



Cisco TelePresence Interoperability

Cisco TelePresence interoperability allows traditional room-based video conferencing systems as well as desktop video collaboration systems to participate within a Cisco TelePresence meeting. This provides the ability to leverage existing investments in traditional video conferencing equipment, while still maintaining the immersive TelePresence experience. Cisco TelePresence interoperability highlights an initial step in the deployment of a medianet, in which various types of video co-exist over a shared IP network infrastructure.

TelePresence and traditional video conferencing are fundamentally different experiences, and are often maintained as separate environments at present. Cisco TelePresence is specifically designed to maintain the in-person experience through life-size video of conference participants and spatial audio. However, many traditional room-based video conferencing systems are designed to show the entire room at once, often with only a single accompanying audio channel. As a result, conference participants are scaled to fit within the viewing display area. Likewise, desktop video collaboration systems are typically designed around limiting the amount of bandwidth used per device. Camera quality, acoustics, lighting, and background may not be as tightly controlled with desktop video collaboration; limiting the in-person experience.

This document discusses current standard definition (SD) and high definition (HD) video interoperability with Cisco TelePresence devices running CTS 1.6 software; using Cisco Unified Video Conferencing (CUVC) multipoint control units (MCUs). Future revisions of this document may discuss interoperability using other means such as the Cisco Media Experience Engine (MXE) platform, as well as direct interoperability of video conferencing devices that implement the Cisco TelePresence Interoperability Protocol (TIP).

Contents

Interoperability Overview	2
Cisco TelePresence Standard Definition (SD) Interoperability	3
SD Interoperability Media and SIP Signaling Flows	4
Cisco TelePresence High Definition (HD) Interoperability	6
HD Interoperability Media and SIP Signaling Flows	8
Static TelePresence Meetings with Interoperability	9

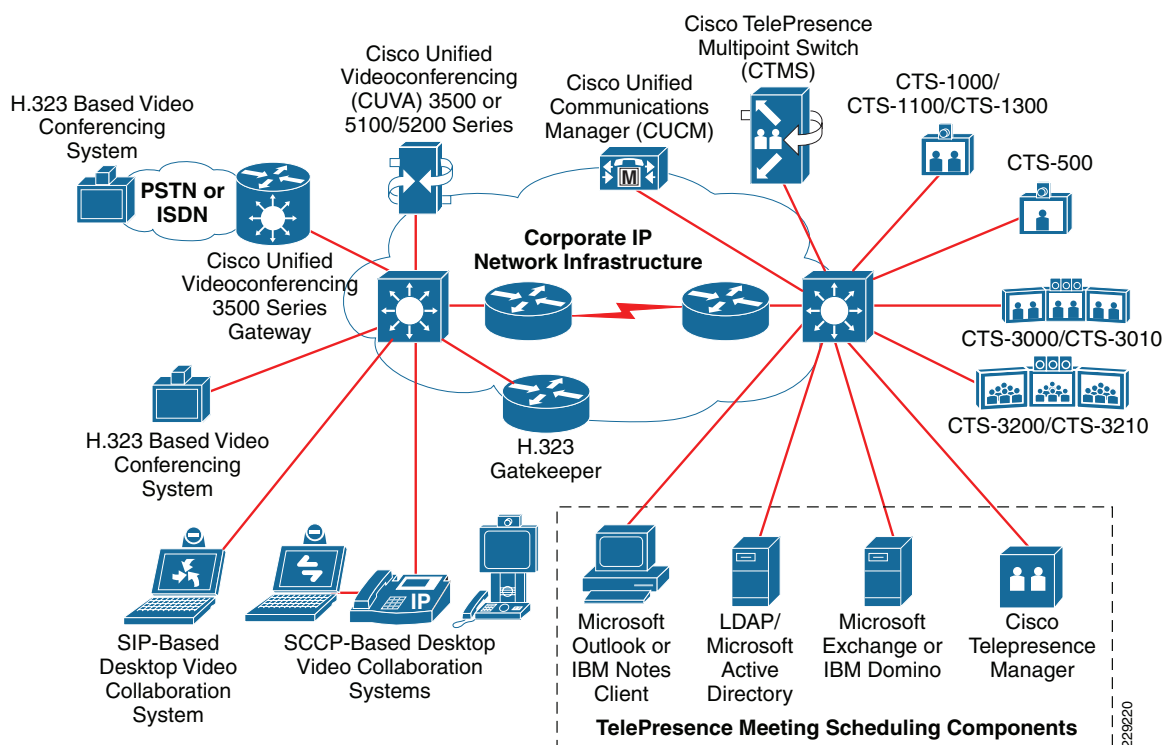


Scheduled TelePresence Meetings Interoperability	13
Network Design Implications	18
Additional Bandwidth Considerations	19
Interoperability Latency Considerations	27
Interoperability Burst Considerations	29
Interoperability QoS Considerations	30
Summary	35

Interoperability Overview

The Cisco TelePresence interoperability solution retains the rich, immersive experience between Cisco TelePresence participants while providing a bridge to existing room-based video conferencing and desktop video collaboration systems. This is accomplished by cascading the Cisco TelePresence Multipoint Switch (CTMS) and the Cisco Unified Video Conferencing (CUVC) MCU together. [Figure 1](#) highlights the components required for Cisco TelePresence interoperability.

Figure 1 **Components of a Cisco TelePresence Meeting with Interoperability**



The Cisco TelePresence solution provides standards-based interoperability with minimal additional hardware requirements, at either standard definition (Common Intermediate Format [CIF]) or high definition (720p) video resolution. Each is discussed separately in the following sections.

Cisco TelePresence Standard Definition (SD) Interoperability

The components required for Cisco TelePresence SD interoperability, as shown in [Figure 1](#), are as follows:

- One or more Cisco TelePresence System endpoints (CTS-500, CTS-1000, CTS-1100, CTS-1300, CTS-3010, CTS-3200, or CTS-3210) running CTS software version 1.4 or higher. (Note that the CTS-1100 and CTS-1300 are supported only with CTS software version 1.5 or higher).
- A Cisco TelePresence Multipoint Switch (CTMS) running software version 1.4 or higher.
- A Cisco TelePresence Manager running software version 1.4 or higher is required for scheduled interoperability meetings. (Note that for static interoperability meetings, the Cisco TelePresence Manager is not required.)
- A Cisco Unified Video Conferencing 3515 MCU or a Cisco Unified Video Conferencing 3545 System functioning as an MCU, running version 5.5 or higher. (Note that a Cisco Unified Video Conferencing 5110, 5115, or 5230 System running version 7.0 or higher can also be used for SD interoperability as well.)
- One or more traditional video conferencing systems (LAN, PSTN, or ISDN-based); or desktop video collaboration systems, such as Cisco Unified Video Advantage (CUVA), the Cisco IP 7985G Video Phone, or Cisco Unified Personal Communicator (CUPC).

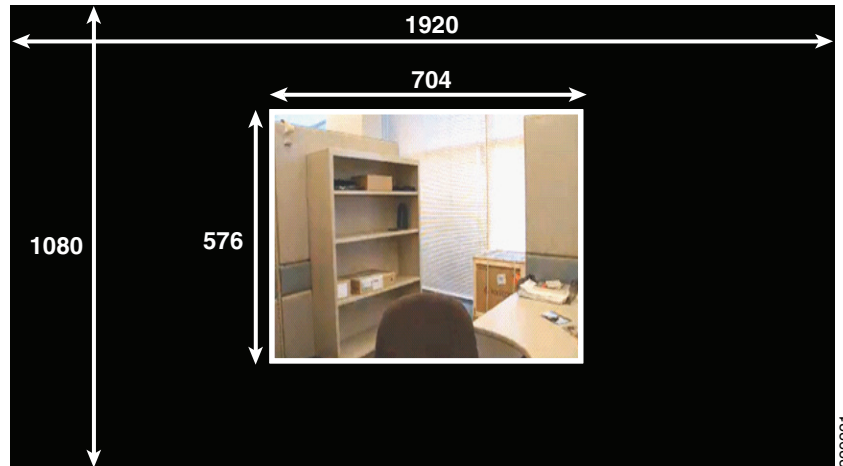
Optional components include the following:

- One or more Cisco gateways for interoperability between H.323 (LAN-based) and H.320 (ISDN-based) video conferencing systems.
- A Cisco IOS gatekeeper (ISR router) for E.164 address resolution, call routing, and call admission control of H.323 based systems.

MCUs other than the Cisco Unified Videoconferencing (CUVC) System are currently not supported for interoperability between Cisco TelePresence devices.

Cisco TelePresence SD interoperability is achieved through an active cascade segment created between the CTMS and CUVC 3500 Series MCU. The interoperability cascade segment between the CTMS and CUVC 3500 Series MCU is limited to CIF resolution at 30 frames per second. CIF has a picture size of 352 x 288 pixels. The CIF video sent by the CUVC 3500 Series MCU (and switched by the CTMS to the CTS endpoint) is scaled to 4CIF (576 x 704 pixels) by the CTS endpoint codec before being displayed on the TelePresence display. The codec surrounds the remaining space with a black border, as seen in [Figure 2](#), where the TelePresence call is running 1080p video resolution (1920 x 1080 pixels).

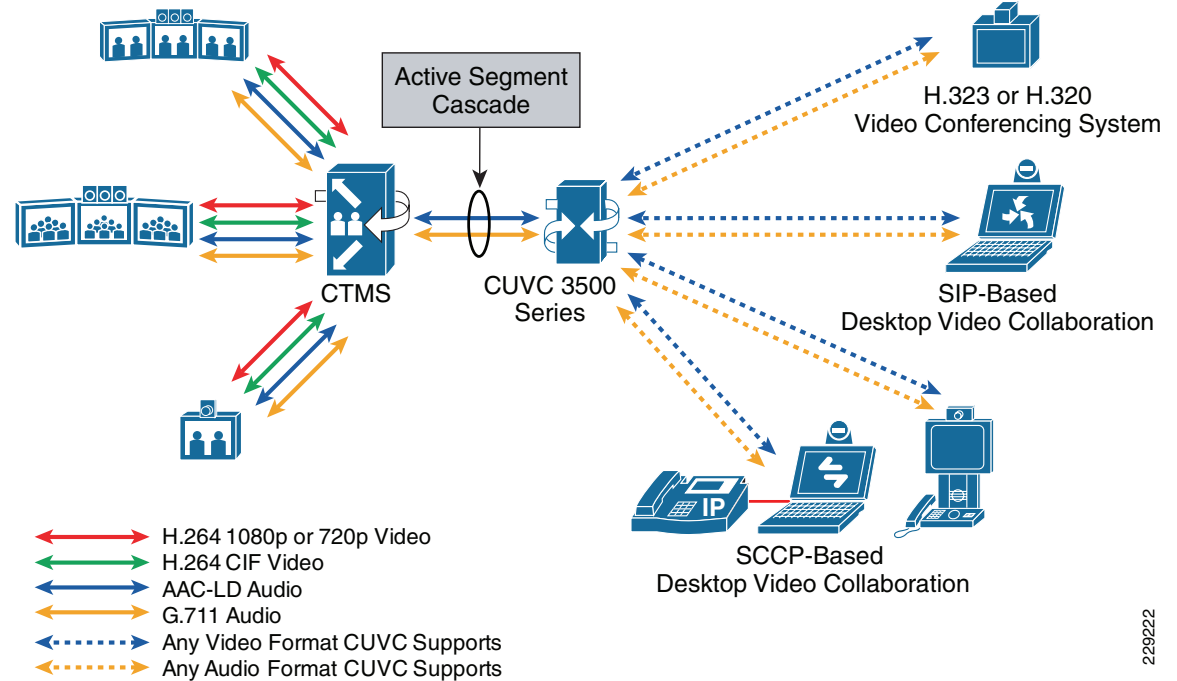
Figure 2 *Cisco TelePresence Standard Definition Interoperability—User Experience*



The Auto Collaborate feature of Cisco TelePresence, which allows TelePresence rooms to share PowerPoint slides or document camera images, does not interoperate with the H.239 signaling on the CUVVC 3500 Series MCU. Therefore, the Auto Collaborate images are not seen by the non-Cisco TelePresence endpoints. Participants may choose to run a separate collaboration session, such as Cisco WebEx, along with a Cisco TelePresence meeting that includes interoperability. Also, the Far End Camera Control (FECC) feature of the CUVVC 3500 Series MCU that allows pan, tilt, or zoom of the cameras of a traditional video conferencing system is not available to TelePresence participants. In other words, the TelePresence participants are not able to remotely control the camera of the traditional video conferencing system.

SD Interoperability Media and SIP Signaling Flows

Figure 3 shows a high-level view of the media flows in a Cisco TelePresence meeting that includes SD interoperability.

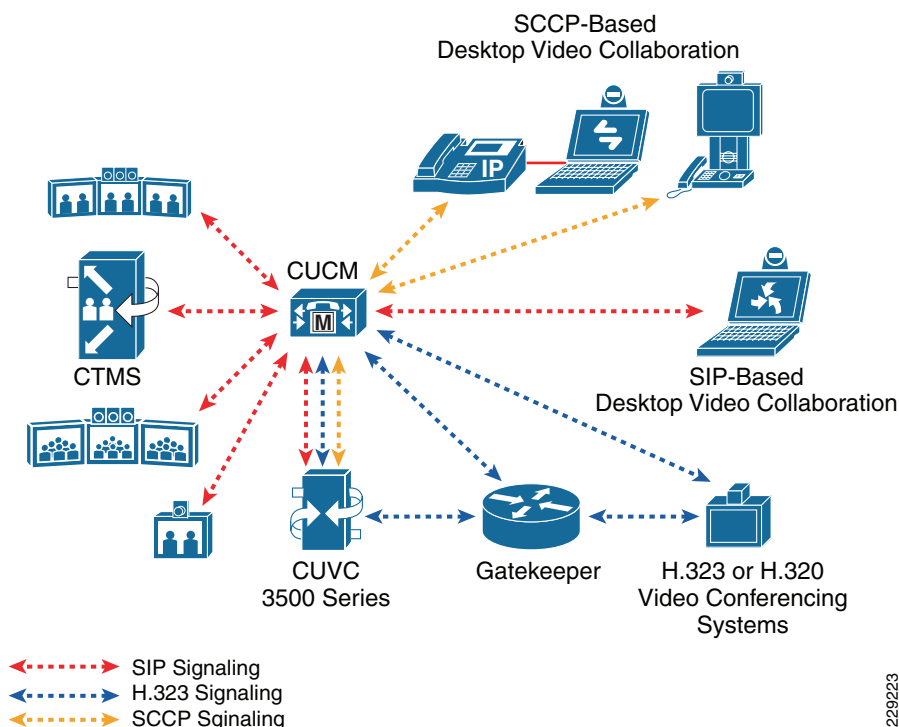
Figure 3 Cisco TelePresence Standard Definition Interoperability Media Flows

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Each CTS endpoint sends a G.711 encoded audio stream to the CTMS. This G.711 stream is in addition to the normal AAC-LD audio streams sent by each CTS endpoint to the CTMS for TelePresence meetings. The CTMS uses voice activity detection within the AAC-LD audio streams to determine which CTS endpoint is the active speaker. If one of the CTS endpoints (or individual codec in a multi-screen system) is the active speaker, the CTMS instructs that CTS endpoint to send an H.264 CIF video stream, in addition to the normal H.264 1080p or 720p video streams. Only the G.711 audio and H.264 CIF video streams are switched by the CTMS to the CUVVC 3500 Series MCU, via the active cascade segment. The normal TelePresence AAC-LD audio streams and the H.264 1080p or 720p video streams are switched by the CTMS to the other CTS endpoints. When the CUVVC 3500 Series MCU receives the G.711 audio and H.264 CIF video streams, it can either directly switch the streams to each of its video conferencing endpoints, or it can first transcode the streams to whatever formats are supported for that particular meeting by the traditional video conferencing endpoints.

The CTMS also receives both a G.711 audio stream and an H.264 CIF video stream from the CUVVC 3500 Series MCU via the active cascade segment. The CTMS determines whether one of the traditional video conferencing endpoints should be the active speaker, based on the audio energy within the G.711 stream. When one of the traditional video conferencing endpoints becomes the active speaker, the G.711 audio and H.264 CIF video streams are then switched by the CTMS to each of the CTS endpoints within the conference.

Figure 4 shows a high-level view of the call signaling flows in a Cisco TelePresence SD meeting that includes interoperability.

Figure 4 Cisco TelePresence Standard Definition Interoperability Call Signaling Flows

The example in [Figure 4](#) has been somewhat simplified, with a single CUCM cluster shown supporting both Cisco TelePresence and legacy video conferencing systems. Network administrators should note that the quality of service (QoS) setting for video media is a system-wide parameter with CUCM version 8.0 and below. Therefore, a single CUCM cluster is not able to instruct the various CUCM-controlled desktop video collaboration endpoints (IP 7985G phones, CUVA, and CUPC) to mark audio and video with different DSCP markings. Interoperability can also be configured with separate CUCM clusters: one supporting Cisco TelePresence; and the other supporting H.323 room-based video conferencing and desktop video collaboration. In this scenario, an SIP trunk is defined between the two CUCM clusters to carry the call signaling.

In the example shown in [Figure 4](#), SIP call signaling occurs between the Cisco TelePresence endpoints and the CUCM. Likewise, call signaling (SIP, H.323, or SCCP) occurs between traditional video conferencing systems, and desktop video collaboration systems and the CUCM (or a gatekeeper for H.323 devices). At the beginning of a TelePresence interoperability meeting, when the first CTS endpoint dials into the CTMS to join the meeting, the CTMS dials out to the CUCV 3500 Series MCU by way of the CUCM to establish the active cascade segment. Additional TelePresence endpoints dial into the CTMS as if they were attending a typical TelePresence meeting. Likewise, traditional video conferencing endpoints and/or desktop video collaboration endpoints dial into the CUCV 3500 Series MCU as if they were attending a typical video conference.

Cisco TelePresence High Definition (HD) Interoperability

The components required for Cisco TelePresence HD interoperability, as shown in [Figure 1](#), are as follows:

- One or more Cisco TelePresence System endpoints (CTS-500, CTS-1000, CTS-1100, CTS-1300, CTS-3010, CTS-3200, or CTS-3210) running CTS software version 1.6 or higher.

- A Cisco TelePresence Multipoint Switch (CTMS) running software version 1.6 or higher.
- A Cisco TelePresence Manager running software version 1.6 or higher is required for scheduled interoperability meetings. (Note that for static interoperability meetings, the Cisco TelePresence Manager is not required.)
- A Cisco Unified Video Conferencing 5110 or 5115 MCU, or a Cisco Unified Video Conferencing 5230 System functioning as an MCU, running version 7.0 or higher.
- One or more traditional video conferencing systems (LAN, PSTN, or ISDN-based); or desktop video collaboration systems, such as Cisco Unified Video Advantage (CUVA), the Cisco IP 7985G video phone, or Cisco Unified Personal Communicator (CUPC).

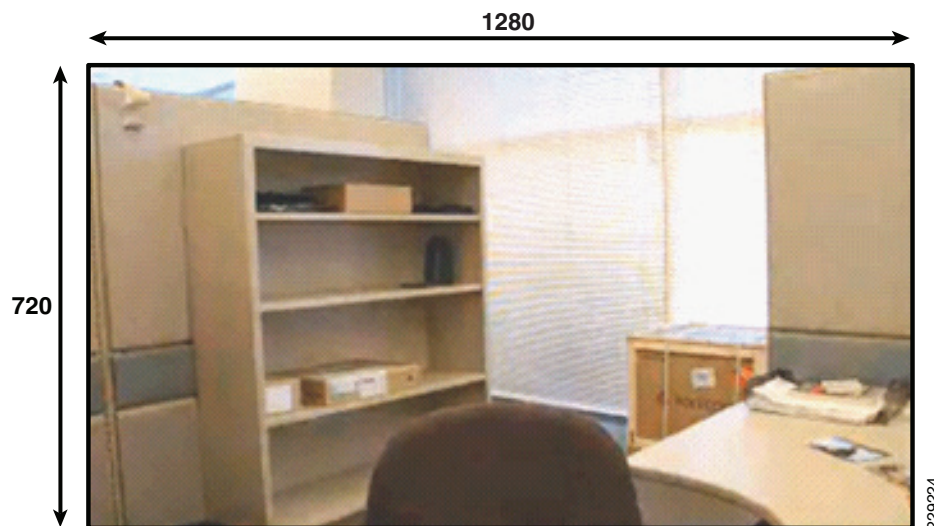
Optional components include the following:

- One or more Cisco gateways for interoperability between H.323 (LAN-based) and H.320 (ISDN-based) video conferencing systems
- A Cisco IOS gatekeeper (ISR router) for E.164 address resolution, call routing, and call admission control of H.323-based systems

MCUs other than the Cisco Unified Videoconferencing (CUVC) System are currently not supported for interoperability between Cisco TelePresence devices.

Cisco TelePresence HD interoperability is achieved through an active cascade segment created between the CTMS and CUVC 5100/5200 Series MCU. The interoperability cascade segment between the CTMS and CUVC 5100/5200 Series MCU is limited to 720p resolution (1028 x 720 pixels) at 30 frames per second. Because the interoperability segment runs at 720p HD as well, video from non-TelePresence devices fills the entire screen of the TelePresence endpoints. Therefore, the end-user experience provided by HD interoperability is much more similar to an all-TelePresence multipoint meeting, as shown in the example in [Figure 5](#).

Figure 5 *Cisco TelePresence High Definition Interoperability—User Experience*



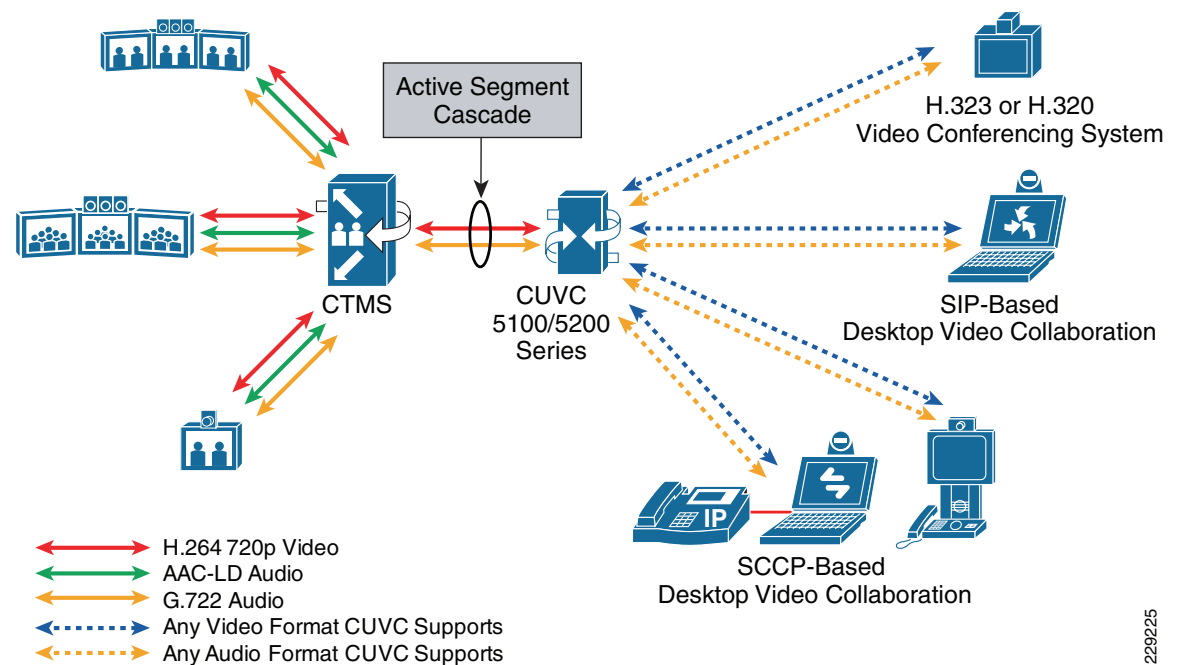
The Auto Collaborate feature of Cisco TelePresence, which allows TelePresence rooms to share PowerPoint slides or document camera images, does not currently interoperate with the H.239 signaling on the CUVC 5100/5200 Series MCU. Therefore, the Auto Collaborate images are not seen by the non-Cisco TelePresence endpoints. Participants may choose to run a separate collaboration session, such as Cisco WebEx, along with a Cisco TelePresence meeting that includes interoperability. Also, the Far End Camera Control (FECC) feature of the CUVC 5100/5200 Series MCU that allows pan, tilt, or zoom

of the cameras of a traditional video conferencing system is not available to TelePresence participants. In other words, the TelePresence participants are not able to remotely control the camera of the traditional video conferencing system.

HD Interoperability Media and SIP Signaling Flows

Figure 6 shows a high-level view of the media flows in a Cisco TelePresence HD meeting that includes interoperability.

Figure 6 Cisco TelePresence High Definition Interoperability Media Flows



Each CTS endpoint sends a G.722 encoded audio stream to the CTMS. This G.722 stream is in addition to the normal AAC-LD audio streams sent by each CTS endpoint to the CTMS for TelePresence meetings. The CTMS uses voice activity detection within the AAC-LD audio streams to determine which CTS endpoint is the active speaker. If one of the CTS endpoints (or individual codec in a multi-screen system) is the active speaker, the CTMS instructs that CTS endpoint to send its H.264 720p video stream. This video stream, along with a mix of audio from the top three G.722 audio streams, is switched by the CTMS to the CUV 5100/5200 Series MCU, via the active cascade segment. The normal TelePresence AAC-LD audio streams and the H.264 720p video stream are switched by the CTMS to the other CTS endpoints. When the CUV 5100/5200 Series MCU receives the G.722 audio and H.264 720p video streams, it can either directly switch the streams to each of its video conferencing endpoints, or it can first transcode the streams to whatever formats are supported for that particular meeting by the traditional video conferencing endpoints.

The CTMS also receives both a G.722 audio stream and an H.264 720p video stream from the CUV 5100/5200 Series MCU via the active cascade segment. The CTMS determines whether one of the traditional video conferencing endpoints should be the active speaker, based on the audio energy within the G.722 stream. When one of the traditional video conferencing endpoints becomes the active speaker, the audio and video streams are then switched by the CTMS to each of the CTS endpoints within the conference.

Figure 7 shows a high-level view of the call signaling flows in a Cisco TelePresence HD meeting that includes interoperability.

Figure 7 Cisco TelePresence High Definition Interoperability Call Signaling Flows

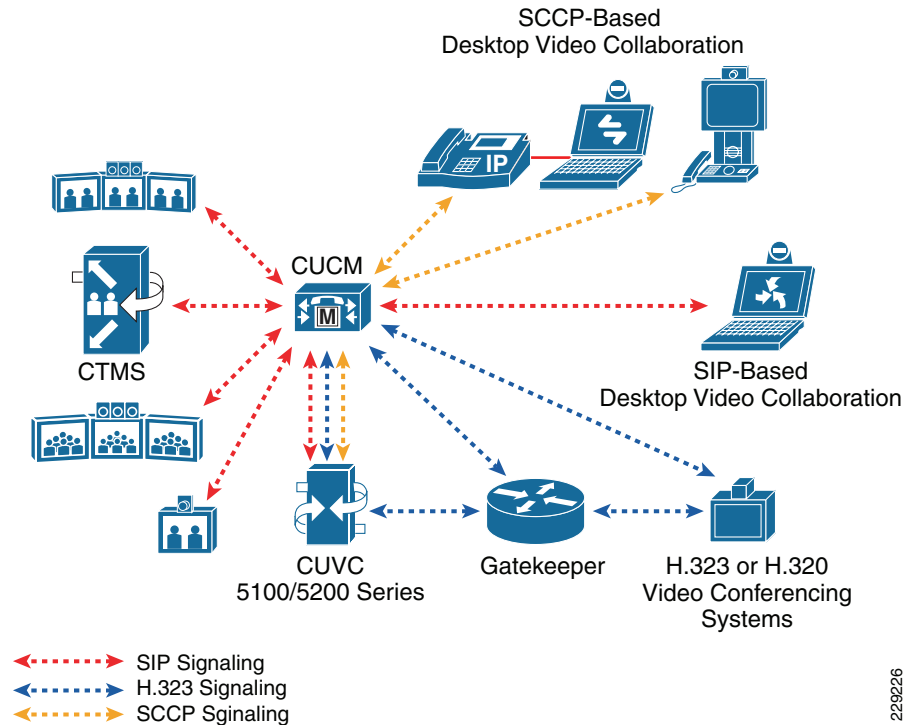


Figure 7 is nearly identical to Figure 4, except that CUCV 5100/5200 Series MCU is required for HD interoperability. SIP call signaling occurs between the Cisco TelePresence endpoints and the CUCM. Likewise, call signaling (SIP, H.323, or SCCP) occurs between traditional video conferencing systems and desktop video collaboration systems and the CUCM (or a gatekeeper for H.323 devices). At the beginning of a TelePresence interoperability meeting, when the first CTS endpoint dials into the CTMS to join the meeting, the CTMS dials out to the CUCV 5100/5200 Series MCU by way of the CUCM to establish the active cascade segment. Additional TelePresence endpoints dial into the CTMS as if they were attending a typical TelePresence meeting. Likewise, traditional video conferencing endpoints and/or desktop video collaboration endpoints dial into the CUCV 5100/5200 Series MCU as if they were attending a typical video conference.

Cisco TelePresence interoperability meetings can be configured to be either static meetings or scheduled meetings from the perspective of the TelePresence deployment. Each is discussed in the following sections.

Static TelePresence Meetings with Interoperability

For static TelePresence meetings that include interoperability, the CTMS knows which service prefix to dial to establish the cascade segment to the CUCV MCU (3500 Series or 5100/5200 Series), based on the configuration of the TelePresence static meeting itself. Figure 8 shows the configuration of a static meeting on the CTMS that includes SD interoperability.

Figure 8 Sample CTMS Static Meeting Configuration with SD Interoperability

Meetings Management > Static Meetings

Access Number:	4085276005
Meeting ID:	9185276005
Meeting Description:	Interop with East 3545 *
Switching Policy:	☒ Room ☐ Speaker
Maximum Rooms:	10 *
Video Announce:	☒ Yes ☐ No
Hosted Meeting:	☐ Yes ☒ No
Host Room Number:	*:
Interop:	☒ Yes ☐ No
Resolution:	☒ CIF ☐ HD 720p
Interop Number:	9193927000 *
Quality:	Highest Detail, Best Motion: 1080p ▼
Meeting Security Policy:	☒ Non-Secured

Save Close

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The **Access Number:** field is the extension that TelePresence endpoints dial to attend the static TelePresence multipoint meeting configured on the CTMS. The **Interop Number:** field holds the extension corresponding to a service prefix configured within the CUV 3500 Series MCU. It allows the CTMS to dial outbound via the CUCM, and the CUCM to then route the call to the correct CUV MCU, to start the interoperability segment of the TelePresence multipoint meeting. The **Resolution:** field indicates to the CTMS whether the interoperability segment uses SD (CIF) or HD (720p) video. In other words, the **Resolution:** field determines whether it is an SD or HD interoperability meeting. The **Quality:** field determines the desired video resolution of the TelePresence endpoints only. For SD interoperability meetings, the network administrator or meeting scheduler can select the full range of either 1080p or 720p resolutions for the TelePresence endpoints.

Figure 9 shows the configuration of a static meeting on the CTMS that includes HD interoperability.

Figure 9 Sample CTMS Static Meeting Configuration with HD Interoperability

Meetings Management > Static Meetings

Access Number: 9193926004

Meeting ID: 9193926004

Meeting Description: Interop West 5230 *

Switching Policy: ☒ Room ☐ Speaker

Maximum Rooms: 10 *

Video Announce: ☒ Yes ☐ No

Hosted Meeting: ☐ Yes ☒ No

Host Room Number: *

Interop: ☒ Yes ☐ No

Resolution: ☐ CIF ☒ HD 720p

Interop Number: 4085279100 *

Quality: High Detail, Best Motion: 720p ▼

Meeting Security Policy: ☒ Non-Secured

Save Close

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The only difference between Figure 8 and Figure 9 is the **Quality** field. For HD interoperability meetings, the network administrator or meeting scheduler can select only 720p video resolutions for the TelePresence endpoints. TelePresence meetings at 1080p resolution, which also include HD interoperability, are not supported as of CTS version 1.6. Network administrators should also note that the only choice for the **Meeting Security Policy** field is *Non-Secured* for both HD and SD interoperability meetings. In other words, secured (that is, encrypted) TelePresence meetings that include interoperability are not supported as of CTS version 1.6.

For Cisco TelePresence interoperability, the CUVIC MCU looks like a SIP trunk to the CUCM, with a route pattern that includes the **Interop Number** defined within the CTMS. An example is shown in Figure 10.

Figure 10 Sample CUCM SIP Trunk Configurations

Cisco Unified CM Administration

Navigation: Cisco Unified CM Administration | Go

Administrator | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Find and List Trunks

+ Add New | Select All | Clear All | Delete Selected | Reset Selected | Apply Config to Selected

Status: 5 records found

Trunks (1 - 5 of 5) Rows per Page: 50

Find Trunks where Device Name begins with Find Clear Filter

Select item or enter search text

Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Security Profile
me-eastctms-1	East Campus CTMS		RTP Devices	919392600-1X				SIP Trunk	Non Secure SIP Trunk Profile
me-eastcuvic3545-1	East Campus CUVIC 3500		Interoperability	9193927XXX				SIP Trunk	Non Secure SIP Trunk Profile
tp-c2-cm-sip	Intercluster SIP Trunk		Default	922813XXXX				SIP Trunk	Non Secure SIP Trunk Profile
tp-c2-cm-sip	Intercluster SIP Trunk		Default	4085279100-#000				SIP Trunk	Non Secure SIP Trunk Profile
tp-c2-cm-sip	Intercluster SIP Trunk		Default	408527XXXX				SIP Trunk	Non Secure SIP Trunk Profile

Add New | Select All | Clear All | Delete Selected | Reset Selected | Apply Config to Selected

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In the example above, the SIP trunk called *me-eastcuvc3545-1* is a SIP trunk defined to a CUVVC 3500 Series MCU. The SIP trunk includes the entire range of dial extensions from 9193927000 through 9193927999. This allows the CUCM to “route” the incoming call from the CTMS to the CUVVC 3500 Series MCU to establish the active segment cascade between the CTMS and CUVVC 3500 Series MCU.

The CTMS also appears as a SIP trunk to the CUCM. In [Figure 10](#), it is shown as the SIP trunk called *me-eastctms-1*. This SIP trunk includes the entire range of dial extensions from 9193926000 through 9193926019.

When the CUVVC MCU is defined on a separate CUCM from the CTMS, the CUVVC MCU needs to be reachable via an SIP trunk defined between the two CUCMs. [Figure 10](#) also shows this example, highlighting another important aspect of HD interoperability. When the CTMS dials out to the CUVVC 5100/5200 Series MCU, the CTMS automatically appends the string *##00* to the dial extension. This additional string is used to signal to the CUVVC 5100/5200 Series that the call is an HD interoperability call from the CTMS. Call patterns on the CUCM must be configured such that the entire dial string is passed through any trunks to the CUVVC 5100/5200 series. In the example above, the entire dial-string *4085279100##000* is passed from the CTMS through the CUCM through the trunk to the second CUCM, and out to the CUVVC 5100/5200 Series MCU.

**Note**

Note that the example above was configured specifically to highlight this.

The service that supports the interoperability meeting must be pre-configured within the CUVVC MCU as well. A sample service definition for the CUVVC 3500 Series MCU running software version 5.5 is shown in [Figure 11](#).

Figure 11 Sample CUVVC 3500 Series Service Configuration for Interoperability

Automatic Service Definition

General

Service prefix: 91939271 ☐ SCCP service

Service description: TelePresence Interoperability

Service type: HD/SD Continuous Presence

Max call rate: 768 Kbps

Audio and video

Max layout: Change...

Support image size up to: 4CIF

Advanced Video Settings... Advanced Audio Settings...

Data collaboration

☒ Support presentation view (H.239) Settings...

☐ Support T120 data conferencing

☐ Allow access to data conferencing from MCU Conference Control

Management and security

☐ Support encryption

Encryption mode: Best effort

Advanced Management and Security...

Upload Cancel Help

Within the CUVVC MCU, a conference ID consists of a service prefix plus a conference number. In the example above, the CUVVC MCU matches the first eight digits of the number to the Service Prefix *91939271*. It then creates an ad hoc conference with a conference ID of *9193927100* for interoperability.

In this manner, multiple static meetings can be defined on the CTMS, each beginning with 91939271. When the static meetings become active, the CUVVC MCU simply creates new ad hoc meetings to support them.

Figure 12 shows the same configuration for a CUVVC 5100/5200 Series MCU running software version 7.0.

Figure 12 Sample CUVVC 5100/5200 Series Service Configuration for Interoperability

Prefix	Description	Max call rate (Kbps)	Review
408527	TelePresence Interoperability via SIP ☐ Audio only ☐ SCCP service ☒ Display welcome screen Display welcome message: Welcome to \$DESC	12288	Set as the Auto attendant service More... Delete Apply Cancel
40852781	TelePresence Interoperability	12288	Review
40852782	SCCP Video	768	Review

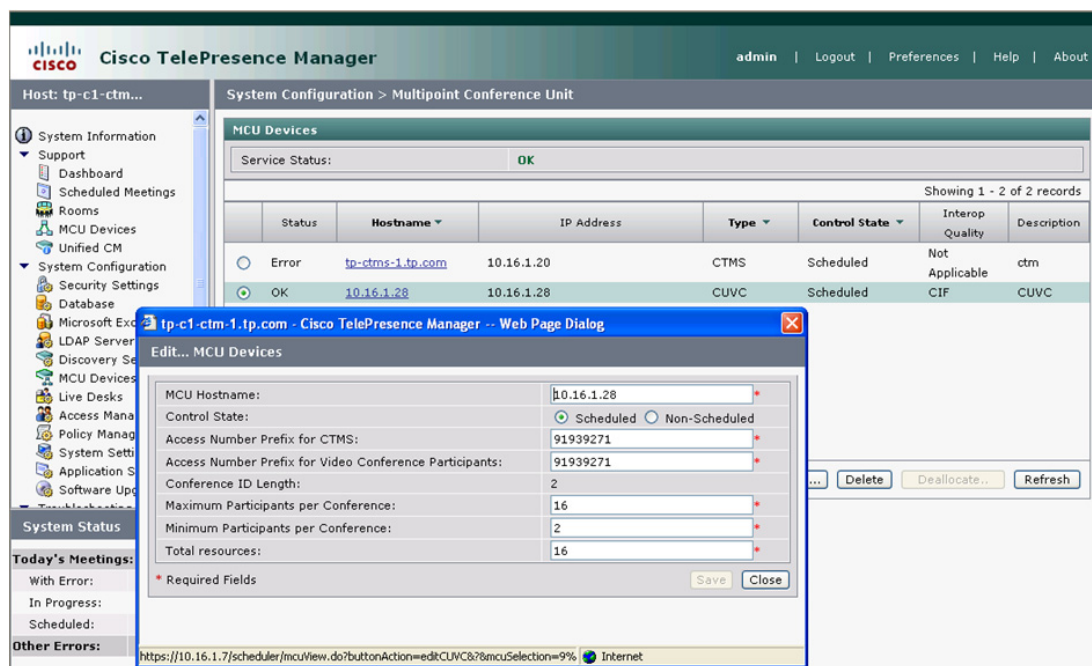
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Scheduled TelePresence Meetings Interoperability

Cisco TelePresence Manager version 1.4 or higher includes the capability to support scheduled multipoint meetings to the CUVVC 3500 Series MCU that include SD interoperability. Cisco TelePresence Manager version 1.6 or higher is required to support scheduled multipoint meetings to the CUVVC 5100/5200 Series MCU that include HD interoperability. Both allow the one-button-to-push feature used to start Cisco TelePresence meetings to be extended to meetings that include interoperability. To add interoperability for scheduled TelePresence meetings, the CUVVC MCU must first be added to Cisco TelePresence Manager as an additional MCU device, along with the CTMS.

Figure 13 shows an example of this.

Figure 13 Sample of CUVC Addition to Cisco TelePresence Manager

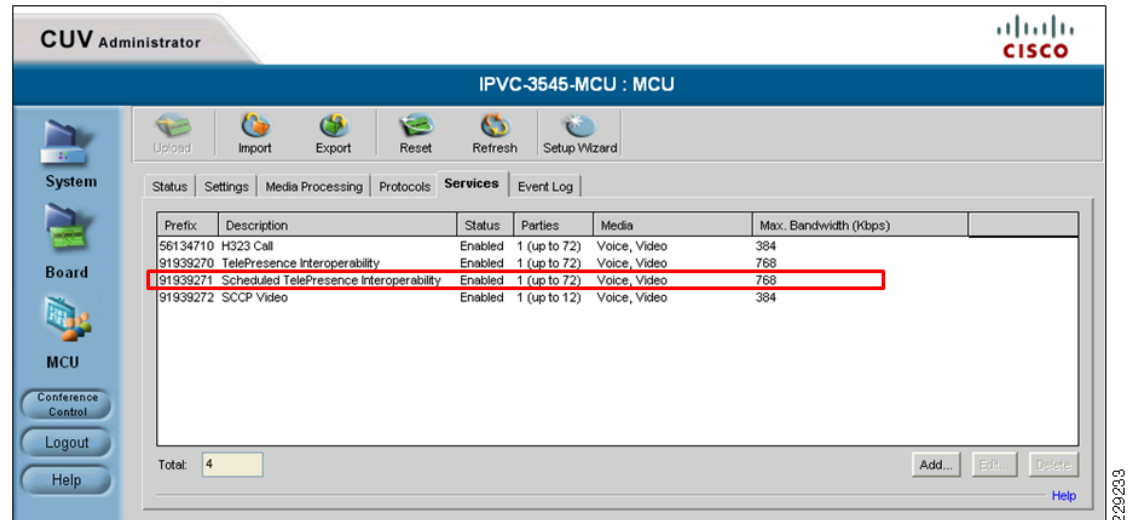


The **Access Number Prefix for CTMS** is the prefix dialed by the CTMS to establish the active segment cascade between itself and the CUVC MCU. The **Access Number Prefix for Video Conferencing Participants** is the prefix dialed by traditional video conferencing systems and desktop video collaboration devices to join the scheduled TelePresence interoperability meeting. Each of these fields is combined with a random conference number generated by the Cisco TelePresence Manager to form the complete conference ID. The length of the conference number is determined by the **Conference ID Length** field. In the example above, the Cisco TelePresence Manager combines the following:

Prefix of 91939271 + Conference Number of XX = Conference ID of 91939271XX

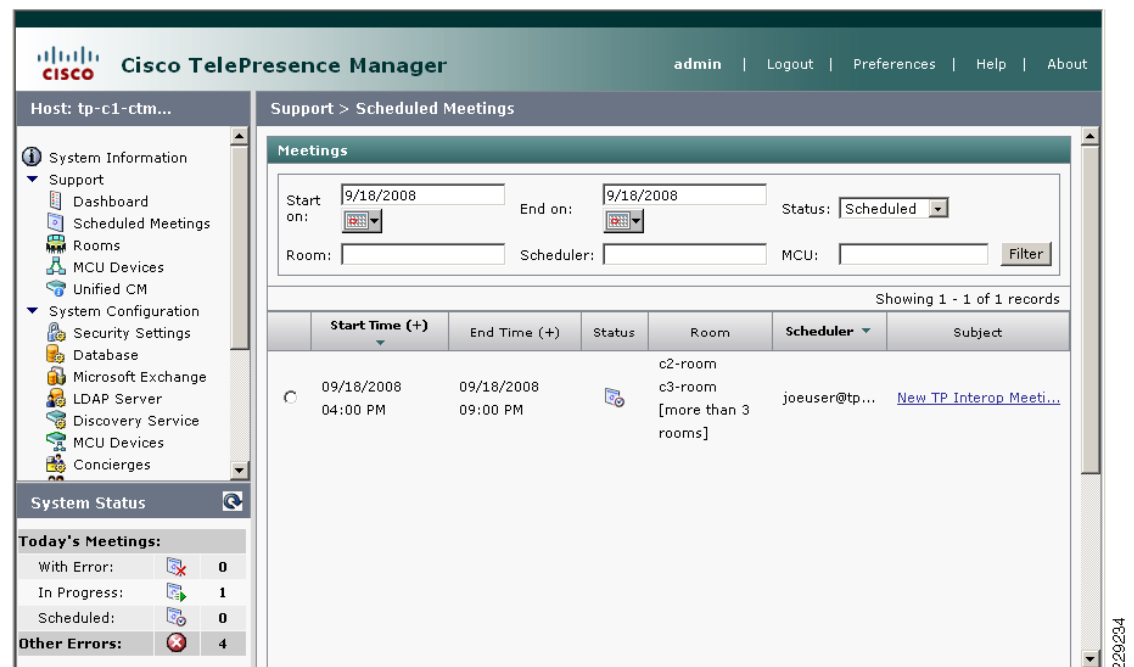
In other words, for this configuration, the Cisco TelePresence Manager dynamically generates conference IDs in the range of 9193927100 to 9193927199 for the interoperability segment as meetings are scheduled. As with static meetings, the CUVC MCU is again configured with a service with the same prefix. This allows the creation of ad-hoc interoperability conferences on the CUVC MCU when scheduled TelePresence calls are initiated. Figure 14 shows an example of the services configured on a CUVC 3500 Series MCU running software version 5.5.

Figure 14 Sample CUVC 3500 Series Services Configuration



When scheduling a TelePresence meeting through a groupware client such as Microsoft Outlook or IBM Notes, the Cisco TelePresence Manager is aware of the TelePresence resources required because it communicates with groupware systems, such as Microsoft Exchange or IBM Domino, to extract meeting scheduling information for these resources. However, the end user cannot specify the need for interoperability within the meeting request itself. The Cisco TelePresence Manager allows a meeting scheduler or administrator to go into a scheduled meeting and add interoperability. Figure 15 shows an example of scheduled meetings, as they appear on Cisco TelePresence Manager.

Figure 15 Sample Cisco TelePresence Manager Scheduled Meetings Screen



By selecting a particular meeting and clicking **Details**, the Meeting Summary screen is displayed. With this screen, the meeting scheduler or administrator can add interoperability by selecting the Video Conferencing tab to display a screen similar to that shown in Figure 16.

Figure 16 **Sample Video Conferencing Tab with a Scheduled Meeting**

Meeting Summary

Meeting Manager helps automate Cisco TelePresence calls by reviewing your meeting request and presetting the appropriate equipment to launch your Cisco TelePresence experience with "One-Button-To-Push". This webpage allows you to specify options not available via your calendar application and allows you to correct some errors.

Subject: New TP Interop Meeting
Scheduler: Joe M. User (joeuser@tp.com)

Meeting Details: 09/18/2008 04:00 PM

Meeting Details **Video Conferencing**

WARNING: This meeting cannot be modified as its scheduled start time has elapsed.

Enable Interoperability with Video Conference : ☒ Yes ☐ No

Number of Video Conference Participants :

Video Conference Access Number : 9193927172

* Required Fields

22:02:35

The meeting scheduler or network administrator can specify the number of video conferencing endpoints allowed within the particular TelePresence meeting. Cisco TelePresence Manager automatically generates the Video Conference Access Number (that is, conference ID) based on the information provided when the CUVC was added, as shown in [Figure 16](#). In the example above, the conference access number was determined based on the following:

Prefix of 91939271 + Conference Number of 72 = Conference Access Number (Conference ID) of 9193927172

As with static meetings, when the first CTS endpoint uses the one-button-to-push feature to start the virtual meeting, the CTMS dials out to the CUVC using the conference access number. The CUVC dynamically creates an ad hoc conference based on service prefix of *91939271* and conference number 72. An example of this is shown in the [Figure 17](#) for a CUVC 3500 Series MCU running software version 5.5.

Figure 17 Sample CUVVC 3500 Series Interoperability Ad Hoc Conference

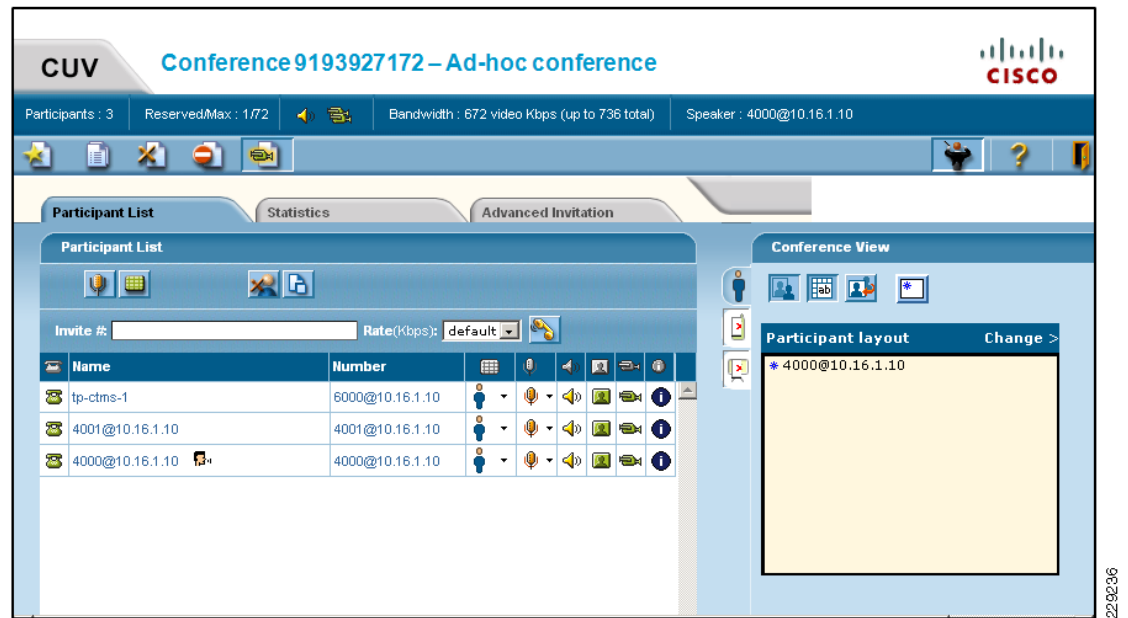
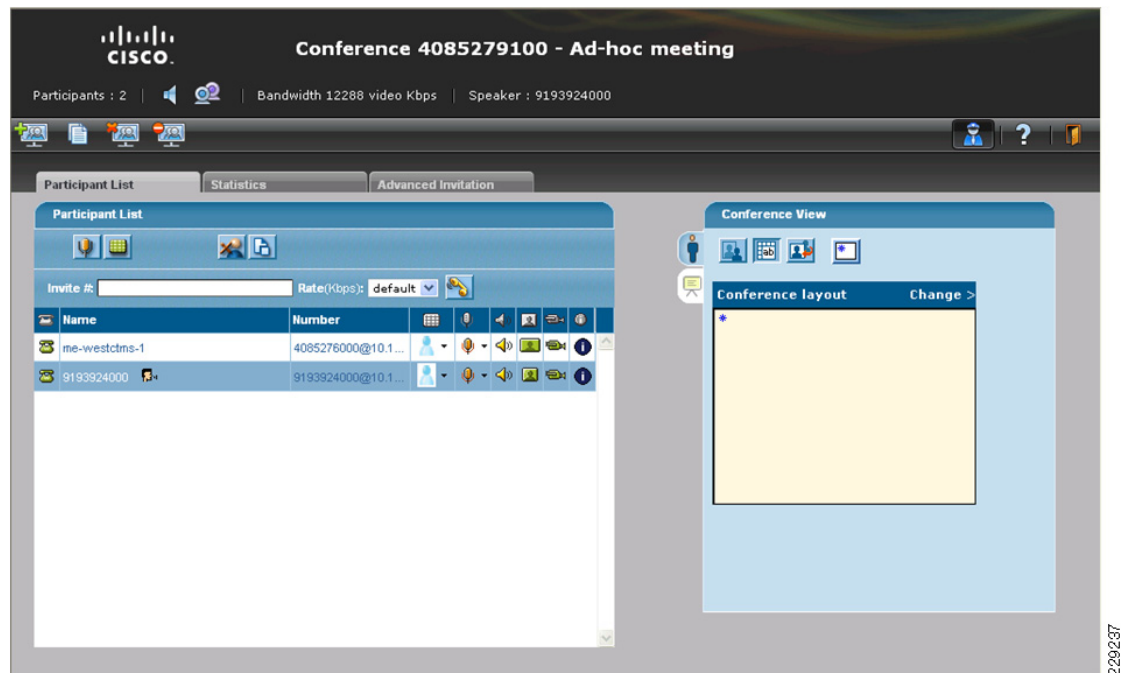


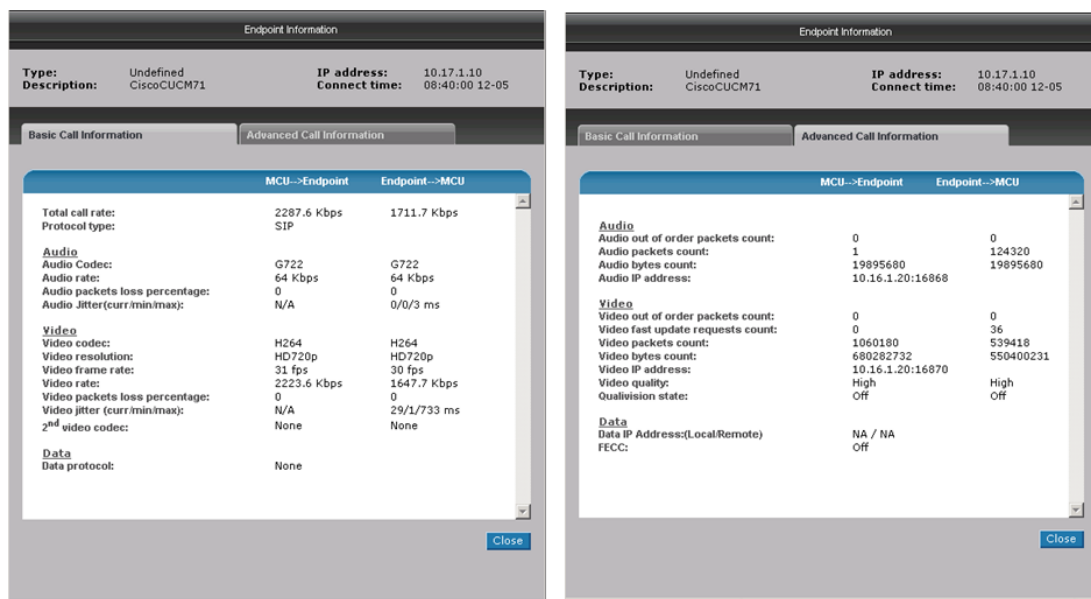
Figure 18 shows the same screen for a CUVVC 5100/5200 Series MCU running software version 7.0

Figure 18 Sample CUVVC 5100/5200 Series Interoperability Ad Hoc Meeting



Note that because only the CTMS establishes the cascade segment between itself and the CUVVC MCU, it is the only TelePresence device that appears to the CUVVC MCU in the meeting. Statistical information for each endpoint can be viewed by clicking on the information button (the blue button with the white “i” shown in Figure 18) next to the particular endpoint. Figure 19 shows an example of displaying the CTMS endpoint, as seen on the CUVVC 5100/5200 Series MCU.

Figure 19 Sample CUVVC 5100/5200 Series Endpoint Statistics



These statistics can be useful in assisting the network design engineer in troubleshooting any quality issues between the CTMS and the CUVVC MCU.

Finally, note that although Cisco TelePresence endpoints can automatically join the TelePresence meeting through the one-button-to-push feature, traditional video conferencing endpoints and desktop video collaboration endpoints do not support this capability. Cisco TelePresence Manager automatically e-mails a confirmation of the scheduled meeting back to the meeting scheduler. This confirmation includes the conference access number to be used by the traditional video conferencing endpoints and desktop video collaboration endpoints to access the interoperability meeting. Also, network administrators should note that TelePresence ad-hoc meetings, which by definition are meetings started by a Cisco TelePresence CTMS meeting scheduler or administrator, do not support interoperability as of CTS version 1.6.

Network Design Implications

The network design implications of adding interoperability to the TelePresence solution include bandwidth considerations, additional latency considerations, burst considerations, as well as QoS considerations of integrating both legacy video conferencing and Cisco TelePresence over a single converged network infrastructure. These considerations vary depending on whether SD interoperability or HD interoperability has been provisioned.

Additional Bandwidth Considerations

For SD interoperability meetings, each CTS endpoint generates an additional audio and video stream along with the normal video and audio streams generated for non-interoperability meetings. This results in higher bandwidth utilization. For HD interoperability meetings, each CTS endpoint generates an additional audio stream along with the normal audio stream generated for non-interoperability meetings. However, no additional video stream is generated along with the normal video stream. Further, the TelePresence video is constrained to 720p resolution. This can result in lower overall bandwidth utilization, if non-interoperability meetings are normally held at 1080p resolution. The following sections discuss this in more detail.

Standard Definition Interoperability

The effect of an additional G.711 audio stream and H.264 CIF video stream on bandwidth utilization is approximately 768 Kbps (704 Kbps for the video stream and 64 kbps for the audio stream), excluding network protocol overhead that can average about 20 percent. Therefore, to support SD interoperability, the network design engineer should plan for approximately $768 \text{ Kbps} \times 1.2 = 922 \text{ Kbps}$ of additional bandwidth to be provisioned for each CTS site.

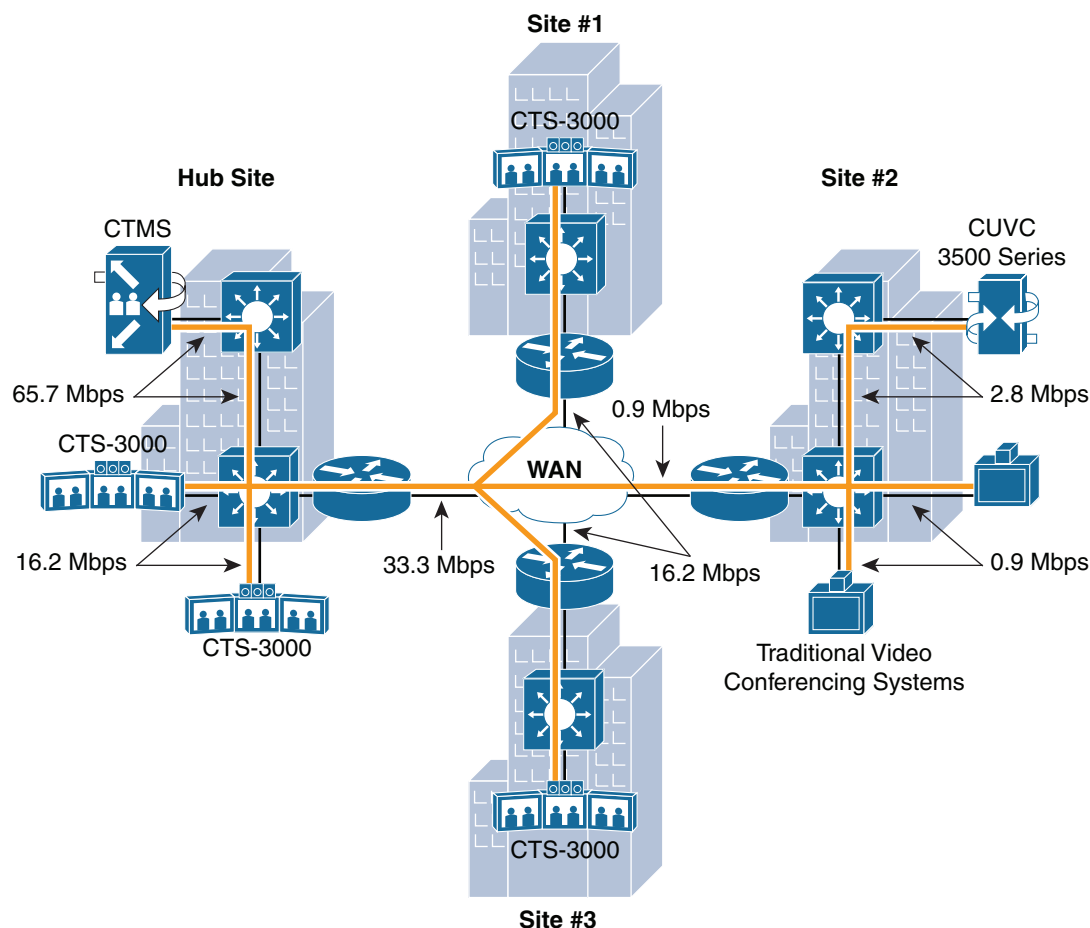
Table 1 shows the adjusted bandwidth requirements for Cisco TelePresence devices when interoperability and low-speed auxiliary video input for the Auto Collaborate feature (CTS endpoints only) are supported.

Table 1 Cisco TelePresence Bandwidth Requirements Including SD Interoperability

Resolution	1080p	1080p	1080p	720p	720p	720p
Motion Handling	Best	Better	Good	Best	Better	Good
Video per Screen (kbps)	4000	3500	3000	2250	1500	1000
Audio per Microphone (kbps)	64	64	64	64	64	64
(5 fps) Auto Collaborate Video Channel (kbps)	500	500	500	500	500	500
Auto Collaborate Audio Channel (kbps)	64	64	64	64	64	64
Interoperability Video Channel (kbps)	704	704	704	704	704	704
Interoperability Audio Channel (kbps)	64	64	64	64	64	64
CTS-500/1000/1100/1300 Total Audio and Video (kbps)	5,524	5,024	4,524	3,774	3,024	2,524
CTS-3000/3010/3200/3210 Total Audio and Video (kbps)	13,524	12,024	10,524	8,274	6,024	4,524
CTS-500/1000/1100/1300 total bandwidth (Including Layer 2-Layer 4 overhead)	6.6 Mbps	6.0 Mbps	5.4 Mbps	4.5 Mbps	3.6 Mbps	3.0 Mbps
CTS-3000/3010/3200/3210 total bandwidth (Including Layer 2-Layer 4 overhead)	16.2 Mbps	14.4 Mbps	12.6 Mbps	9.9 Mbps	7.2 Mbps	5.4 Mbps

The site or sites that contain the CTMS and the CUVC must also have sufficient bandwidth to accommodate the calls. Figure 20 shows a sample network where the CTMS and CUVC are located at separate physical sites. In this example, the CTS and traditional video conferencing endpoints are in a single virtual meeting with interoperability enabled. CTS endpoints are running at 1080p video resolution in the example.

Figure 20 *SD Interoperability Bandwidth Requirements with CTMS and CUVC Located at Separate Sites*



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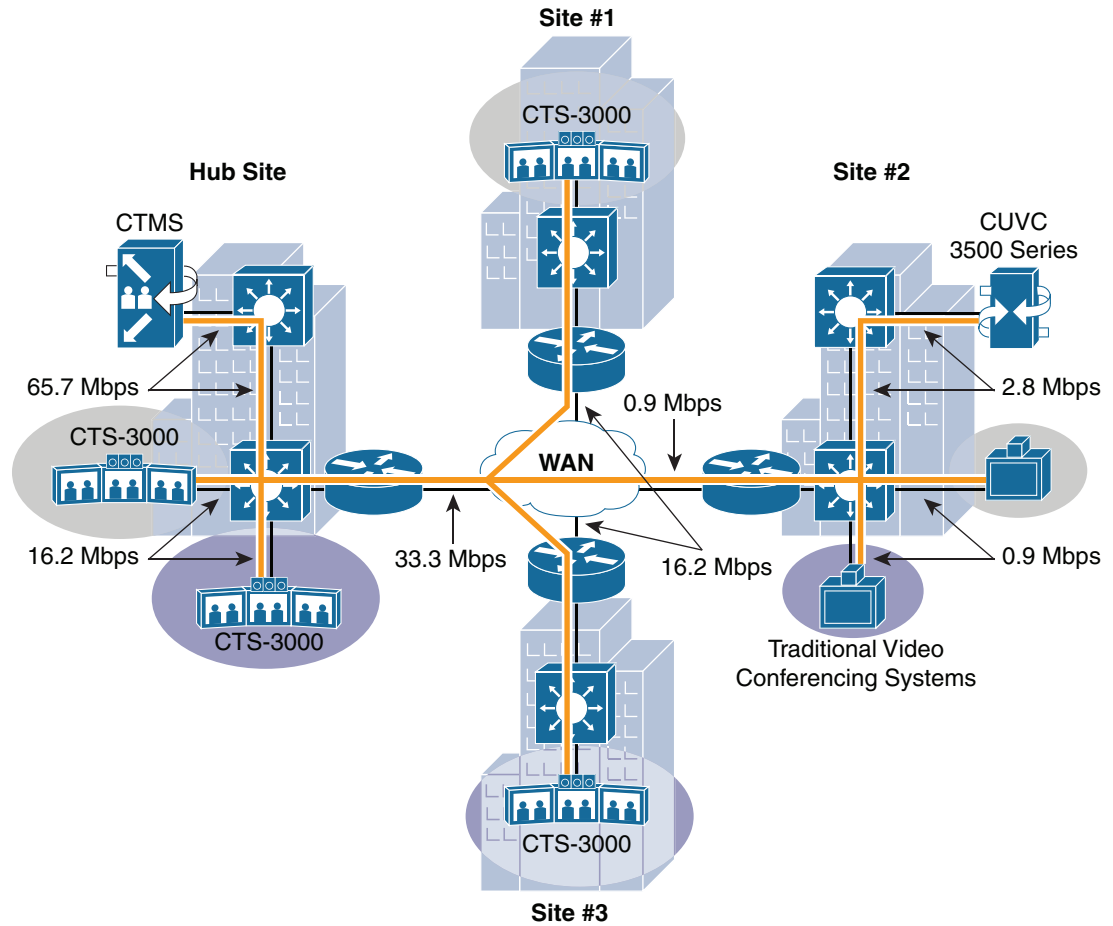
In the example above, the Hub Site contains the CTMS while Site #2 contains the CUVC MCU. The total amount of bandwidth required to support TelePresence with SD interoperability at the Hub Site WAN connection can be calculated by the following:

$16.2 \text{ Mbps per CTS-3000} \times 2 \text{ Telepresence Sites} + 0.9 \text{ Mbps for the cascade segment between the CTMS and CUVC} = \text{approximately } 33.3 \text{ Mbps}$

Note in [Figure 20](#) that the cascade segment between the CTMS and CUVC MCU uses H.264 CIF video and G.711 audio at a combined rate of approximately 0.9 Mbps, including network overhead. However, the CUVC can transcode the audio and video to whatever rate is supported by both the CUVC and traditional video conferencing endpoints before being sent. In the network design of [Figure 20](#), this would not affect the WAN bandwidth requirements of the Hub Site or Site #2.

The design engineer must also be aware that multiple TelePresence multipoint meetings can each support interoperability. [Figure 21](#) shows the same network configuration and endpoints. However, this time two separate virtual meetings are ongoing, each with SD interoperability enabled.

Figure 21 SD Interoperability Bandwidth Requirements with Multiple Calls



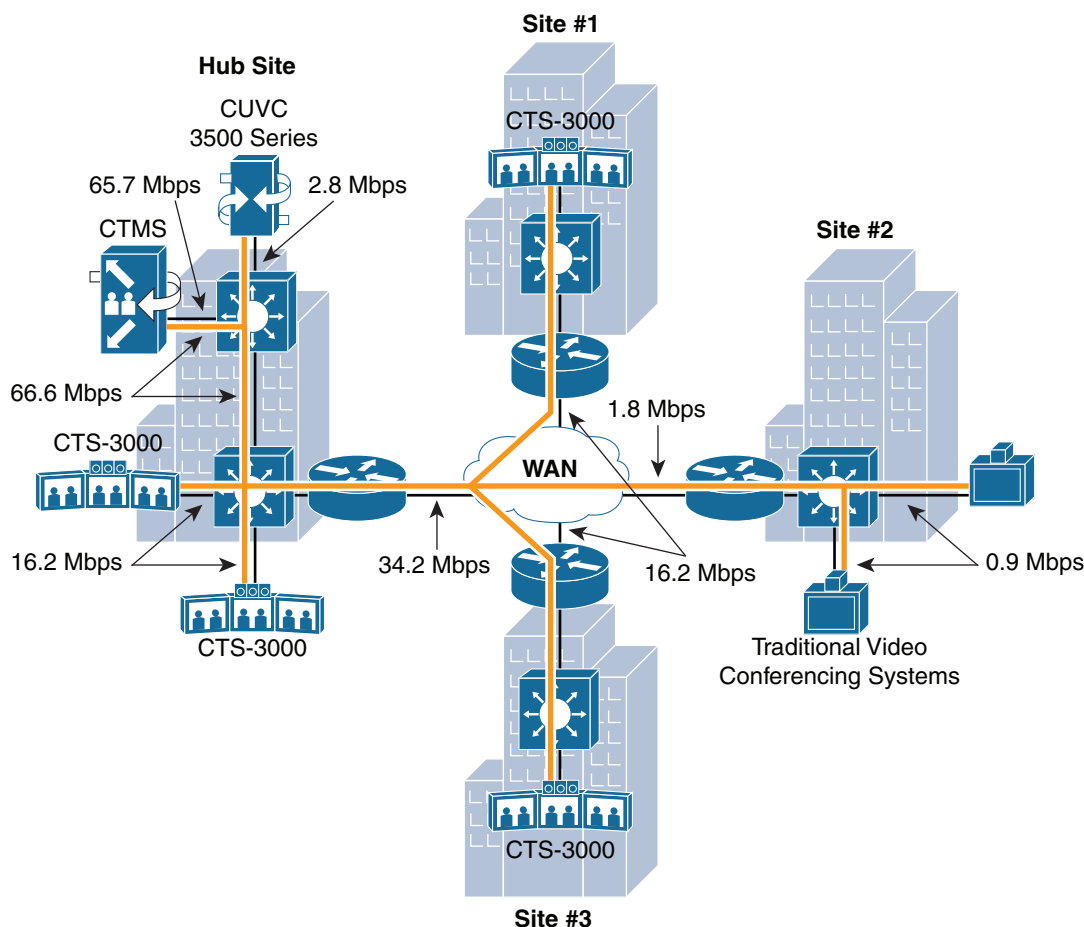
As can be seen, total amount of bandwidth required to support TelePresence with SD interoperability at the Hub Site is calculated by the following:

$16.2 \text{ Mbps per CTS-3000} \times 2 \text{ Telepresence Sites} + 0.9 \text{ Mbps} \times 2 \text{ Meetings for the cascade segment between the CTMS and CUVC} = \text{approximately } 34.2 \text{ Mbps}$

For this type of network configuration, as more meetings with SD interoperability are simultaneously ongoing, the amount of bandwidth required across the network from the Hub Site to Site #2 increases.

Figure 22 shows an alternate design. The network layout is the same, but the CTMS and CUVC are co-located at the Hub Site.

Figure 22 SD Interoperability Bandwidth Requirements with CTMS and CUVA Co-Located



For the example above, the traditional video conferencing systems support a combined audio and video rate of 0.9 Mbps each (768 Kbps CIF rate with 20 percent network overhead). However, CUVC can transcode the received media from the CTMS to a new rate before sending it to the traditional video conferencing endpoints. As a result, the amount of bandwidth required to support TelePresence with SD interoperability when both the CTMS and CUVC MCU are co-located at the Hub Site can be higher or lower, depending on what rates the traditional video conferencing endpoints support.

High Definition Interoperability

Because the video resolution of TelePresence endpoints is constrained to 720p, the overall bandwidth utilization may be lower per HD interoperability meeting, depending on whether normal non-interoperability meetings are configured to use 1080p or 720p resolution. The cascade segment between the CTMS and CUVC consists of an H.264 720p video stream that can run at 2.25 Mbps, 1.5 Mbps, or 1 Mbps, depending on whether Best, Better, or Good motion handling has been selected in the configuration of the CTMS. The cascade segment also includes a 64 Kbps G.722 audio stream. Both the video and audio rates above are excluding network protocol overhead that can average approximately 20 percent. Therefore, the network design engineer should plan for up to approximately $2.3 \times 1.2 = 2.8$ Mbps of bandwidth to be provisioned for the cascade segment between the CTMS and the CUVC per call that requires HD interoperability.

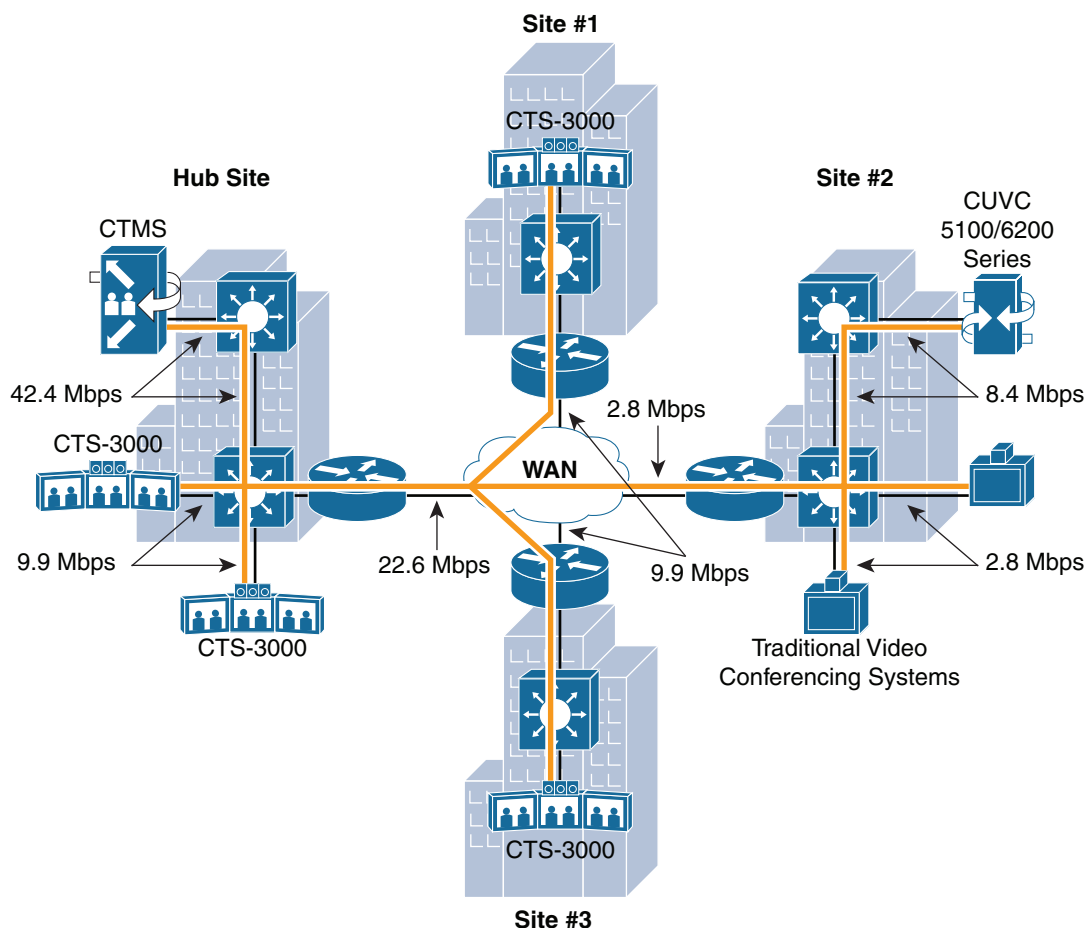
Table 2 shows the bandwidth requirements for each CTS endpoint within an HD interoperability call.

Table 2 *Cisco TelePresence Bandwidth Requirements Including High Definition Interoperability*

Resolution	720p	720p	720p
Motion Handling	Best	Better	Good
Video per Screen (kbps)	2250	1500	1000
Audio per Microphone (kbps)	64	64	64
(5 fps) Auto Collaborate Video Channel (kbps)	500	500	500
Auto Collaborate Audio Channel (kbps)	64	64	64
Interoperability Video Channel (kbps)	704	704	704
Interoperability Audio Channel (kbps)	64	64	64
CTS-500/1000/1100/1300 Total Audio and Video (kbps)	3,774	3,024	2,524
CTS-3000/3010/3200/3210 Total Audio and Video (kbps)	8,274	6,024	4,524
CTS-500/1000/1100/1300 total bandwidth (Including Layer 2-Layer 4 overhead)	4.5 Mbps	3.6 Mbps	3.0 Mbps
CTS-3000/3010/3200/3210 total bandwidth (Including Layer 2-Layer 4 overhead)	9.9 Mbps	7.2 Mbps	5.4 Mbps

The site or sites that contain the CTMS and the CUVC must again have sufficient bandwidth to accommodate the calls. [Figure 23](#) shows a sample network where the CTMS and CUVC are located at separate physical sites. In this example, the CTS and traditional video conferencing endpoints are in a single virtual meeting with HD interoperability enabled. The video resolution selected is 720p, Best motion handling.

Figure 23 *HD Interoperability Bandwidth Requirements with CTMS and CUVC Located at Separate Sites*



In the example above, the Hub Site contains the CTMS while Site #2 contains the CUVC MCU. The total amount of bandwidth required to support TelePresence with HD interoperability at the Hub Site WAN connection can be calculated by the following:

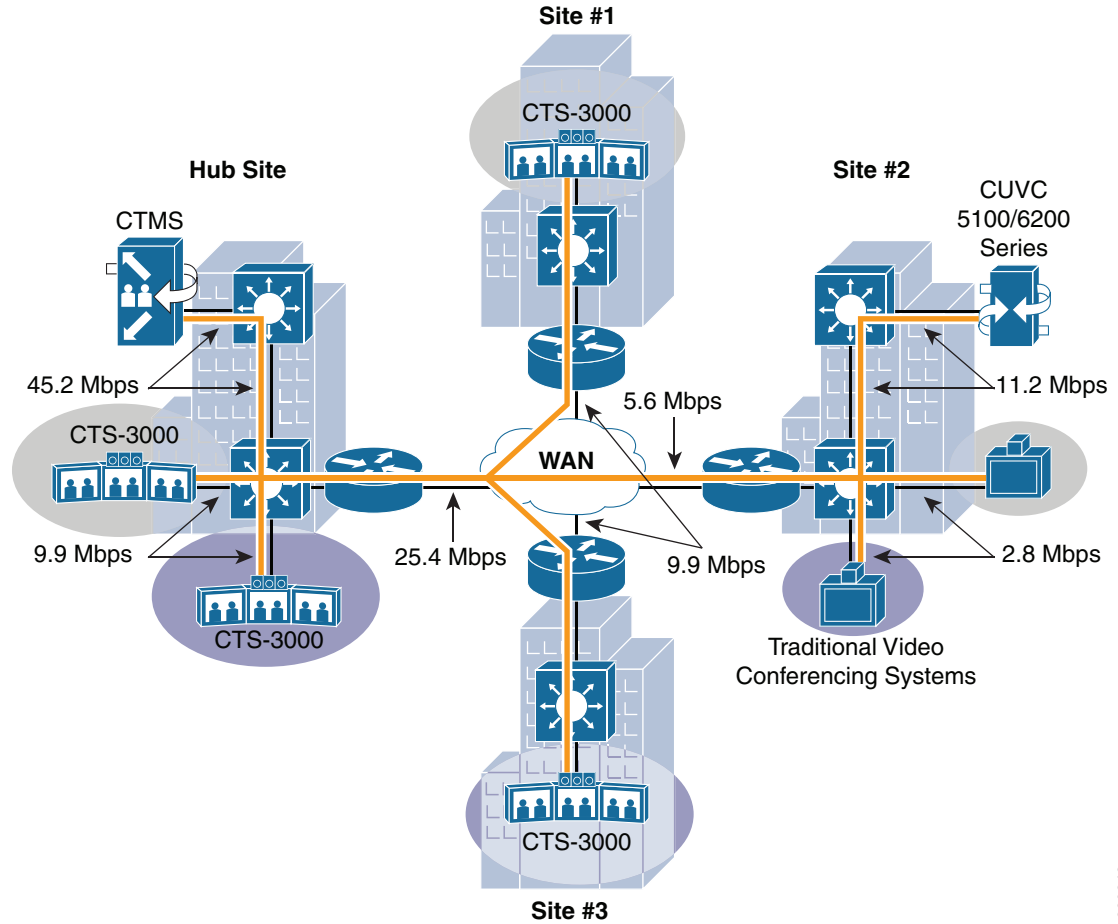
9.9 Mbps per CTS-3000 * 2 Telepresence Sites + 2.8 Mbps for the cascade segment between the CTMS and CUVC = approximately 22.6 Mbps

The overall amount of video traffic in this example is lower than the example for SD interoperability shown in [Figure 20](#). This is because all CTS endpoints within an HD interoperability meeting are also limited to 720p resolution, along with the cascade segment between the CTMS and CUVA.

Note in [Figure 23](#) that the cascade segment between the CTMS and CUVC MCU uses H.264 720p video and G.722 audio at a combined rate of approximately 2.8 Mbps, including network overhead. However, the CUVC can transcode the audio and video to whatever rate is supported by both the CUVC and traditional video conferencing endpoints before being sent. In the network design of [Figure 23](#), this would not affect the WAN bandwidth requirements of the Hub Site or Site #2.

As with SD interoperability, the design engineer must also be aware that multiple TelePresence multipoint meetings can each support HD interoperability. [Figure 24](#) shows the same network configuration and endpoints. However, this time two separate virtual meetings are ongoing, each with HD interoperability enabled. Again, 720p resolution with Best motion handling has been selected for the example.

Figure 24 HD Interoperability Bandwidth Requirements with Multiple Calls



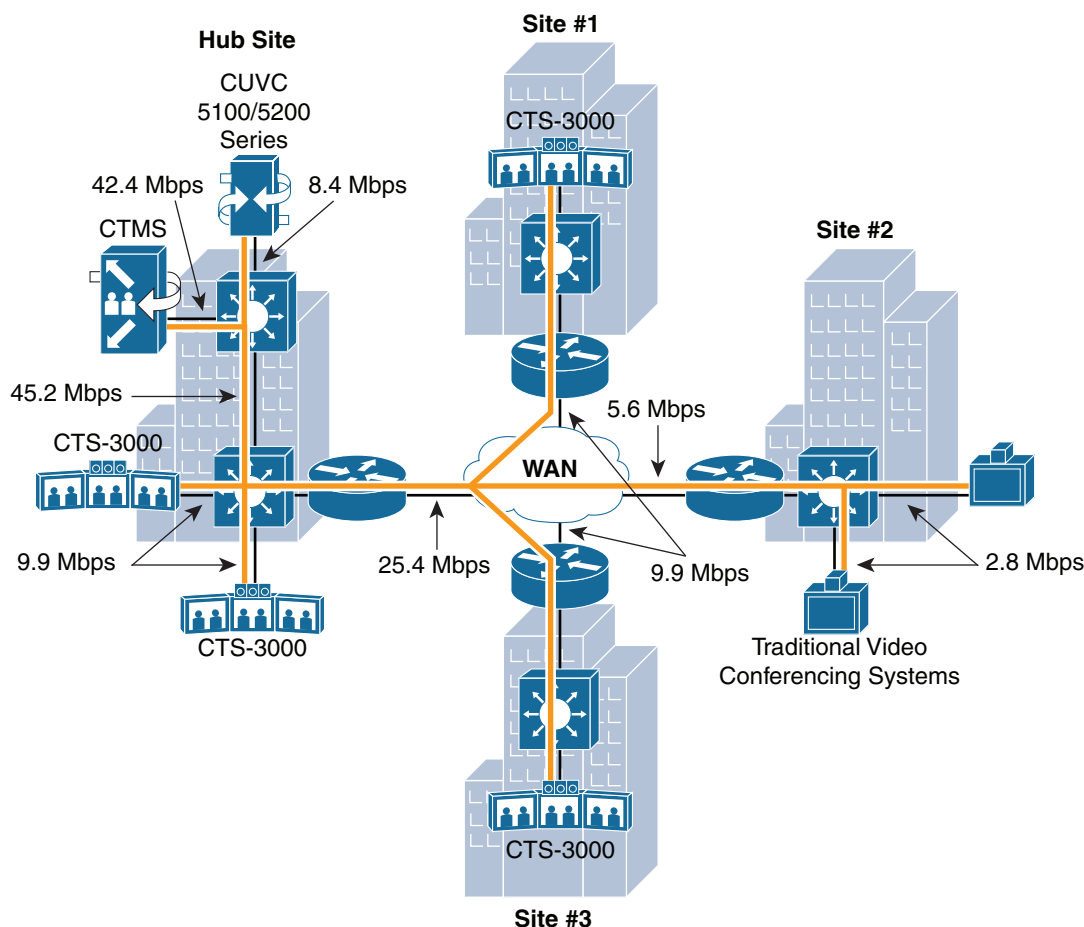
As can be seen, the total amount of bandwidth required to support TelePresence with HD interoperability at the Hub Site is calculated by the following:

$9.9 \text{ Mbps per CTS-3000} \times 2 \text{ Telepresence Sites} + 2.8 \text{ Mbps} \times 2 \text{ Meetings for the cascade segment between the CTMS and CUVC} = \text{approximately } 25.4 \text{ Mbps}$

For this type of network configuration, as more meetings with HD interoperability are simultaneously ongoing, the amount of bandwidth required across the network from the Hub Site to Site #2 increases.

Figure 25 shows an alternate design. The network layout is the same, but the CTMS and CUVC are co-located at the Hub Site. Again, 720p resolution with Best motion handling has been selected for the example.

Figure 25 *HD Interoperability Bandwidth Requirements with CTMS and CUVA Co-Located*



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For the example above, the traditional video conferencing systems support a combined audio and video rate of 2.8 Mbps each (2.25 Mbps 720p H.264 video + 64 Kbps G.722 audio with 20 percent network overhead). However, CUVC can transcode the received media from the CTMS to a new rate before sending it to the traditional video conferencing endpoints. As a result, the amount of bandwidth required to support TelePresence with HD interoperability when both the CTMS and CUVC MCU are co-located at the Hub Site can again be higher or lower, depending on what rates the traditional video conferencing endpoints support.

Both the SD and HD interoperability examples shown in [Figure 20](#) through [Figure 25](#) demonstrate that the network design engineer must be aware of the following when adding support for TelePresence interoperability to traditional video conferencing systems:

- The location and bandwidth requirements of the TelePresence endpoints
- The location and bandwidth requirements of the non-TelePresence endpoints
- The bandwidth requirements of the cascade segment between the CTMS and CUVC
- The location of the CTMS and CUVC with respect to each other and the network infrastructure
- The number of interoperability meetings that may be ongoing simultaneously

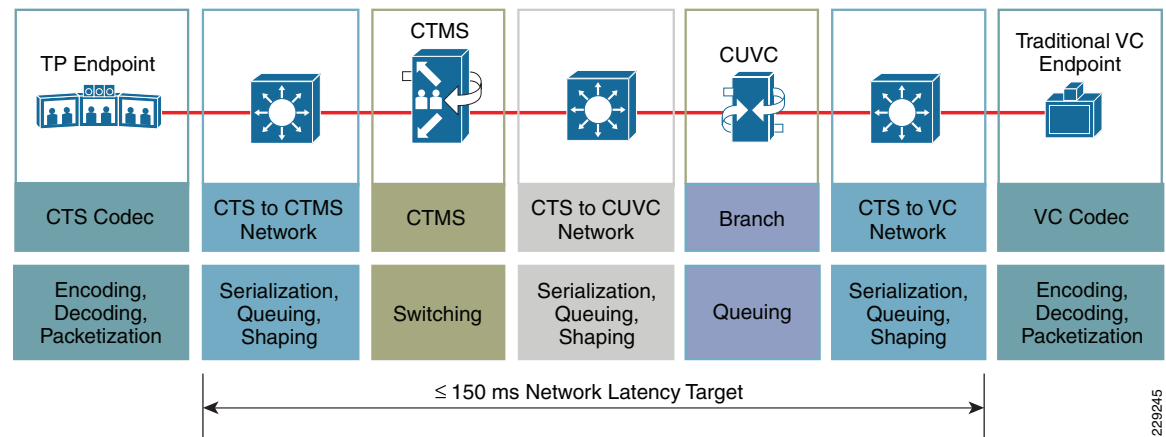
The network design engineer should also note that traditional videoconferencing endpoints may also mark their media differently from TelePresence endpoints, resulting in the media being mapped to a different traffic class on networking equipment. Therefore, the design engineer must allocate sufficient

bandwidth within the appropriate traffic classes to support both the TelePresence endpoints and the non-TelePresence endpoints. [Interoperability QoS Considerations, page 30](#) discusses methods of integrating both TelePresence and legacy video conferencing systems across a common IP infrastructure, while allocating separate bandwidth for each.

Interoperability Latency Considerations

The introduction of interoperability may increase the overall latency of TelePresence calls. [Figure 26](#) shows the latency induced from each segment of a TelePresence call that includes interoperability.

Figure 26 Latency Components of a TelePresence Meeting that Includes Interoperability



The end-to-end network latency target for TelePresence calls is less than or equal to 150 msec, excluding processing time within the TelePresence codecs. Assuming approximately equivalent processing time for encoding, decoding, and packetization between traditional video conferencing and/or desktop video collaboration endpoints, the 150 msec target for network flight time can be extended to TelePresence meetings that include interoperability as well.



Note

If the traditional video conferencing and/or desktop video collaboration endpoint has significantly higher encoding, decoding, and packetization processing times, the target end-to-end network latency may need to be decreased.

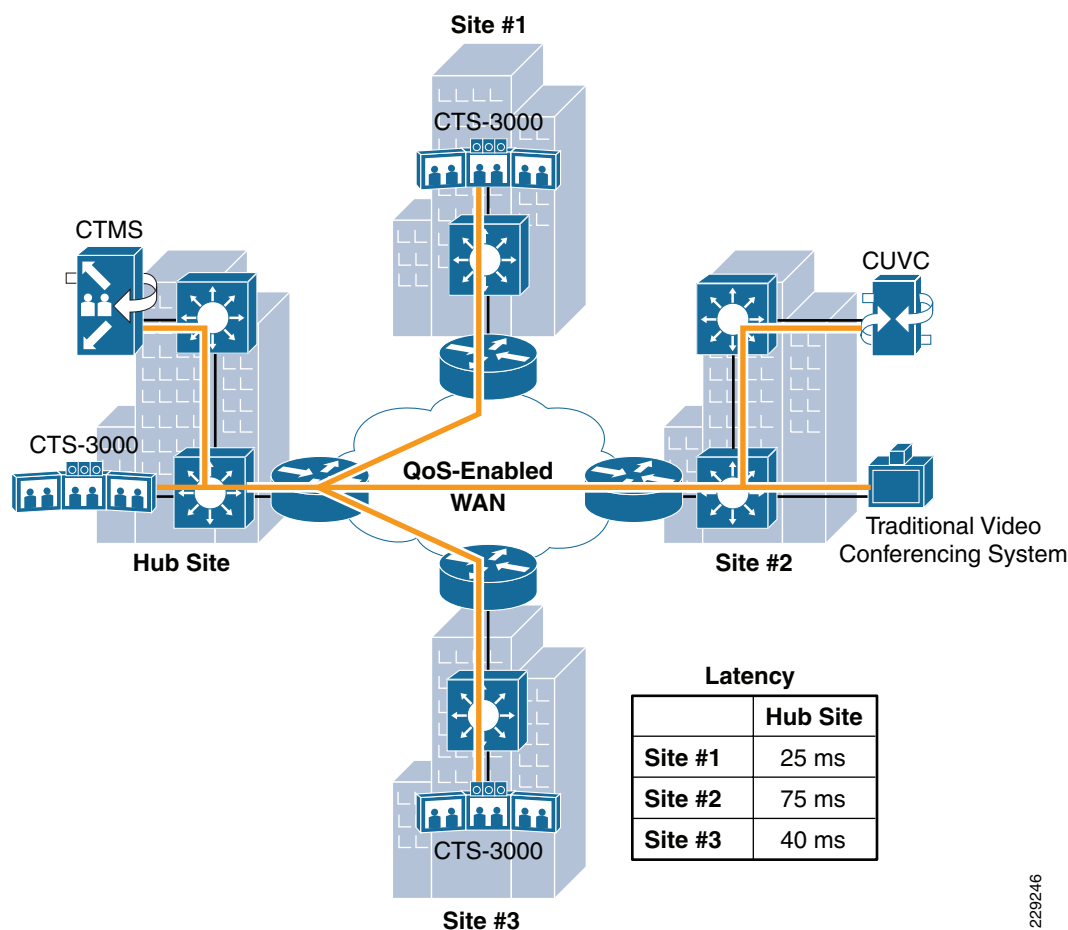
The end-to-end network latency includes the following:

- The serialization, queuing, and shaping delay incurred from the TelePresence endpoint to the CTMS. This delay is variable and primarily based on circuit distance between the CTS endpoint and the CTMS, as well as any possible shaping delays. Queueing delays are generally smaller compared to delays because of circuit distance and shaping.
- The switching delay within the CTMS. This is typically under 10 msec.
- The serialization, queuing, and shaping delay incurred from the CTMS to the CUVC. This delay is again variable and primarily based on circuit distance between the CTMS and the CUVC, as well as any possible shaping delays. Queueing delays are generally smaller compared to delays because of circuit distance and shaping.

- The switching and transcoding delays within the CUVC. This delay is variable, based on whether the CUVC is simply switching audio and video to the traditional room-based video conferencing or desktop video collaboration endpoints, or is performing transcoding of the audio and video.
- The serialization, queueing, and shaping delay incurred from the room-based video conferencing or desktop video collaboration endpoint to the CUVC. This delay is variable and primarily based on circuit distance between the endpoint and the CUVC, as well as any possible shaping delays. Again, queueing delays are generally smaller compared to delays because of circuit distance and shaping.

Figure 27 shows an example of how to calculate the end-to-end network latency of a TelePresence meeting with interoperability enabled. The latency numbers used are examples only.

Figure 27 Interoperability Network Latency Example



The end-to-end latency can be calculated by summing the latency for each segment of the TelePresence call that includes interoperability. The latency of the LAN within the campus is considered negligible for this example.

Site #3 CTS-3000 to CTMS Serialization Delay:	40 msec
CTMS Switching Delay:	10 msec
CTMS to CUVC Serialization Delay:	60 msec
CUVC Switching/Transcoding Delay:	40 msec
CUVC to Traditional Endpoint Serialization Delay:	0 msec
Total One Way End-to-End Latency	150 msec

In the example above, the total latency just meets the target of 150 msec. The design engineer should note that because of the additional serialization latency of the active cascade segment between the CTMS and the CUVC, as well as the additional switching and potentially transcoding latency of the CUVC, it may be difficult to meet the target of 150 msec one-way end-to-end delay with interoperability. Exceeding the 150 msec one-way end-to-end delay recommendation does not immediately cause the TelePresence interoperability call to stop working. Experience has shown that as one-way end-to-end latency increases, normal human speech patterns begin to degrade and become awkward. The design engineer should attempt to minimize end-to-end delay to maximize the quality of the TelePresence call with interoperability.

For example, in [Figure 27](#) it can be seen that the end-to-end latency benefits from having the CUVC co-located with the traditional video conferencing endpoint. This design results in serialization delay between the CTMS and CUVC, but minimizes the serialization delay between the CUVC and the traditional video conferencing endpoint. Alternatively, the CUVC can be co-located with the CTMS. This minimizes the serialization delay between the CTMS and the CUVC, but introduces a serialization delay component between the CUVC and the traditional video conferencing endpoint. Either design results in the same end-to-end network delay for the TelePresence meeting that includes interoperability. Finally, the design engineer should note that if an additional traditional video conferencing endpoint is located at the Hub Site, an additional 60 msec of serialization latency needs to be added. In such a situation, the one-way end-to-end latency would exceed the target of 150 msec.

Interoperability Burst Considerations

CTS version 1.6 introduced two new features designed to reduce the burstiness of Cisco TelePresence video. Burstiness was primarily the result of the codec needing to transmit a new reference frame, referred to as an Instantaneous Decoder Refresh (IDR) in H.264 terminology. TelePresence IDRs have been observed to be as large as 64–65 Kbytes in size per camera, sent in one video frame interval (33 msec). Previous to CTS version 1.5, IDRs were generated by the active speaker (transmitter of a video stream) when any endpoint in a multipoint call experienced packet loss and requested a new reference point. IDRs were also generated when the active speaker transitioned, and a new endpoint began to transmit a video stream.

CTS 1.6 removed the requirement for IDRs during recovery from packet loss by introducing Long Term Reference Pictures (LRTPs). LRTPs are basically reference points that are held for longer periods of time within the receiving codec. Whenever any endpoint within a CTS 1.6 multipoint call experiences packet loss, the transmitting codec can simply instruct all receiving endpoints to use the LRTP as the new reference point. Subsequent video packets include encoded information relative to the LRTP. By using LRTPs, the transmitting codec no longer needs to transmit an IDR when packet loss is seen by any of the endpoints during a multipoint call. This reduces overall burstiness per frame interval.

CTS 1.6 removed the requirement for IDRs during speaker transitions by introducing Gradual Decoder Refreshes (GDRs). With GDRs, the transition from one speaker to another speaker occurs over multiple video frames. The existing video is frozen on the display until the entire reference picture from the new transmitter is received. The display then transitions to the new video source. Overall, GDRs are typically larger (from approximately 90 Kbytes to 110 Kbytes per camera have been observed in lab testing) than IDRs. However, they are spread over multiple video frames. This again reduces the overall burstiness per frame interval.

The design engineer should note that TelePresence calls that include interoperability do not negotiate the use of LRTPs or GDRs. Therefore the burstiness of CTS 1.6 multipoint calls that include interoperability is much more similar to what was seen in CTS software version 1.5 and below.

Interoperability QoS Considerations

To implement a medianet, a clear understanding of the QoS model deployed within the enterprise is necessary. Furthermore, the model should specify the classification and Differentiated Services Code Point (DSCP) marking of each type of video traffic. Cisco currently recommends the RFC 4594-based QoS model shown in [Figure 28](#) for deployment within enterprise networks.

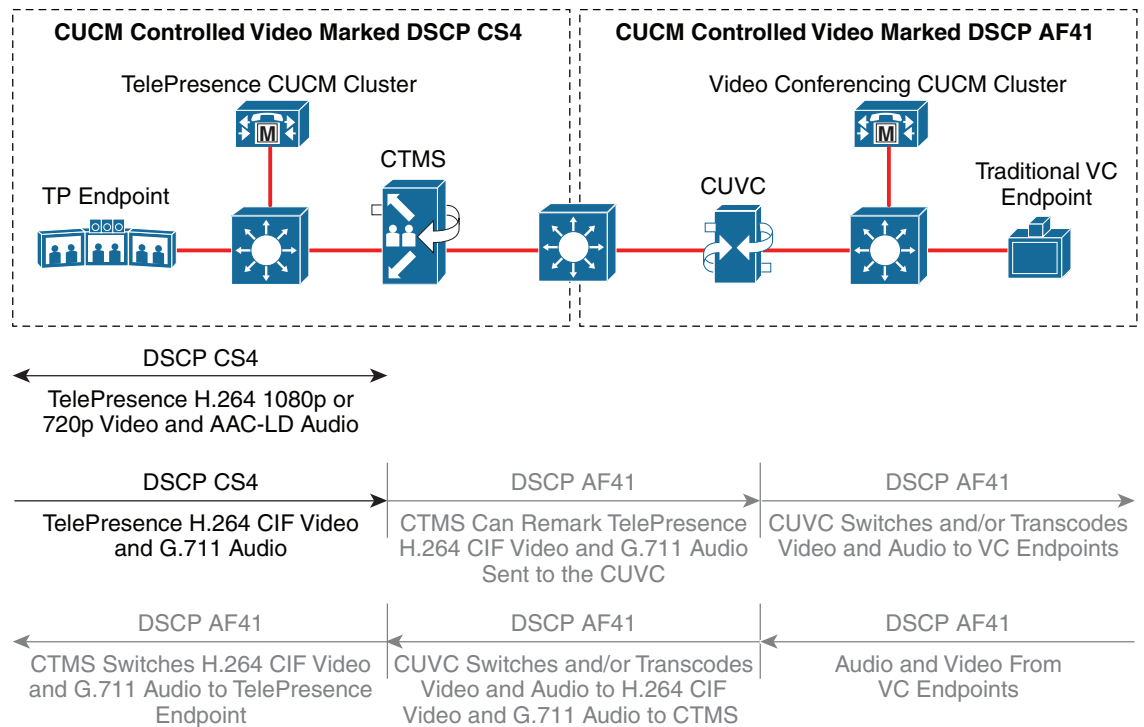
Figure 28 *RFC-4594 Based Enterprise QoS Model*

Application Class	PHB	Admission Control	Queueing and Dropping	Application Examples
VoIP Telephony	EF	Required	Priority Queue (PQ)	Cisco IP Phones
Broadcast Video	CS5	Required	Optional (PQ)	Cisco IP Surveillance, Cisco Enterprise TV
Realtime Interactive	CS4	Required	Optional (PQ)	Cisco TelePresence
Multimedia Conferencing	AF4	Required	BW Queue + DSCP WRED	Cisco Unified Personal Communicator
Multimedia Streaming	AF3	Recommended	BW Queue + DSCP WRED	Cisco Digital Media System (VoD)
Network Control	CS6		BW Queue	EIGRP, OSPF, BGP, HSRP, IKE
Call-Signaling	CS3		BW Queue	SCCP, SIP, H.323
OAM	CS2		BW Queue	SNMP, SSH, Syslog
Transactional Data	AF2		BW Queue + DSCP WRED	Cisco WebEx, Cisco MeetingPlace, ERP Apps
Bulk Data	AF1		BW Queue + DSCP WRED	Email, FTP, Backup Apps, Content Distribution
Best Effort	default		Default Queue + RED	Default Class Traffic
Scavenger	CS1		Min BW Queue (Deferential)	YouTube, iTunes, BitTorrent, Xbox Live

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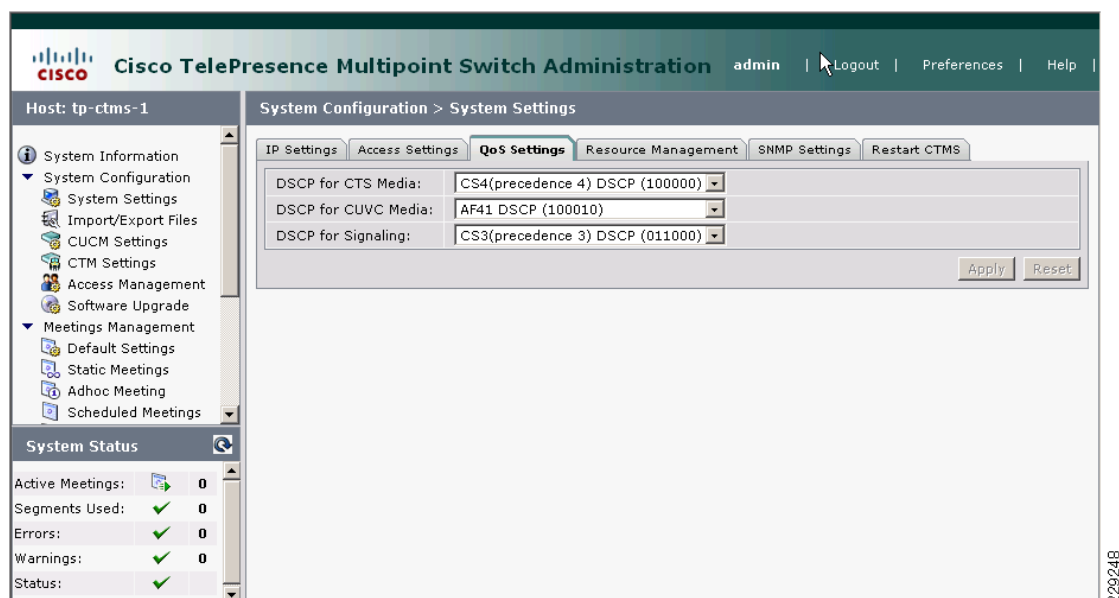
Within this model, Cisco TelePresence traffic is given a DSCP marking of CS4 and placed into the Real-Time Interactive traffic class. Traditional room-based video conferencing and desktop video collaboration is given a DSCP marking of AF41, and placed into the Multimedia Conferencing traffic class. Alternatively, other non-TelePresence high-end video conferencing systems may also be marked CS4 along with TelePresence, and placed in the Real-Time Interactive traffic class. Because CUCM version 8.0 and below has a single system-wide configuration for the DSCP marking of all video traffic, Cisco has recommended the deployment of a separate CUCM cluster for TelePresence. Otherwise, desktop video collaboration systems and traditional video conferencing systems controlled by CUCM would not be able to mark their video traffic differently from TelePresence.

[Figure 29](#) shows an example of the marking of video and audio in a TelePresence meeting with SD interoperability enabled, assuming the customer has chosen to mark traditional room-based video conferencing and desktop video collaboration with a DSCP marking of AF41.

Figure 29 Media Marking within a TelePresence Meeting with SD Interoperability Enabled

In the example above, the TelePresence CUCM cluster has been set to instruct video endpoints to send media with a DSCP marking of CS4. Currently, CUCM cannot instruct the CTMS endpoint to send the high-definition media (H.264 1080p or 720p video and the AAC-LD audio) marked as DSCP CS4, but send the standard-definition media (H.264 CIF video and G.711 audio) for interoperability as AF41. Therefore, both the SD and HD media from the TelePresence endpoint is sent with a DSCP marking of CS4.

However, the CTMS is capable of changing the marking of the SD interoperability media streams (H.264 CIF video and G.711 audio) as it switches them to the CUCV. This is controlled within the QoS Settings screen of the CTMS, as shown in [Figure 30](#).

Figure 30 QoS Settings within the CTMS

Therefore, the interoperability media can be remarked to AF41 by the CTMS as it is switched through to the CUVC via the active cascade segment. The CUVC can then switch and/or transcode the media to the endpoints keeping the same AF41 marking.

**Note**

Depending on the configuration of the CUVC, audio and video may be sent to separate IP addresses.

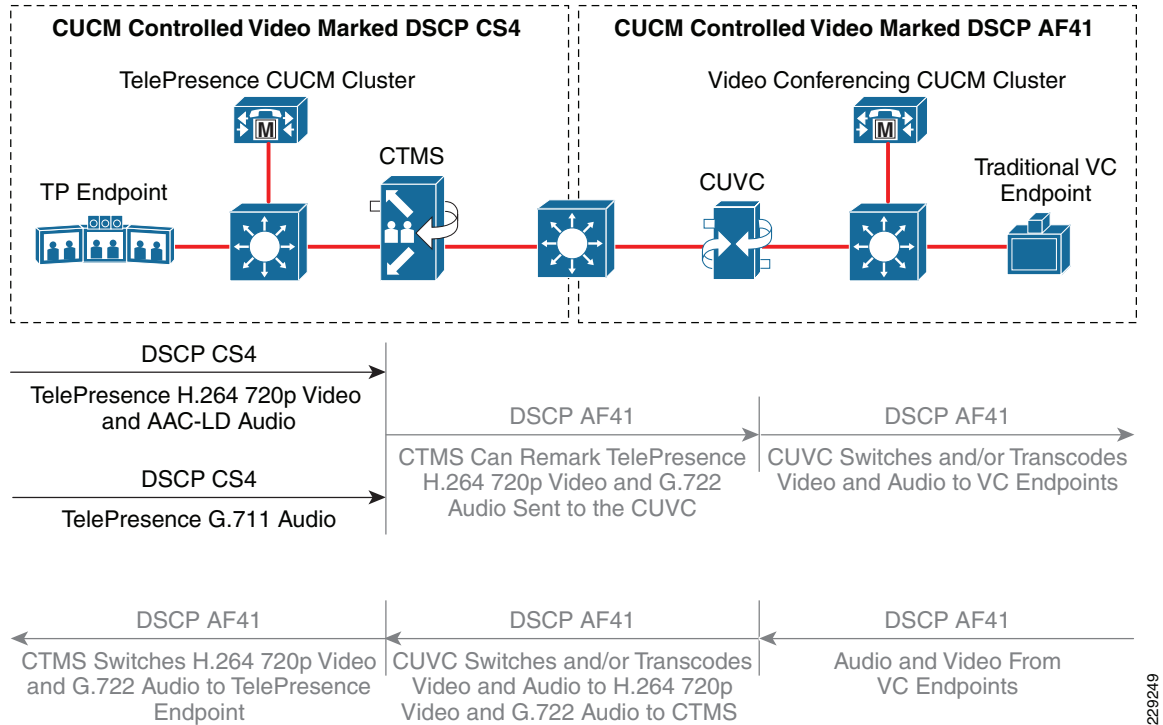
The Video Conferencing CUCM cluster, shown in [Figure 29](#), controls the marking of media originated by video endpoints controlled by it. If the system-wide setting of the CUCM cluster is set to instruct video conferencing endpoints to send media with a DSCP marking of AF41 (based on recommendations above), the audio and video media sent from the CUVC to the CTMS will have a DSCP marking of AF41.

**Note**

The design engineer should note that not all video conferencing endpoints are necessarily controlled by the CUCM cluster. Additionally, not all operating systems allow CUCM to control the DSCP marking of the media originating from the conferencing endpoint. For those endpoints not controlled by the CUCM, Cisco Catalyst switches can be used to classify and mark ingress video and audio traffic to the desired DSCP marking.

The CTMS does not re-mark inbound media from the CUVC that is destined for the CTS endpoints. Therefore, the H.264 CIF video and G.711 audio sent from the CTMS to the CTS endpoints remain marked AF41. This leads to somewhat of an asymmetric marking of the interoperability media streams across the network infrastructure, of which the design engineer should be aware.

[Figure 31](#) shows a similar example of the marking of video and audio in a TelePresence meeting with HD interoperability enabled.

Figure 31 Media Marking within a TelePresence Meeting with HD Interoperability Enabled

Again, the network design engineer should note that the H.264 720p video streams and AAC-LD audio streams sent and received by TelePresence endpoints are all marked as CS4. The H.264 720p video streams and G.722 audio streams sent by a TelePresence endpoint to the CTMS are also marked as CS4. However, any H.264 720p video stream and G.722 audio stream sent by the CUVC is not remarked back to CS4 by the CTMS. Therefore, with this type of QoS setting, the audio and video received from the traditional video conferencing devices that is switched through the CTMS to the TelePresence endpoints have a different DSCP marking (AF41) on the network.

When configuring policy maps on Cisco routers and switches within the campus WAN edge or branch WAN edge, the design engineer should be aware that the audio and video media within a TelePresence call that includes interoperability may sit in multiple traffic classes. The following is an example of the application of the 12-class QoS map on a Catalyst 6500 OC-3 POS interface:

Example 1 Sample Policy Map

```
class-map match-all BROADCAST-VIDEO
  match ip dscp cs5
class-map match-all BULK-DATA
  match ip dscp af11 af12 af13
class-map match-all NETWORK-CONTROL
  match ip dscp cs6
class-map match-all MULTIMEDIA-CONFERENCING! Desktop Video Collaboration and Room-Based
! Conferencing
  match ip dscp af41 af42 af43
class-map match-all OAM
  match ip dscp cs2
class-map match-all REAL-TIME-INTERACTIVE! TelePresence and other High-End Video
! Conferencing
  match ip dscp cs4
class-map match-all SCAVENGER
  match ip dscp cs1
```

```

class-map match-any CALL-SIGNALING
  match ip dscp cs3
class-map match-all TRANSACTIONAL-DATA
  match ip dscp af21 af22 af23
class-map match-all VOIP-TELEPHONY
  match ip dscp ef
class-map match-all MULTIMEDIA-STREAMING
  match ip dscp af31 af32 af33
!
!
policy-map OC-3-WAN-EDGE
  class VOIP-TELEPHONY
    police cir 4750000 bc 148437 be 148437 conform-action transmit exceed-action drop
  violate-action drop
    priority
  class REAL-TIME-INTERACTIVE
    police cir 49500000 bc 720000 be 720000 conform-action transmit exceed-action drop
  violate-action drop
    priority
  class NETWORK-CONTROL
    bandwidth percent 5
  class CALL-SIGNALING
    bandwidth percent 5
  class OAM
    bandwidth percent 5
  class MULTIMEDIA-CONFERENCING
    bandwidth percent 10
  class MULTIMEDIA-STREAMING
    bandwidth percent 5
  class BROADCAST-VIDEO
    bandwidth percent 5
  class TRANSACTIONAL-DATA
    bandwidth percent 5
  class BULK-DATA
    bandwidth percent 4
  class SCAVENGER
    bandwidth percent 1
  class class-default
    bandwidth percent 20
    random-detect

```

In the policy map above, separate traffic classes have been defined for Real-Time Interactive traffic that contains TelePresence media with a DSCP marking of CS4, and Multimedia Conferencing traffic that contains both desktop video collaboration and traditional room-based video conferencing media with a DSCP marking of AF41. Each traffic class is allocated a percentage of the bandwidth across the OC-3 POS. For this example, the Real-Time Interactive class is mapped to an LLQ configuration via the **priority** command, and policed at approximately 32 percent of the bandwidth of the OC-3 circuit, or 49.5 Mbps. Alternatively, Telepresence could be placed in a CBWFQ if desired. The Multimedia-Conferencing class is mapped to a CBWFQ and allocated approximately 10 percent of the bandwidth of the OC-3 circuit, or 15.5 Mbps. Note that the queue-limits for each traffic class have been left at their default sizes in Example 1 above, for simplicity. The queue limits may need to be increased in the Real-Time-Interactive and Multimedia-Conferencing classes to accommodate packet bursts generated by TelePresence.

This type of policy map configuration guarantees that both the Real-time Interactive and the Multimedia Conferencing traffic are allocated a percentage of the bandwidth of the circuit. Further, TelePresence traffic is protected from traditional video conferencing and desktop video collaboration systems taking all of the bandwidth of the circuit, and vice-versa. Both TelePresence and traditional video conferencing systems are also protected from other video or data applications taking all of the bandwidth.

Summary

Cisco TelePresence interoperability allows traditional room-based video conferencing systems as well as desktop video collaboration systems to participate within a Cisco TelePresence virtual meeting. This solution provides standards-based interoperability with minimal additional hardware requirements through a cascade segment between the CUVC MCU and the CTMS. As of CTS version 1.6, both SD and HD interoperability are supported. For SD interoperability, an additional 704 Kbps CIF video stream and a 64 Kbps G.711 audio stream is generated by each TelePresence endpoint. For HD interoperability, all TelePresence endpoints are constrained to H.264 720p video, but no additional video stream is generated to support the interoperability. However, an additional 64 Kbps G.722 audio stream is generated by each TelePresence endpoint. The additional bandwidth requirements, potential for additional latency within the overall meeting, and QoS configurations need to be considered when designing the network infrastructure to support TelePresence interoperability.

