integrating with H.323-compliant call processing agent is accomplished by using Unified CM to provide SIP-to-H.323 protocol interworking capability.

SIP

Unified MP 7.*x* integrates directly with Cisco Unified CM via a SIP trunk. Figure 14-3 illustrates this integration method.

Figure 14-3 SIP Integration with Unified CM



A SIP trunk must be configured in Unified CM with a destination address of the Unified MP Application server, and then a route pattern must be used to route calls via the SIP trunk to Unified MP. A SIP proxy server must also be configured in Unified MP call control with an IP address or hostname of a Unified CM call processing subscriber. There are additional guidelines to allow for redundancy that are described in the section on Redundancy, page 14-19. For more details on Unified CM and Unified MP integration configuration, refer to the *Installation and Upgrade Guide for Cisco Unified MeetingPlace*, available at

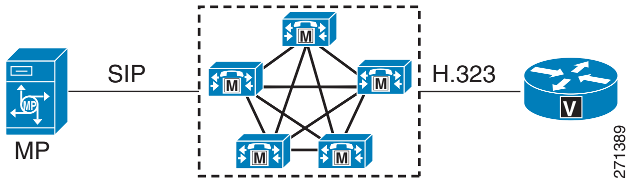
http://www.cisco.com/en/US/products/sw/ps5664/ps5669/prod_installation_guides_list.html

Unified MP supports receiving both Early Offer (EO) and Delayed Offer (DO) SIP Invite messages. Unified MP 7.0(2) and later releases initiate EO SIP Invites for outbound calls, but Unified MP 7.0(1) initiates DO SIP Invites for outbound calls. By default, Unified CM sends calls to Unified MP using DO SIP invites. Unified CM can be configured to use EO; however, this requires the use of a media termination point (MTP) resource. For more information, see the section on SIP Delayed Offer and Early Offer, page 5-19.

H.323

Unified MP 7.*x* does not support H.323 natively. Therefore, in order to integrate with H.323 devices such as an H.323 VoIP gateway or gatekeeper, Cisco Unified CM 6.1(2) or later release must be deployed in front of the Unified MP Application server to provide protocol translation capability between H.323 and SIP. (See Figure 14-4.)

Figure 14-4 H.323 Integration via Unified CM



H.323 Video Endpoint

For deployments incorporating H.323 video endpoints, Cisco recommends deploying a gatekeeper with which H.323 video endpoints can register. (See Figure 14-1 for a deployment with an H.323 video endpoint and gatekeeper.)

In Unified MP 7.*x*, the RAS-aggregator trunk is no longer needed when deploying H.323 video endpoints and gatekeeper because Unified CM acts as a SIP front-end device and performs SIP and H.323 protocol interworking for the gatekeeper. Thus, Unified CM performs digit manipulation and bandwidth control when necessary for calls destined for video endpoints registered with the gatekeeper. The following example illustrates the gatekeeper configuration:

```
gatekeeper
 zone local ccmtrunk cisco.com 1.1.1.1! Unified CM registers with gatekeeper in ccmtrunk
zone
 zone local video-ep cisco.com! Video endpoint registers with gatekeeper in video-ep zone
 no zone subnet ccmtrunk default enable
 zone subnet ccmtrunk 2.2.2.2/32 enable
 no zone subnet video-ep default enable
 zone subnet video-ep 3.3.3.3/32 enable
 zone prefix ccmtrunk 1... gw-priority 10 ccm_trunk_1  ! Calls to Unified MP is sent to
Unified CM
 gw-type-prefix 1#* default-technology
 no use-proxy video-endpoint default inbound-to terminal
 no use-proxy video-endpoint default outbound-from terminal
 no shutdown
```

Call Admission Control, QoS, and Bandwidth

Call admission control, Quality of Service (QoS), and proper bandwidth allocation are the main mechanisms to ensure voice and video quality. This section describes how these mechanisms apply to Unified MP.

Call Admission Control

Call admission control with Unified MP should be performed by Unified CM as the call processing agent. Unified CM can perform call admission control based on the available bandwidth in the location with which the SIP trunk to the Unified MP Application server is associated. Alternatively, Unified CM supports the use of the Resource Reservation Protocol (RSVP) which can also provide call admission control. For further information regarding call admission control strategies, refer to the chapter on Call Admission Control, page 9-1.

QoS Markings

Call Signaling

SIP signaling traffic from the Unified MP Application server is marked CS3 (DSCP 0x18), and GWSIM traffic is not marked.

Media Stream

The audio stream from the Unified MP Media server is marked EF (DSCP 0x2E), and the video stream is marked AF41 (DSCP 0x22) by default.

Web Collaboration Traffic

Web collaboration traffic from the Unified MP Web Collaboration server is marked best-effort (DSCP 0x00).

Bandwidth

Call Control Bandwidth

Call control bandwidth is extremely small but critical. Co-locating the Unified MP Application server with Unified CM or Cisco Unified Border Element helps protect against issues with call control. Remote locations need proper QoS provisioning to ensure reliable operation.

GWSIM

GWSIM traffic flows between the Unified MP Application server and the Unified MP Web Collaboration servers. GWSIM traffic is minimal but important traffic, therefore Cisco recommends placing all Unified MP Web Collaboration servers in the same location as the Unified MP Application server to help ensure proper operation. Separation of components by a WAN would require proper QoS provisioning to ensure reliable operation.

Traffic between the Unified MP Application server and Unified MP Web Collaboration server includes some database synchronization and might therefore at times be bursty in nature, but this traffic is not real-time.

Real-Time Transport Protocol (RTP) Traffic Bandwidth

RTP traffic consists of voice and video traffic. The Unified MP Media server supports G.711, G.729, G.722, and iLBC as audio codecs, and it supports H.261, H.263, and H.264 as video codecs.

Capacity on Unified MP is affected by the codec chosen, as discussed in Capacity and Sizing, page 14-14. For further information regarding bandwidth calculations per codec type, refer to the chapters on Network Infrastructure, page 3-1, and IP Video Telephony, page 16-1.

Web Collaboration Bandwidth

Web collaboration is the largest bandwidth consumer, and this traffic becomes especially significant for remote users across WAN links. Users at remote sites that cause web collaboration to traverse a WAN will require special consideration. The client flash session bandwidth or room bandwidth setting for these users should be lowered, reducing the load across the WAN. Because web collaboration data is delivered unicast, the largest burst of data should be multiplied by the number of clients at a remote site. For example, assume a remote site has 100 users, 10 of which are on a web collaboration session at any one time. If bursts of 1.5 Mbps occur in the data from the remote server to each user, 15 Mbps bursts can be experienced across the WAN connection.

When WAN links become congested from excessive web collaboration data or other sources, the degradation of all traffic is compounded by packet loss, retransmissions, and increased latency. Sustained congestion will have a sustained degrading impact on all remote collaboration sessions. The following settings on the client web collaboration sessions control the rate at which the participant receives data as well as the rate at which the presenter sends data:

- Modem — Bandwidth limited to 28 kbps
- DSL — Bandwidth limited to 250 kbps
- LAN — Bandwidth up to 1,500 kbps

Bandwidth bursts above 1,500 kbps are possible if high-resolution images or photos are shared. Sharing normal to complex presentations or documents should not generate bursts above 1,500 kbps unless large complex images are embedded. Bandwidth settings are not automatically adjusted when congestion occurs; they must be adjusted manually. Bandwidth settings default to LAN and must be set at the initialization of each web collaboration session. A new session is set to LAN regardless of previous settings.

Cisco recommends using a LAN or DSL connection for web conferencing in order to have a better user experience. With a LAN connection, each participant should have 1,500 kbps of downstream bandwidth and the presenter should also have 1,500 kbps of upstream bandwidth. With a DSL connection, each participant should have 600 kbps of downstream bandwidth and presenter should also have 600 kbps of upstream bandwidth. This web conferencing bandwidth requirement is based on the default meeting room resolution setting (800 x 600 pixels). If the meeting room resolution is adjusted below or above the default setting, the web conferencing bandwidth requirement will decrease or increase proportionally. If a modem connection is used, Cisco recommends not running other applications simultaneously with Unified MP web conferencing, which will compete for available bandwidth.

DTMF Support

Unified MP supports the following standard dual-tone multi-frequency (DTMF) transmission methods:

- RFC2833 and KPML DTMF transmissions when using SIP
- Inband acoustic DTMF transmission

For further information regarding DTMF transmission, refer to the chapter on Media Resources, page 6-1.

External Directory Integration via Unified CM

Integrating an external directory with Unified MP through Unified CM 5.0 or later provides two main functions:

- Automatic profile creation in Unified MP
- External authentication using a third-party directory

When Unified MP is integrated with Unified CM and an external LDAP directory, user profiles are created automatically the first time users log into Unified MP. The profiles enable users to schedule meetings immediately and use the system. Integrating an external LDAP directory requires an LDAP-based user authentication method for end users who try to log in via the web.

Unified MP users must be authenticated against the external corporate directory with which Unified CM is integrated. Unified MP directory integration service provides Unified MP profiled users with single sign-on capability, which allows users who have already been authenticated once to have access to all resources and applications on the network without re-entering their user credentials. For details about various Unified MP user authentication methods, refer to the Unified MP product documentation available at http://www.cisco.com.

Unified MP supports only the same external LDAP systems and versions that Unified CM supports. Unified MP integrates with Unified CM via Cisco AVVID XML Layer (AXL) Simple Object Access Protocol (SOAP) over secure HTTP (HTTPS). Unified MP cannot synchronize user profiles directly from the external LDAP system.

Unified CM supports the following directory servers:

- Microsoft Active Directory 2000, 2003, and 2007
- Netscape and SunOne LDAP Directory Server Version 4 and Version 5
- Cisco Unified CM directory

  Synchronizations of user data from the Unified CM directory enable the Unified MP system to support Cisco Unified Communications users who are configured in Unified CM.

For further information regarding Unified CM directory integration, refer to the chapter on LDAP Directory Integration, page 17-1.

Unified MP and Cisco WebEx Integration

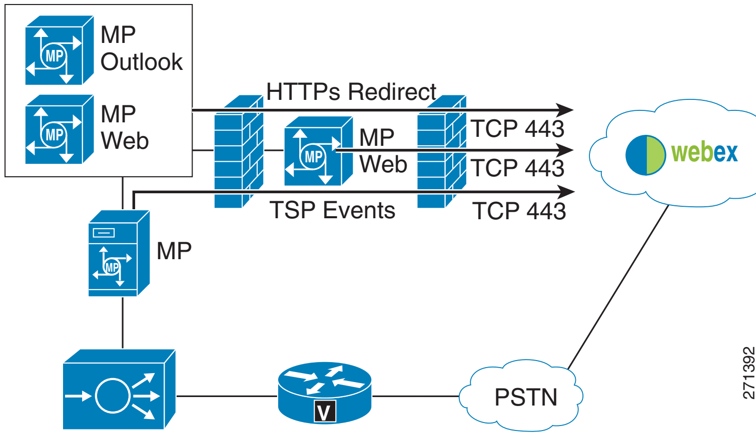
Integration between Cisco Unified MP and WebEx provides both on-premises and hosted on-demand collaboration solutions. Unified MP licensing is not required for implementing Unified MP and WebEx integration, and a Unified MP Web User License (UL) is not required for using WebEx as the web conferencing provider. You can use either of the following two options to integrate Unified MP and WebEx.

Option 1: Unified MP and Webex Integration with Unified MP Scheduling

This option allows Unified MP profiled users to schedule meetings via the Unified MP web user interface or via the Outlook Calendar plug-in. With this integration option, audio conferencing is provided by Unified MP and web conferencing is provided by WebEx. WebEx Meeting Center is the WebEx meeting template provided. A WebEx meeting is created automatically when the first meeting participant joins the web conference, therefore WebEx will not receive any notifications when a meeting is scheduled on Unified MP. There is no video conferencing available with this integration option; instead, WebEx provides basic webcam video streaming.

Figure 14-5 illustrates this option. For further information regarding WebEx meeting templates, refer to the Cisco WebEx product documentation available at http://www.cisco.com.

Figure 14-5 Cisco Unified MP and WebEx Integration, Option 1



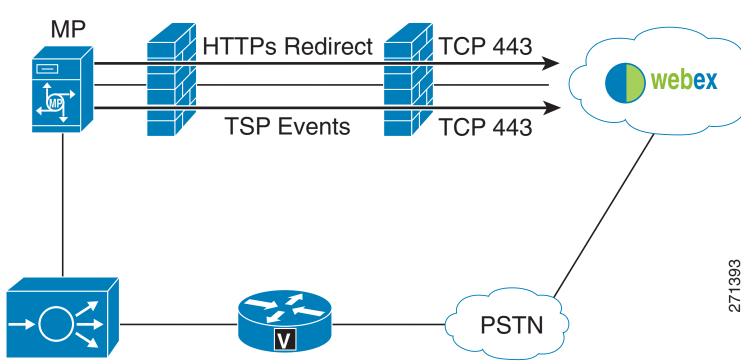
Meeting participants can create a synchronized web and audio recording. Audio recording invokes an out-dial event from the WebEx media tone network to the Unified MP Media server via the PSTN. The recorded meetings are not available from the Unified MP Web user interface, and users must log in to their WebEx accounts to access the recorded meetings.

When a user joins a WebEx conference, Unified MP first authenticates the user then sends the user's request to the Unified MP Web Collaboration server (either internal or external), which redirects it to the WebEx Media Tone Network via secure HTTP. This redirect behavior is completely transparent to the user, and user authentication is performed solely by the on-premises Unified MP system. All meeting-related service requests are exchanged and processed via telephony service provider (TSP) application programming interface (API) calls between the Unified MP Application server and WebEx.

Option 2: Unified MP and Webex Integration with WebEx Scheduling

This option allows users to schedule meetings via the WebEx web user interface or WebEx Outlook Calendar plug-in. With this option, audio conferencing is provided by Unified MP and web conferencing is provided by WebEx. Unified MP provides Reservationless voice conferencing, and participants are put in the waiting room until the meeting host joins. There is no video conferencing available with this integration option; instead, WebEx provides basic webcam video streaming. Figure 14-6 illustrates this option.

Figure 14-6 Cisco Unified MP and WebEx Integration, Option 2



Meeting participants can start audio-only recording via a voice user interface such as telephone, or they can start audio and web recording from WebEx meeting room. Audio recording invokes an out-dial event from the WebEx media tone network to the Unified MP Media server via the PSTN. The recorded meetings are not available from the Unified MP Web user interface, and users must log in to their WebEx accounts to access the recorded meetings.

When a Unified MP profiled user tries to schedule a WebEx meeting or access My WebEx link from the Unified MP web user interface, WebEx automatically creates the user account based on the Unified MP user profile. Several Unified MP user profile attributes are inherited by WebEx, including username, password, first name, last name, and email address. Because a WebEx Site is dedicated to a specific customer and the WebEx user profile is based on the Unified MP user profile, there should not be any user profile conflicts.

Note Use of the WebEx service incurs fees based on the contract signed between the customer and Cisco WebEx. The WebEx service fee is independent of Cisco Unified MP implementation and licensing.

Capacity and Sizing

Unified MP 7.0(2) and later releases support a concurrent maximum of 1500 audio sessions (**G.711, G.729** Global Audio Mode setting), 300 video sessions (**Standard Rate** Global Video Mode setting), and 1000 web sessions.

Unified MP Media Server

The Global Audio Mode configuration in Unified MP Application Administration determines system voice capacity. Global Audio Mode can be configured in either of the following ways:

- G.711, G.729 — With this configuration, a single audio blade in the Unified MP Media server can support a maximum of 250 voice ports. It would require 6 audio blades to reach the maximum supported system limit of 1500 concurrent audio sessions.
- G.711, G.722, G.729, iLBC — With this configuration, a single audio blade can support a maximum of 166 voice ports. With 6 audio blades, the maximum supported number of concurrent audio sessions using these additional codecs is 996.

The Global Video Mode configuration in Unified MP Application Administration determines the system video capacity. Global Video Mode can be configured in the following two ways:

- Standard Rate (video call speed up to 384 kbps) — In this mode, an Enhanced Media Processor (EMP) in the Unified MP Media server can support a maximum of 48 video ports.
- High Rate (video call speed up to 2048 kbps) — In this mode, the EMP can support a maximum of 24 video ports.

For a complete list of the video formats supported by Unified MP, refer to the *Video Format Support* section of the *Cisco Unified MeetingPlace, Release 7.0 -- Video Endpoint Compatibilit*y page, available at http://docwiki.cisco.com.

Unified MP Media servers can be either the Cisco Unified Videoconferencing 3515 Multipoint Control Unit (MCU) or the Cisco Unified Videoconferencing 3545 System. The Unified Videoconferencing 3515 media server is a fixed platform that comes with an audio blade and an EMP pre-installed. The Unified Videoconferencing 3545 media server is a modular platform consisting of a chassis that supports multiple audio blades or EMPs in various combinations.

Virtual Cascading

If multiple audio blades and EMPs are installed in the Unified Videoconferencing 3545 media server, the media server uses virtual cascading to overflow voice and video streams from one audio blade or EMP to another. The audio blade has built-in cascading ports that do not decrease the audio session capacity. With a single EMP deployed in the Unified MP system, all video ports are available for video conferencing. With multiple EMPs deployed, the media server will automatically reserve video ports for cascading purposes. For Standard Rate video, 8 video ports are reserved for cascading, leaving 40 video ports available. For High Rate video, 4 video ports are reserved for cascading, leaving 20 video ports available. The following two examples illustrate audio and video port usage when cascading.

Example 14-1 Audio Conference

A Unified Videoconferencing 3545 media server is deployed with two audio blades and two EMPs. A meeting is scheduled with 350 audio ports and the Global Audio Mode is configured for G.711, G.729.

- The media server allocates 251 ports from the first audio blade, out of which 250 ports are used for audio participants and one port is used for voice cascading or connecting to the second audio blade.
- The media server allocates 101 ports from the second audio blade, out of which 100 ports are used for audio participants and one port is used for voice cascading.

Example 14-2 Video Conference

A Unified Videoconferencing 3545 System media server is deployed with two audio blades and two EMPs. For this example, assume a meeting is scheduled with 65 video ports and the Global Video Mode is configured for Standard Rate video.

- The media server allocates 41 ports from the first EMP, out of which 40 ports are used for video participants and one port is used for video cascading or connecting to the second EMP.
- The media server allocates 26 ports from the second EMP, out of which 25 ports are used for video participants and one port is used for video cascading.

Guidelines for Sizing Unified MP Audio Conferencing

Cisco recommends the following three methods for calculating Unified MP audio conferencing capacity.

Calculation Based on Number of Knowledgeable Workers

Cisco recommends provisioning an audio user license (UL) for every 20 knowledge workers. A knowledgeable worker is anyone who uses Cisco Unified MP frequently.

For example, in a system with 40 knowledgeable workers, you should provision 2 audio ULs.

Calculation Based on Average Monthly Usage

If you know the average voice conferencing usage (average minutes per month), use Table 14-3 to calculate the Unified MP audio conferencing capacity.

Table 14-3 Unified MP Audio Conferencing Capacity Based on Average Monthly Usage

Average Monthly Usage (minutes)	Baseline Usage (minutes per user license per month)
20,000 to 50,000	1,500
50,000 to 500,000	2,000

Table 14-3 Unified MP Audio Conferencing Capacity Based on Average Monthly Usage

Average Monthly Usage (minutes)	Baseline Usage (minutes per user license per month)
500,000 to 1,000,000	3,000
1,000,000 to 2,000,000	3,500
2,000,000 to 8,000,000	4,000

Calculation Based on Actual Peak Usage

Actual voice conferencing usage during peak hours usually can be obtained from existing voice conferencing system logs or service provider bills. Cisco recommends provisioning 20% to 30% extra capacity based on the actual peak usage in order to protect against extra conferencing volume.

Note A user license (audio, web, or video) is not granted to any particular user but, rather, is a system-wide resource shared by all users in the Unified MP system.

Factors Affecting System Sizing

The following factors affect system sizing in addition to the estimates provided by the methods described above for the system baseline port requirement.

- When migrating from an "operator-scheduled" model to a user-scheduled or reservationless model on Cisco Unified MeetingPlace, you might need to add another 20% to the baseline.
- The average sized meeting has a default of 4.5 callers per meeting. Use the value that is applicable to your case if it is different than the default.
- Increase the baseline estimate accordingly if the following condition applies:

  (Estimated meetings per day) * (Estimated users) > 80% of baseline
- If the largest single meeting exceeds 20% of the estimated capacity, increase the estimate accordingly.
- If there are continuous meetings with dedicated ports, then you must add those additional ports ((Meetings) * (Dedicated callers)) to the baseline.

The total number of ports will include all the above factors in addition to the baseline.

When planning for Unified MeetingPlace capacity expansion, also consider whether the following conditions apply to your system:

- The total estimated port capacity exceeds 80% of the maximum supported ports as listed in Table 14-4.
- An audio codec other than G.711 is desired. However, transcoders based on Cisco Integrated Services Routers (ISR) can be used if needed to achieve maximum capacity for other codec types in the meetings.
- Line Echo Cancellation (LEC) is provided by an external device such an ISR, rather than unified MeetingPlace providing echo cancellation functionality.

Guidelines for Sizing Unified MP Video Conferencing

Cisco recommends the following three methods for calculating Unified MP video conferencing capacity.

Calculation Based on Number of Knowledgeable Workers

Cisco recommends provisioning an audio UL for every 40 knowledgeable workers.

Calculation Based on Number of Voice Conferencing ULs

Cisco recommends provisioning video conferencing capacity in the range of 17% to 25% of existing voice conferencing ULs. The percentage depends on business requirements regarding video conferencing and on the size of the Unified MP system.

Calculation Based on Existing Video MCU

Cisco recommends deploying a direct replacement for an existing video conferencing system. A video conferencing license on the existing system can be replaced by a Unified MP UL.

Unified MP Web Collaboration Server

When installed on a Cisco MCS-7845H2/I2 server, a Unified MP Web Collaboration server can support a maximum of 500 web sessions or 500 web user licenses (ULs). Unified MP Web servers can be clustered to increase capacity to a maximum of 1000 web sessions per system with two MCS-7845H2/I2 Unified MP Web Collaboration servers. This configuration can support a single web conference with 1000 participants or 20 web conferences with 50 participants each.

Note Highly redundant systems can have a cluster of up to three External web servers in addition to a cluster of up to three Internal web servers. This enables load balancing and redundancy. However, a single Unified MP system supports a maximum of only 1000 web sessions across all Unified MP Web servers.

Alternatively, a single Cisco MCS-7835H2/I2 server can support a maximum of 250 web sessions or 250 web ULs. Cisco MCS-7835H2/I2 servers can also be clustered (up to three maximum) for increased capacity and redundancy.

Note Cisco does not recommend mixing Unified MP Web server models. Unified MP Web Servers load-balance by default and split the load across web servers, but they do not have the capability to split the load according to the MCS model type. Therefore, Cisco recommends deploying Unified MP Web servers on identical MCS models.

There is no change in web conferencing capacity when Secure Socket Layer (SSL) Transport Layer Security (TLS) is implemented for Unified MP web conferencing. Therefore, Cisco recommends using SSL/TLS to enable secure web conferencing.

Guidelines for Sizing Unified MP Web Conferencing

Cisco recommends the following four methods for calculating Unified MP Web conferencing capacity.

Calculation Based on Number of Knowledgeable Workers

Cisco recommends provisioning an audio UL for every 40 knowledgeable workers.

Calculation Based on Number of Voice Conferencing ULs

Cisco recommends provisioning web conferencing capacity in the range of 33% to 50% of existing voice conferencing ULs. The percentage depends business requirements regarding web conferencing and on the size of the Unified MP system.

Calculation Based on Actual Peak Usage

Actual web conferencing usage during peak hours usually can be obtained from existing web conferencing system logs or service provider bills. Cisco recommends provisioning 20% extra capacity based on the actual peak usage in order to protect against extra conferencing volume.

Calculation Based on Existing Web Conferencing System

Cisco recommends deploying a direct replacement for an existing web conferencing system. A web conferencing license on the existing system can be replaced by a Unified MP UL.

Note Cisco recommends provisioning 20% of Unified MP ULs as floater ports and provisioning 30% of total ULs as overbooked ports.

System Capacity Limits for Deployments

Table 14-4 lists the system capacity limits by Unified MeetingPlace version and Cisco Media Convergence Server (MCS) model.

Table 14-4 Maximum Port Capacities

Cisco Unified MeetingPlace Release and Server Model	Maximum Number of High Capacity Mode Ports	Maximum Number of Multiple Codec Ports	Estimated Maximum Number of Users	Estimated Maximum Meetings per Day	Estimated Maximum Audio Minutes per Month	Maximum Number of Web Conferencing Sessions
Release 7.*x*, (MCS 7845)	1,500 (G.711 and G.729 with no LEC)	N/A	30,000	1,500	6 Million	1,000 on 3 MeetingPlace Web (non-SSL[1]) or 1,000 WebEx SaaS[2]
Release 7.*x*, (MCS 7845)	N/A	996	20,000	996	3.9 Million	1,000 on 3 MeetingPlace Web (non-SSL[1]) or 1,000 WebEx SaaS[2]

1. SSL = Secure Sockets Layer
2. SaaS = Software-as-a-Service

These are guidelines for determining the number of audio ports for a new system. These guidelines were developed using industry standard conferencing usage assumptions. Actual usage may vary based on several factors including multiple time zones, business hours (24/7 vs. 8/5), and community usage behavior. Guidelines were based on formulas provided in the Guidelines for Sizing Unified MP Audio Conferencing, page 14-15, as typical usage factors.

If an existing service provider can supply current conferencing peak hour port usage or monthly minutes for several months, those also should be used in estimating for an on-premises Unified MeetingPlace system, but experience indicates that growth should be expected. If you have actual Unified MeetingPlace usage data, please gather the actual conferencing port usage from your existing system to determine the number of ports required. An actual system might be able to support more users or fewer users, based on actual conference usage. In addition, your actual system might support more meetings or fewer meetings.

After a Unified MeetingPlace system is installed, monitoring the Monthly Port Utilization Reports will provide actual usage for measuring growth and actual usage patterns.

Redundancy

This section describes redundancy considerations for the following Unified MP components:

- Unified MP Application Server, page 14-19
- Unified MP Media Server, page 14-21
- Unified MP Web Collaboration Server, page 14-22
- Call Control, page 14-22

Unified MP Application Server

Unified MP 7.*x* allows for a primary and a single standby Unified MP Application server. Each Unified MP Application server in a failover deployment is configured with the same IP address associated to its physical network interface controller (NIC) and a unique IP address associated to a virtual network interface. The requirement for both Unified MP Application servers to share the same IP address mandates that both Application servers are connected to the same virtual LAN (VLAN) or IP subnet. This is not an issue when both servers are placed in a single data center; however, a dual data center design is supported only if the same VLAN (IP subnet) spans both data centers. All Unified MP components as well as Unified CM communicate with this shared IP address. The physical NIC (with the shared IP address) of the standby server remains disabled until the primary server fails and the manual failover process is initiated. The standby server has the following network connectivity requirements:

- Connected to the same VLAN (IP subnet) as the primary
- Less than 250 ms round-trip delay between standby and primary
- Less than 1% packet loss between standby and primary
- Minimum bandwidth of 384 kbps between standby and primary

The virtual network interface is used for Informix database replication between the primary and standby servers. The database replication ensures that database tables related to users, groups and meetings are synchronized between primary and standby servers. TCP Port 2008 is used between Application Servers for database replication and must be open if firewalls are deployed. Cisco recommends placing the virtual network interfaces of the primary and standby servers in the same VLAN. A utility accessed via the server's command line interface (CLI) named **failoverUtil** is used to set up the primary and standby Unified MP Application servers and to establish database replication between the two. For further information regarding the **failoverUtil** utility, refer to the *Configuration Guide for Cisco Unified MeetingPlace*, available at

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

Note If the Unified MP Application server fails, calls in progress will be dropped. Users would need to redial into their conferences once the standby Application server has been promoted to primary. There is no automatic failover mechanism available for the Unified MP Application server.

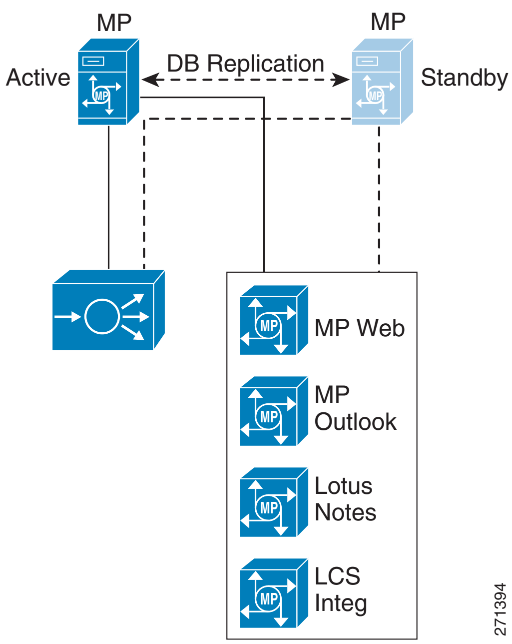
Another key requirement for a Unified MP 7.*x* solution is that the active Unified MP Application server must be co-located with the active Unified MP Media server(s). Therefore, single data center and dual data center designs have slightly different considerations.

Single Data Center Design

In a single data center design, failover of the Unified MP Application server occurs within the same geographic location. For this type of deployment, there would typically be one set of Unified MP Media servers shared by the primary and standby Unified MP Application servers. If the primary Unified MP Application server fails, the Unified MP Media server(s) must be synchronized with the standby (now primary) server. Unified MP Web Collaboration server(s) would also be shared. Figure 14-7 illustrates the failover process for the Unified MP Application server in a single data center deployment.

Note For highly redundant solutions, it is also possible to have a set of standby Unified MP Media servers and Unified MP Web Collaboration servers in a single data center.

Figure 14-7 Failover of Unified MP Application Server in a Single Data Center Deployment

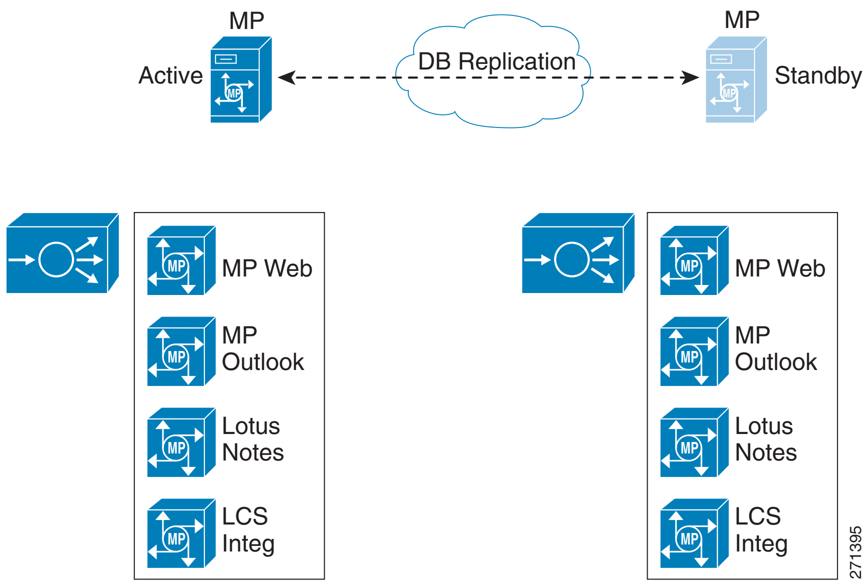


Dual Data Center Design

In a dual data center design, failover of the Unified MP Application server occurs between different geographic locations across an IP WAN. Again, although both servers are separated geographically, both the primary and standby Application servers must be connected to the same VLAN to ensure proper failover operation. For this type of deployment, the standby Application server must be co-located with a redundant Unified MP Media server(s) with which it is synchronized. If the identical number of Unified MP Media server audio and video blades are not maintained in the standby data center, system capacity will be reduced during failover scenarios where the standby Application server is promoted to primary.

As in normal design, the round-trip delay between the active Unified MP Application server and the Unified MP Web Collaboration server(s) must not be greater than 150 ms, so redundant Web servers are required only if the 150 ms round-trip time is exceeded with the standby data center. Figure 14-8 illustrates the failover process for the Unified MP Application server in a dual data center deployment.

Figure 14-8 Failover of Unified MP Application Server in a Dual Data Center Deployment



Unified MP Media Server

The Unified MP Application Server automatically performs failover to alternate Unified MP Media Servers (audio or video blades) in the system. For example, if the Application Server detects a loss of connectivity with an audio blade, it removes it from the list of active audio blades so that subsequent audio sessions will connect to an active audio blade. To avoid reduction in Unified MP Media Server capacity during an audio or video blade outage, one option is to add additional Media Servers on the Application Server. The Application Server will not exceed the number of sessions for which it is licensed. Another option is to revert to the standby Unified MP Application Server with its own set of Unified MP Media Servers (as in a dual data center design). These two options are not mutually exclusive; a standby Unified MP Application Server with its own set of Unified MP Media Servers can gain further redundancy by adding more Media Servers.

Unified MP Web Collaboration Server

Multiple Unified MP Web Collaboration servers can be clustered to provide increased capacity and redundancy of web servers. A Unified MP Web Collaboration cluster can contain up to three web conferencing servers. The web servers connect to one Unified MP Application server and share the same SQL database which can reside externally or on one of the web conferencing servers. Cisco recommends using an external SQL database for the web server cluster to maximize redundancy. If one web conferencing server becomes unavailable, active meetings will fail-over to the other server(s) within the cluster. If the web server that is mapped to the DNS name used in the web meeting URL is offline, there must be a Unified MP web server to handle these requests. If a different web server in the cluster is offline, Unified MP Web will discontinue directing subsequent web meetings to that web server and operation will continue normally.

Unified MP 7.*x* supports deploying an internal Unified MP Web Collaboration cluster and an external Unified MP Web Collaboration cluster simultaneously (with up to three web servers per cluster); however, they must use two different SQL databases. An internal cluster means that all web conferencing servers are implemented behind the corporate firewall and provide full access (scheduling and attending meetings) to end users. An external cluster means that all web conferencing servers are implemented inside a DMZ and provide attend-only access to end users.

The Unified MP Web component inherently provides load balancing for the web server cluster. When users web into a meeting, the Unified MP Web server checks the load on all web conferencing servers within the cluster and assigns the user to the server with the least load. Load balancing on the Unified MP Web servers is accomplished by sharing web sessions across all Unified MP Web servers in the cluster; therefore, in a cluster with three web servers, meetings will occur simultaneously on all three web servers during normal operation. For more information on Unified MP Web Collaboration server capacity, see the section on Unified Web Collaboration Server Capacity and Sizing, page 14-14.

Call Control

Unified MP allows you to define multiple SIP outdial connections that point to Cisco Unified CM call processing subscribers. For redundancy, multiple SIP proxy servers should be configured to direct calls to alternate call processing subscribers in the Unified CM cluster. Note that the Unified MP Application server will send outbound calls to "SIP proxy server 1" only and will not send calls to "SIP proxy server 2" unless communication is lost with "SIP proxy server 1". Only then will Unified MP automatically send a SIP INVITE message to the next available call processing agent in the list. Failure of the call processing agent should not affect existing calls. The existing media connection is torn down after the user disconnects.

Note The term **SIP Proxy Server** is simply the terminology seen on the Unified MP Application Server configuration pages, and it does not imply that integration with any SIP Proxy server is supported. Unified MP supports only SIP integration to a Cisco Unified CM cluster.

For inbound calls, a single configured SIP trunk in Unified CM can be accessed by all call-processing subscribers in the cluster. If a failover Unified MP Application server is deployed, a second SIP trunk must be configured in Unified CM. Care should be taken so that this second SIP trunk will be used only for calls during failover scenarios when the failover application server is actually activated. For more details on Unified CM and Unified MP integration configuration, refer to the *Installation and Upgrade Guide for Cisco Unified MeetingPlace*, available at

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

For information regarding gatekeeper redundancy, refer to the chapter on Call Processing, page 8-1.