Main page: Cisco Unified MeetingPlace, Release 8.0

Up one level: Planning Your Deployment

This chapter discusses design considerations and recommendations for Cisco Unified MeetingPlace Release 8.0 deployments that include video. These deployments include audio and may or may not include a web component. If you do not want to use video in your deployment, see Choosing an Audio Without Video Deployment for Cisco Unified MeetingPlace Release 8.0.

This chapter does not include a comprehensive list of all possible audio and video deployments but instead describes common scenarios.

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- Determining which Audio and Video Deployment to Use
- Audio and Video Deployment Option 1: Using SCCP or SIP Video Endpoints
- Audio and Video Deployment Option 2: Using H.323 Video Endpoints
- Audio and Video Deployment Option 3: Using SIP, SCCP, and H.323 Video Endpoints
- Choosing Between the Hardware Media Server and the Express Media Server

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About Audio and Video Deployments

All Cisco Unified MeetingPlace Release 8.0 systems require Cisco Unified Communications Manager and all Cisco Unified MeetingPlace Release 8.0 systems automatically ship with Cisco Unified Communications Manager Release 8.0. For audio and video deployments, the minimum release of Cisco Unified Communications Manager that we support is Release 6.1.2.

Cisco Unified MeetingPlace Release 8.0 supports multiple deployments for combined video and audio support. The deployment that you use depends on your endpoints and on whether you use an Express Media Server or a Hardware Media Server. For information about choosing between an Express Media Server or a Hardware Media Server, see the Choosing Between the Hardware Media Server and the Express Media Server.

These deployments depend on native KPML support, although SCCP and SIP endpoints that support RFC2833 DTMF will work. The preferred signaling protocol is SIP.

Related Topics

- For information about interoperability with Cisco Unified Communications Manager, see http://www.cisco.com/en/US/solutions/ns340/ns414/ns728/networking_solutions_products_genericcontent0900ae
- For information about using Reservationless Single Number Access (RSNA) with your video deployment, see Reservationless Single Number Access (RSNA) Deployment in Choosing an Audio Without Video Deployment for Cisco Unified MeetingPlace Release 8.0.

Determining which Audio and Video Deployment to Use

You choose the audio and video deployment to use based on the type of endpoints that you offer.

For a list of the video endpoints that Cisco Unified MeetingPlace Release 8.0 supports, see the *Compatibility Matrix for Cisco Unified MeetingPlace Release 8.0* at: http://docwiki.cisco.com/wiki/Cisco_Unified_MeetingPlace_Release_8.0_--_Compatibility_Matrix_for_Cisco_Unified_M

Audio and Video Deployment Option 1: Using SCCP or SIP Video Endpoints

You integrate SCCP or SIP video endpoints by defining a SIP trunk on Cisco Unified Communications Manager and Cisco Unified MeetingPlace. Cisco Unified MeetingPlace supports SIP delay-offer, thus a static Media Termination Point (MTP) is optional with all calls across a SIP trunk between Cisco Unified Communications Manager and Cisco Unified MeetingPlace. You can also create a separate SIP trunk security profile, with the outbound transport type set to TCP, and associate it with the SIP trunk to Cisco Unified MeetingPlace. The SIP trunk routes all outbound and inbound calls through a single Cisco Unified Communications Manager cluster.

Examples of SCCP and SIP endpoints are as follows:

- SCCP: Cisco Unified Video Advantage, Cisco Unified IP Phone 7985
- SIP: Cisco Unified Personal Communicator with Video Camera, Cisco Unified Communications Integration for Microsoft Office Communicator (CUCIMOC) with Video Camera Release 8.x, and Cisco Unified IP Phone 9900 series with Video Camera

This deployment uses Cisco Unified Communications Manager Release 6.1.4.39. The call path is as follows:

1. SIP or SCCP (voice or video) endpoint
2. Cisco Unified Communications Manager Release 6.1.4.39
3. Cisco Unified MeetingPlace

Audio and Video Deployment Option 2: Using H.323 Video Endpoints

Cisco Unified MeetingPlace Release 8.0 does not provide native support for H.323 endpoints. If you want to use H.323 endpoints, you must use RAS (Registration, Admission, and Status) and an IOS gatekeeper. The IOS gatekeeper sends all video calls through via Cisco Unified Communications Manager Release 6.1.4.39 to convert the H.323 signaling to SIP. This deployment accommodates existing deployed video solutions. All H.323 video endpoints must register to a IOS gatekeeper. Some video features currently supported by H.323 may not work with SIP. This deployment option also includes support for third party endpoints.

This deployment uses an IOS gatekeeper-controlled call-control domain which integrates with Cisco Unified Communications Manager Release 6.1.4.39. The call path is as follows:

1. H.323 endpoint
2. H.323/RAS signaling
3. IOS gatekeeper
4. Cisco Unified Communications Manager Release 6.1.4.39
5. SIP
6. Cisco Unified MeetingPlace

Audio and Video Deployment Option 3: Using SIP, SCCP, and H.323 Video Endpoints

This deployment is for systems that will use SIP, SCCP, and H.323 video endpoints. It is a combination of audio and video deployment options 1 and 2.

Choosing Between the Hardware Media Server and the Express Media Server

- About the Hardware Media Server and the Express Media Server
- General Differences
- Differences in Audio and Video Codecs
- Line Echo Cancellation
- About Ad-Hoc Conferencing

About the Hardware Media Server and the Express Media Server

Cisco Unified MeetingPlace Release 8.0 introduces an Express Media Server, a software version of the Hardware Media Server that was used in previous releases of Cisco Unified MeetingPlace. All systems that will use audio or video need a media server and the media server can be either a Hardware Media Server or an Express Media Server.

- The Express Media Server is a set of software modules, including an audio mixer and a video switcher, that resides on the Application Server. The Express Media Server creates a single box software-only solution for Cisco Unified MeetingPlace. The Express Media Server is based on the Cisco Unified MeetingPlace Express Video Telephony (VT) product.
- The Hardware Media Server is comprised of Audio and Video Blades.

All Cisco Unified MeetingPlace Release 8.0 systems automatically come with an Express Media Server. If you want to use a Hardware Media Server, you must purchase, install, and configure it first.

The Express Media Server and the Hardware Media Server cannot be used together in the same system; however, you can easily switch from one to the other.

Before you can choose between a Hardware Media Server and the Express Media Server, you should review the differences between the two.

- The Express Media Server can be deployed in ad-hoc mode, scheduled mode, or a combination of the two modes.
- The Hardware Media Server can only be deployed in scheduled mode.

There can be more or less differences based on your configuration and the environment in which the media servers are deployed.

General Differences

Table: General Differences Between the Express Media Server and the Hardware Media Server

Feature	Express Media Server	Hardware Media Server
Type	Software residing on the Application Server	Hardware, comprised of Audio and Video Blades. For video to work, each Audio Blade must have a Video Blade associated with it.
Installation	Installed automatically when you install the Application Server	Separate installation required
Configuration	Configured through the Administration Center of the Application Server	Configured through the Media Server Administration or Media Server Administrator
Conferencing	Supports scheduled and ad-hoc	Supports scheduled only
Recording	Cannot record the audio or video	Can record the audio and video

	portions of ad-hoc meetings. Can record the audio portion of scheduled meetings. Cannot record the video portion of scheduled meetings. Can support 20 simultaneous recordings with each recording using one port.	portions of scheduled meetings.
Cascading	Does not support any internal cascading of audio and video data	Uses internal cascading for scalability
Resource management	Based on the number of System Resource Units (SRUs), which is based on the type of Cisco MCS on which the Application Server is installed. High complexity codecs are used on a first come, first served basis. The number of available audio ports does not decrease when the user configures a high complexity audio codec.	Based on the following: • The number of Audio and Video Blades that are physically installed on the chassis. • For audio: The global audio mode (higher capacity or higher quality) that is set for the system. • For video: The mode (standard or high rate) that is set at the user profile level. The number of available audio ports decreases to two-thirds when the system is configured for high-quality mode.
High complexity codecs	G.729 is considered a high complexity codec	G.729 is *not* considered a high complexity codec unless line echo cancellation is turned on for it.
Video composition	Not supported. All non-speaker participants see the video of the active speaker only and the active speaker sees the video of last speaker.	Supports video composition. All non-speaker participants see the video of the last N speakers (where N is based on the layout selected by the system administrator). N speakers see N-1 remaining speakers (minus themselves) and one additional participant. N participants are composed into a single layout.
Muting		

	When audio is muted, video is not muted.	When audio is muted, video is muted.
Transrating	Not supported. The system uses flow-control mechanisms to force all connections to the same bandwidth. It restricts a meeting to use only a range of the bandwidth to accommodate the lowest speed participant in the meeting.	Supports true transrating, so you can have an ISDN connection in the same meeting as a 2Mb connection, without affecting the interaction between high rate participants.
Transcoding between H.263 and H.264 AVC	Not supported between H.263 and H.264 AVC endpoints in the same meeting. A meeting can include either H.263 or H.264 video endpoints.	Supported.
H.261	Not supported.	Supports H.261 in high rate mode and provides transcoding between H.261 and H.263/H.264.
Video resolution	Supports the following: • 320x180 • 640x360 (for Cisco soft clients) • 640x480 (for Roundtable phones) • 1280x720 (720p) for Cisco soft clients • CIF, QCIF	Allows 320x180, but it will probably be mangled. Does not allow the other resolutions. Works with QCIF, QSIF, SIF, and CIF and provides transcoding between them.
Custom video types	Can make custom video types. Has more predefined video types.	Not supported.
Video type management	Supported. Users can configure their own video types for a meeting (codec and bitrate)	Not supported. Only two system defaults are available: standard rate and high rate.
H.263 at 4CIF	Not supported.	Supported.
Echo cancellation, in-band (voice band) DTMF detection, audio codec iLBC, Automatic gain control (AGC), jitter buffer configuration	Not supported.	Supported.

Differences in Audio and Video Codecs

Audio Codecs Supported	Video Codecs Supported	
Hardware Media Server	• G.711 • G.722[1] • G.729a[1] • G.729b[1]	• H.261 • H.263 • H.264/AVC[2]

	• iLBC[1]	
Express Media Server	• G.711 • G.729a[3] • G.722[3]	• H.263 • H.264/AVC

1. When using these codecs, capacity changes from 250 ports per blade to 166 ports per blade.

2. Any combination in the same meeting.

3. For more information on using these codecs and capacity, see the Resource Management and System Capacity for Systems Using the Express Media Server.

Line Echo Cancellation

The Express Media Server does not include an echo canceller for controlling echo on incoming audio connections, so any echo originating from a phone or long distance connection can disrupt the conference. In general, echo cancellation is not required in a conference bridge, providing the following conditions are true:

1. All the voice gateways connecting the public switched telephone network to the internal network are provisioned to provide echo cancellation. Usually 64ms of echo cancellation is sufficient, but intercontinental calls may benefit from 128ms.
2. Substantially all calls are originating and terminating within the same continent. If you have 128ms of echo cancellation in the voice gateways, then intercontinental calls between developed countries will likely be covered, provided no connections through satellites are employed. Intercontinental calls involving 3rd world countries are likely to have echo exceeding 128ms., which cannot be controlled by most voice gateways.
3. Internal phones and headsets are well maintained. Defective phones or headsets, including echo cancelling headsets with dead batteries, are a common source of echo.

Installations needing echo cancellation in the conference bridge should employ a Hardware Media Server instead of the Express Media Server.

Note that acoustic echo, typically from a speakerphone in a conference room, cannot be effectively cancelled by either type of media server. This type of echo should be controllable through proper configuration and use of a good quality speaker phone.

About Ad-Hoc Conferencing

- Overview of Ad-Hoc Conferencing
- How the System Uses Ad-Hoc Voice Ports
- Recommendations for Users Using Ad-Hoc Conferencing

Overview of Ad-Hoc Conferencing

Ad-hoc meetings are not controlled by Cisco Unified MeetingPlace. Cisco Unified Communications Manager manages and controls the exchange of conference control messages and the voice and video media control messages between endpoints in ad-hoc meetings. The Cisco Unified MeetingPlace Express Media Server, which contains an audio mixer and a video switcher, acts as a conference bridge to Cisco Unified Communications Manager and provides only the media resources.

Users initiate ad-hoc meetings through the user interfaces of products other than Cisco Unified MeetingPlace. For example, users can initiate either a voice-only or a voice-and-video ad-hoc meeting by using the Conf or Meet-Me buttons on Cisco Unified IP Phones that are registered to Cisco Unified Communications Manager. Users can also add web conferencing to conversations initiated through Cisco Unified Personal Communicator.

Note: Users with H.323 video terminals, which are registered to a Cisco IOS gatekeeper and have an H.225 trunk to Cisco Unified Communications Manager, can also initiate ad-hoc conferences. There is no CONF button on the terminals, but once users initiate an ad-hoc conferences from Cisco endpoints using the Meet-Me number, they can call in to the same ad-hoc conf from the terminals.

In contrast, scheduled and reservationless meetings are set up, managed, and controlled by Cisco Unified MeetingPlace. Users can schedule meetings through the Cisco Unified MeetingPlace user web interface or through the Microsoft Outlook plug-in and can call in to or call out of the meeting by using the phone, the user web interface, or video endpoints.

By configuring your Cisco Unified MeetingPlace server as a Cisco video conference bridge (IPVC-35xx) in Cisco Unified Communications Manager, you can provide ad-hoc conferencing for the these endpoints:

- All voice and video endpoints that support Cisco Unified Communications Manager meet-me and ad-hoc conferences-voice and video only.
- Cisco Unified Personal Communicator-voice, web, and video.

Ad-Hoc Meetings	Scheduled/Reservationless Meetings
Set up, managed, and controlled by Cisco Unified Communications Manager	Set up, managed, and controlled by Cisco Unified MeetingPlace
Meetings scheduled through IP phones or Cisco Unified Personal Communicator	Meetings scheduled through the Cisco Unified MeetingPlace end user web interface or through Microsoft Outlook
	Call into or out of meetings by phone, web, or video endpoints
Does not support RSNA	Supports RSNA
Does not support recording	Supports recording

Related Topics

- For information about meet-me and ad-hoc conferences, see the "Conference Bridges" chapter of the *Cisco Unified Communications Manager System Guide* for your specific release, go to http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.
- For details about Cisco Unified Personal Communicator, go to http://www.cisco.com/en/US/products/ps6844/index.html.

How the System Uses Ad-Hoc Voice Ports

The Cisco Unified MeetingPlace system uses ad-hoc voice ports in these situations:

- One port for each endpoint in a voice call that includes three or more endpoints. (A direct, voice-only call between two endpoints does not use any ad-hoc voice ports.)
- One port when a Cisco Unified IP Phone is used to create the meeting through the Meet-Me button.
- Another port for each additional endpoint that calls into the ad-hoc conference.
- One voice port for each utilized video port.

Recommendations for Users Using Ad-Hoc Conferencing

The video switcher of the Express Media Server switches the video stream to the current active speaker, which is the *loudest* speaker as determined by the audio mixer.

Provide these recommendations to help reduce background noise during conference calls:

- Mute your phone when you are not speaking.
- Do not use a speakerphone, which might generate echoes, ringing sounds, or audio feedback.
- If you use a microphone that is built into your video camera, keep the camera away from fans, vents, and other sources of noise.
- For softphones, such as Cisco Unified Personal Communicator with a Cisco VT Camera or Cisco IP Communicator with Cisco Unified Video Advantage:
    - Do not use the microphone that is built into your computer. These microphones tend to pick up a lot of background noise.
    - We strongly recommend that you use a headset that is equipped with a microphone.
    - Whenever multiple microphones are available, make sure that your computer and video endpoint are configured to use the desired microphone.

      For example, suppose that you use Cisco Unified Personal Communicator with a Cisco VT Camera on a Windows XP operating system, and that you have a headset that is equipped with a microphone. To make sure that your system is configured to use the headset microphone, select **Start > Control Panel > Sounds and Audio Devices**. Then select the **Audio** tab, and make sure that your headset is selected as the sound recording device.
      See the documentation for your specific endpoint product to optimize audio settings and resolve audio problems.