CHAPTER 8

Interoperability with Legacy Video Conferencing Devices

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Contents

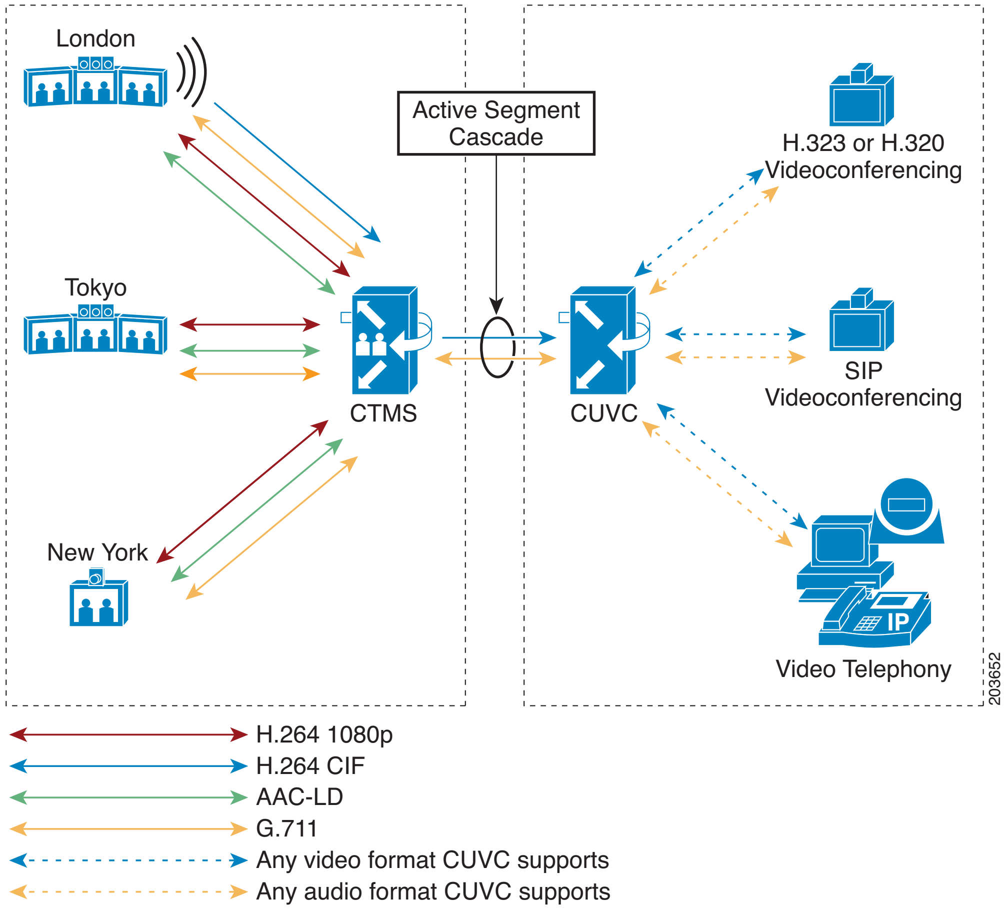
Overview

Cisco TelePresence is based on open standards, including SIP, H.264 and G.711. With Cisco TelePresence System (CTS) Release 1.3 and CTMS Release 1.1, Cisco TelePresence now supports interoperability between Cisco TelePresence systems and standard definition video conferencing/video telephony using the Cisco Unified Video Conferencing 3500 series MCU (CUVC).

How Cisco TelePresence Interoperability Works

As shown in Figure 8-1, CTS endpoints send a copy of their audio in G.711 format to the CTMS server. CTMS then determines which CTS segment is emitting the most dominant audio and requests that segment to send a copy of that segment's video in Common Intermediate Format (CIF) resolution. CTMS mixes the G.711 channels for all CTS endpoints into a single G.711 audio and switches CIF and G.711 to CUVC. As the dominant audio segment changes throughout the meeting, CTMS switches the CIF video stream accordingly. Audio-only participants on the CUVC side can join directly into the CUVC.

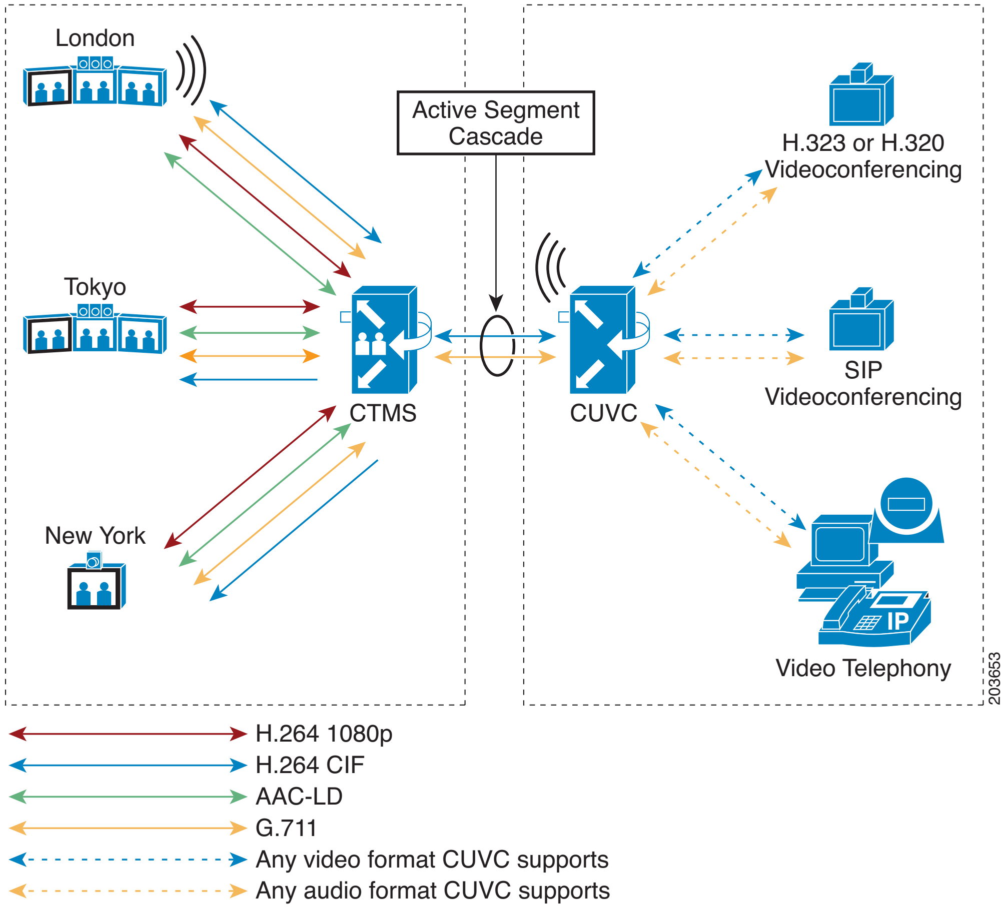
Figure 8-1 Cisco TelePresence Interoperability: From CTS/CTMS to CUVC



As shown in Figure 8-2, audio and video coming from CUVC to CTMS is switched to all CTS endpoints when the audio coming from CUVC is deemed to be the most dominant segment. The CIF image from CUVC is presented on the left screen of each CTS-3000 surrounded by black borders. With the CTS-1000, the CIF image is displayed when CTMS senses that the CUVC participant is the active speaker. The incoming 352x288 CIF image is stretched to 4CIF when displayed on either the CTS-1000 or CTS3000 high-definition plasma screen.

Note If you wish to see more than one CUVC participant displayed at one time, you can customize the CUVC MCU layout configuration to display up to 16 CUVC participants. We recommend the 1x1 layout for consistency with the Telepresence experience.

Figure 8-2 Cisco TelePresence Interoperability: From CUVC to CTS/CTMS



Benefits

Cisco TelePresence Interoperability maintains the immersive experience for Cisco TelePresence meeting participants. It also provides standards-based interoperability with minimal additional hardware (CUVC 3500 Series MCU) requirements. CTS and CTMS software upgrades are available at no charge.

Caveats

- Cisco TelePresence Interoperability increases the required amount of bandwidth to and from each CTS by an additional 704 kbps to transmit and receive the CIF and 64 kbps to transmit and receive G.711 streams.
- The interoperability segment is limited to CIF resolution video at 768 kbps and G.711 audio in CTMS Release 1.1.
- CUVC is the only supported MCU in this release. CUVC requires software version 5.5.0.0.54 or later.

- CUVC participants will not experience the spatial audio generated by CTS. CUVC participants hear all Cisco TelePresence participants mixed together in G.711.
- CUVC as an SCCP conference resource managed by Cisco Unified CM is not a supported option.The CUVC must be defined as a SIP trunk (and optionally an H.323 gateway) to Cisco Unified CM.
- CTS participants hear CUVC participants mixed together in G.711; audio will be heard from the same segment as that which is showing the CIF video.
- CTS and CUVC participants are not able to share slides or documents using H.239. Use another application for document sharing between CTS and CUVC, such as MeetingPlace or WebEx.
- Far End Camera Control (FECC) is not available to CTS participants during an interoperability call.
- Each interop conference uses one CTMS port and one CUVC port for each conference because each meeting utilizes a port for the cascade link. For example, CTMS normally can support 48 segments, and CUVC-3515-24 can support up to 24 participants. During interop meetings, CTMS supports a maximum of 47 segments and CUVC supports a maximum of 23 participants.
- Encryption is not supported for CTMS, Release 1.1, including interoperability calls.

Prerequisites

Interoperability between traditional video conferencing devices and CTS requires three components:

- CTMS
- Cisco Unified Communications Manager (Cisco Unified CM)
- CUVC MCU

The software and hardware requirements for Cisco TelePresence Interoperability are as follows:

CUVC Software and Hardware Requirements

- CUVC MCU Release 5.5.0.0.54:MP or later
- CUVC EMP Release 5.5.2.0.2:EMP or later

For additional information about configuring CUVC, refer to the *Configuration Guide for the Cisco Unified Videoconferencing 3545 MCU Release 5.5.*

Cisco Unified CM Software Requirements

- Cisco Unified CM Software Release 6.0 or later

For additional information about configuring Cisco Unified CM for Cisco TelePresence System, refer to the *Cisco Unified Communications Manager Installation Guide for the Cisco TelePresence System*.

For information about configuring Cisco Unified CM, refer to *Cisco Unified Communications Manager Version 6.0* and *Cisco Unified Communications Manager Version 6.1*.

CTS Endpoint Software Requirements

- CTS Software Release 1.3 or later

For more information about CTS Administration Software, refer to the *Cisco TelePresence System Release 1.3 Administrator's Guide*.

Configuring Cisco TelePresence Interoperability

To configure Cisco TelePresence Interoperability, you must complete the following configuration tasks:

- Configure Cisco Unified CM to support Cisco TelePresence Interoperability
- Configure CUVC to support Cisco TelePresence Interoperability
- Add at least one static meeting to CTMS with interoperability enabled

Configuring Cisco Unified CM for Cisco TelePresence Interoperability

To configure Cisco Unified CM for Cisco TelePresence Interoperability, you must create a SIP trunk security profile and a SIP trunk using the same configuration parameters as you would in defining a SIP trunk for CTMS. Then you must add a new route pattern in Cisco Unified CM that points to the CUVC. CTMS dials the CUVC using the number defined in the “CUVC number” field of the CTMS meeting definition when the first CTS participant joins the meeting.

Creating a SIP Trunk Security Profile

To create a SIP trunk security profile:

Step 1 Click ***System***. Under **Security Profile**, click ***SIP Trunk Security Profile***.

Step 2 Click the ***Add New*** button at the bottom of the page or click the **+ *sign*** at the top of the page.

Step 3 Enter the settings as indicated in Table 8-1 to configure the SIP trunk security profile. Leave default settings for fields not included in Table 8-1.

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Table 8-1 SIP Trunk Security Profile Settings

Field	Required	Setting
Name	Yes	Enter a text string identifying this SIP trunk security profile.
Description	—	Enter a text string describing this SIP trunk security profile.
Device Security Mode	Yes	Select ***Non Secure***.
Incoming Transport Type	Yes	Select ***TCP+UDP***.
Outgoing Transport Type	Yes	Select ***TCP***.
Incoming Port	Yes	Enter ***5060***.

Step 4 Click the ***Save*** button at the bottom of the page.

Note Use the same SIP Trunk Security Profile settings for the CTMS SIP trunk and the CUVC SIP trunk.

Creating a SIP Trunk

To create a SIP trunk for CTMS calls to the CUVC:

Step 1 Click ***Device***. Click ***Trunk***.

Step 2 Click the ***Add New*** button at the bottom or click the **+ *sign*** at the top of the Trunk Configuration page.

Step 3 Select ***SIP Trunk*** from the **Trunk Type** pull-down menu, then click ***Next***.

Step 4 Enter the settings as indicated in Table 8-2 to configure the SIP trunk. Leave default settings for fields not included in Table 8-2.

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Table 8-2 SIP Trunk Settings

Field	Required	Setting
Device Information		
Device Name	Yes	Enter a text string identifying this SIP trunk.
Description	—	Enter a text string describing this SIP trunk.
Device Pool	Yes	Select ***Default***.
SIP Information		
Destination Address	Yes	Enter the IP address of the CUVC.
SIP Trunk Security Profile	Yes	Select the SIP trunk security profile that you created for CTMS.
SIP Profile	Yes	Select ***Standard SIP Profile***.

Step 5 Click the ***Save*** button at the bottom of the page.

Configuring a Route Pattern

A route pattern allows a Cisco Unified CM-managed device to access another device by dialing its number. Such devices may include gateways, Cisco TelePresence Multipoint Switch (CTMS) systems, or Cisco Unified Video Conferencing (CUVC) units. Each device requires its own unique route pattern.

To configure a route pattern:

Step 1 Click ***Call Routing***. Under **Route/Hunt**, click ***Route Pattern***.

Step 2 Click the ***Add New*** button at the bottom or click the **+ *sign*** at the top of the Route Pattern Configuration page.

Step 3 Enter the settings as indicated in Table 8-3 to configure the SIP trunk. Leave default settings for fields not included in Table 8-3.

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Table 8-3 Route Pattern Configuration Settings

Field	Required	Setting
Pattern Definition		
Route Pattern	Yes	Enter the route pattern, including numbers and wildcards (do not use spaces); for example, for NANP, enter 9.@ for typical local access, or 8XXX for a typical private network numbering plan. The uppercase characters A, B, C, and D are valid characters. **Note** The portion of the route pattern's digit string sent to the CUVC must begin with a valid service prefix as defined on the CUVC. **Note** See the "Wildcards and Special Characters in Route Patterns and Hunt Pilots" section in the *Cisco CallManager System Guide* for more information about wildcards.
Description	—	Enter a text string describing this route pattern.
Gateway/Route List	Yes	Select the SIP trunk that you created for CUVC.
Call Classification	Yes	Select ***OnNet***.

Step 4 Click the ***Save*** button at the bottom of the page.

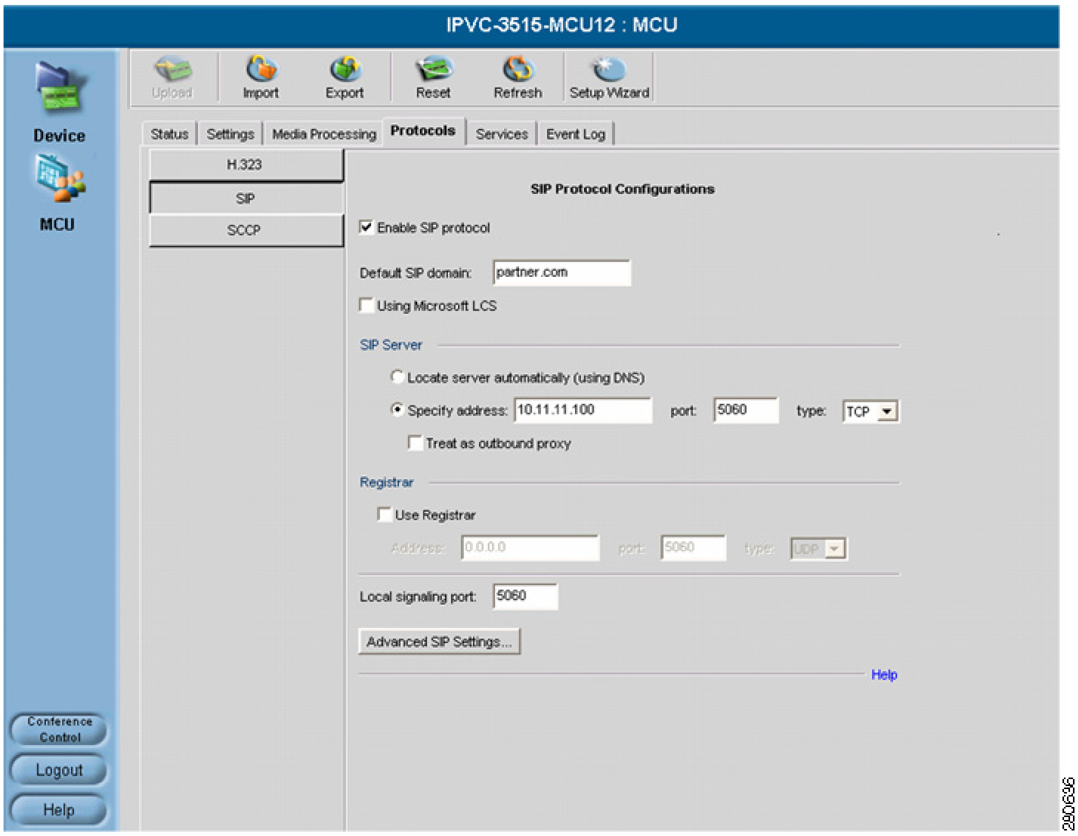
Configuring CUVC for Cisco TelePresence Interoperability

After basic CUVC configuration has been completed and the unit has been connected to the host network, you need to perform the following tasks to configure the CUVC for Cisco TelePresence Interoperability:

- Verify that the MCU and EMP are running the correct software version.
  - Log into the CUVC Administration console and click the ***Media Processing*** tab.
  - Verify that the MCU is running Version 5.5.0.0.54:MP.
  - Verify that all EMPs are running Version 5.5.2.0.2:EMP.
- EMP resources must be available and the MCU service must be defined to use HD/SD Continuous Presence with a Max Call Rate of 768 kbps or greater. "HD Switch Mode" is not supported. If you choose a service configuration such as "HD Switched Video," the cascade connection between CTMS and CUVC will fail to connect.

- Enable SIP signaling on the CUVC. A SIP proxy is not required for interoperability.
  - From the ***Protocols*** tab, click **SIP.** Click the ***Enable SIP Protocol*** box as shown in Figure 8-3.

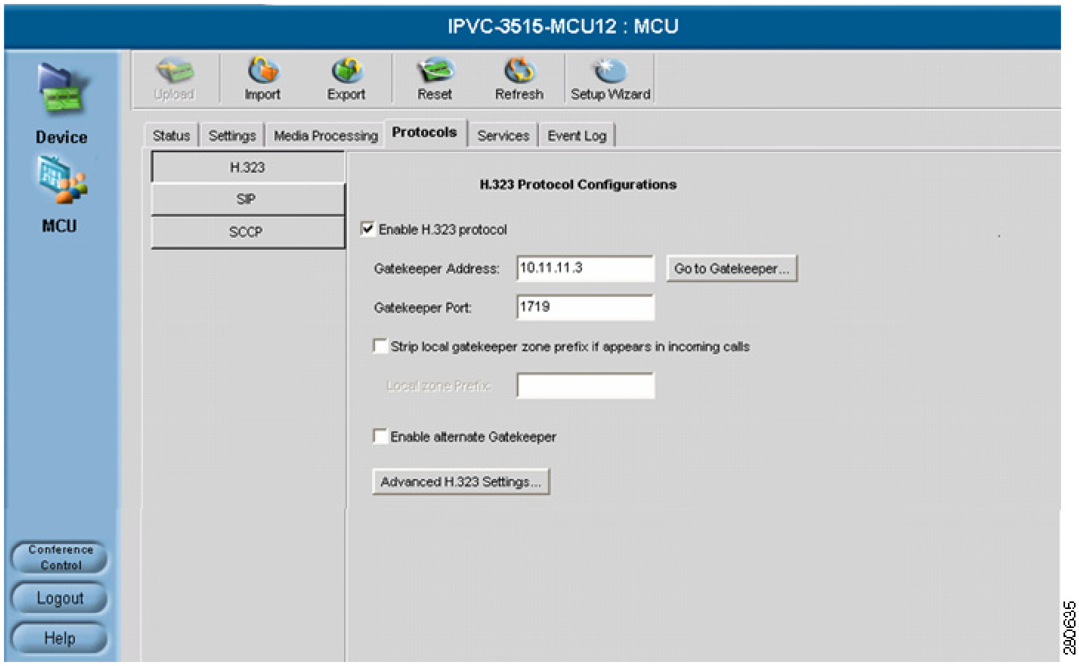
Figure 8-3 SIP Configuration Screen



Note SIP must be enabled on the CUVC. CTMS must connect to CUVC using SIP. When CTMS dials the CUVC number, it sends a SIP INVITE to Cisco Unified CM for that number. Cisco Unified CM must be configured to route that number to CUVC using a SIP trunk. CUVC can connect to legacy video conferencing participants using either SIP or H.323.

- (Optional) Disable the SCCP protocol to save ports.
  - From the ***Protocols*** tab, click **SCCP** as shown in Fix XX**.** Click the ***Disable SCCP Protocol*** box.
- Enable H.323 signaling on the CUVC.
  - From the ***Protocols*** tab, click **H.323** and the click the ***Enable H.323 Protocol*** box as shown in Figure 8-4.

Figure 8-4 H.323 Configuration Screen

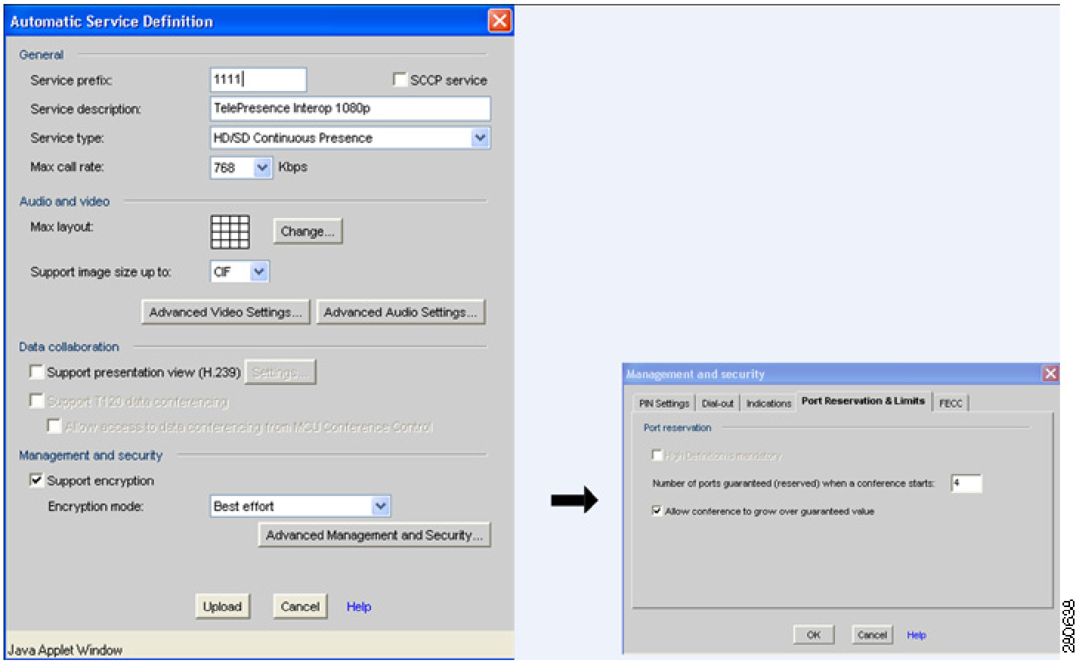


- If you are using an H.323 Gatekeeper, define the Gatekeeper IP address and the Gatekeeper Port as 1719.
- If you are using an MCU as an H.323 trunk in Unified Communications, go to ***Advanced H.323 Settings*** on the CUVC Protocols configuration page and record the RAS port number and Signaling Port parameter value. Use the Signaling Port value for the Unified Communications H.323 Trunk Port definition.

- Configure CUVC meeting characteristics for Cisco TelePresence Interoperability. CUVC meeting characteristics are configured individually using the service prefix. Service prefixes define the CUVC meeting characteristics, and are defined on the CUVC Administration page under ***Services***.

  From the **Automatic Service Definitions** page under ***Services*** as shown in Figure 8-5, configure the following attributes:
  - Max Call Rate: 768 Kbps
  - Max layout: 1x1
  - Supported image size up to: CIF

Figure 8-5 Automatic Service Definition Screen



Click the **Advanced Management and Security** button under ***Automatic Services*** to open the Management and Security Screen. Click the **Port Reservation and Limits** tab. Configure the following attributes:

- Define the minimum number of guaranteed ports for interoperability conferences.
- Check the ***Allow conference to grow over guaranteed value*** box.
- Click ***OK***.

From the **Advanced Audio Settings** page under ***Automatic Services***, configure the following attributes:

- Verify that G.711 is in the supported codecs list; select ***G.711***.

From the **Advanced Video Settings** page under ***Automatic Services***, configure the following attributes:

- Check ***Enable "No Self See"*** box.
- Verify that H.264 is in the supported codecs list; select ***H.264***.
- Click ***Settings*** to enter welcome text as required.

Click ***Upload*** for these changes to take effect.

Note The route pattern created for CUVC in Cisco Unified CM should match the CUVC number and service prefix configured in CUVC.

- Suppress reflected video for the initial caller. When a caller joins the CUVC as the first participant of a Cisco Telepresence conference, the default behavior is for that caller to see his or her own image. To have that initial caller see a black screen instead, you must suppress reflected video. To suppress the reflected video:
  - From the MCU configuration page, select the ***Settings*** tab, then click ***Advanced***. The page refreshes and displays a ***Command*** button.
  - Click ***Commands***. The Advanced Commands window appears.
  - In the Commands field, enter: **mc:notselfseeforfirstpart**
  - In the Parameter field, enter the service prefix number followed by either **0** (to deactivate) or **1** (activate) suppressed reflection.
  - Click ***Send*** to issue request to the CUVC. If the command is successful, the response field indicates "OK."
- Network Configuration: The CUVC MCU and EMP cards each have their own Ethernet cables and IP addresses. Make sure that the switch to where the two cables attach is defined to allow QoS to be passed through to the network as a trusted device. This requirement applies to all video devices including CTS and CTMS. Configuration example is as follows:

```
interface TenGigabitEthernet 4/2
  description ===connection to telepresence gateway 2===
ip address 10.xx.xx.xx 255.255.255.252
ip pim sparse-dense-mode
mls qos trust dscp end
```

For general information about the CUVC configuration tasks, refer to the *Configuration Guide for the Cisco Unified Videoconferencing 3545 MCU Release 5.5*.

Configuring CTMS for Cisco TelePresence Interoperability

The next step is to configure a static meeting in CTMS in "interop mode." When the first CTS caller joins the teleconference, CTMS dials out to the CUVC using the Interoperability number configured in the CTMS meeting definition. CTMS dials out to the CUVC using the SIP trunk defined in Cisco Unified CM. CTMS cannot use H.323 signaling for this connection. When configuring the SIP trunk, specify a device pool with a region configured for a video bandwidth of 768 kpbs or higher.

Creating Static Meetings in CTMS for Interoperability

To create a static meeting:

Step 1 Click ***Static Meetings*** under the **Meetings Management** folder in the Navigation Pane.

Step 2 The Static Meetings setting screen initially displays a table providing the following information about already defined static meetings.

Table 8-4 Static Meetings Table Field Descriptions

Field	Description
Access Number	Displays the access number that rooms call to attend this meeting.
Description	Displays the defined description for this static meeting.

Table 8-4 Static Meetings Table Field Descriptions

Field	Description
Switching Policy	Displays the defined switching policy (site or segment) for this static meeting.
Max Rooms	Displays the maximum number of sites that can participate in this static meeting.
Quality	Sets the maximum bit rate and video resolution to be used for the meeting.
Interop	A green check indicates that this particular Cisco TelePresence multipoint meeting supports Cisco Unified Video Conferencing (CUVC) systems (interoperability mode). A red "X" indicates that this meeting is not configured to cascade with CUVC systems.
CUVC Number	(Optional) Number dialed to CUVC for interoperability meetings.

- To delete one of the defined static meetings, click the radio button to the left of the table entry, and then click ***Delete***.
- To edit one of the defined static meetings, click the radio button to the left of the table entry, and then click ***Edit***.
- To define a new static meeting, click ***New***.

Step 3 When you click ***Edit*** or ***New***, CTMS Administration software takes you to the Static Meeting Settings table. Enter settings as described in Table 8-5:

Table 8-5 Static Meeting Settings

Field or Button	Setting
Access Number	Defines the telephone number that participants call to attend this static meeting.
Meeting Description	Text describing or identifying this static meeting. The maximum number of characters for this field is 62 characters.
Switching Policy	Defines how CTMS calls are displayed during a meeting. CTMS displays active speakers on screen. There are two active speaker display options: • ***Segment***: (Speaker) With segment switching, each individual table segment (defined as a display and a camera) is displayed on the screen as that segment becomes the active speaker. • ***Site***: (Room) When you select "site," all table segments for a particular room are displayed on screen when any segment in that room is the active speaker. Click the appropriate radio button to select. **Note** If you are running CTS 1.3 or later, you can control how Cisco TelePresence calls are displayed from the Cisco TelePresence phone interface. Press the ***Speaker*** softkey to display the active segment; press the ***Room*** softkey to display all segments from a particular site.

Table 8-5 Static Meeting Settings

Field or Button	Setting
Maximum Rooms	Defines the maximum number of Cisco TelePresence rooms allowed to dial into in a static multi-point meeting. The range for this setting is from 2 to 48.
Video Announce	If this option is selected, when a new room joins the meeting, the new room is displayed on-screen for 2 seconds. Options are ***Yes*** and ***No***. Click the appropriate radio button to select.
Hosted Meeting	Hosted meetings mean that one particular room is identified as the host for a meeting; other meeting rooms will not be added to the meeting until the host room dials in. If you have selected "Video announce," then each meeting room will be displayed in 2-second intervals in the order that they joined the meeting. Options are ***Yes*** and ***No***. Click the appropriate radio button to select.
Host Room Number	Defines the host Cisco TelePresence System room number.
Quality	This field sets the system bandwidth and screen resolution. A higher bandwidth increases video quality, but may also cause packets to be dropped and video to be interrupted. Choices: • Highest Detail, Best Motion: 4Mbps 1080p • Highest Detail, Better Motion: 3.5Mbps, 1080p • Highest Detail, Good Motion: 3Mbps, 1080p • Highest Detail, Best Motion: 3Mbps, 720p • Highest Detail, Better Motion: 2Mbps, 720p • Highest Detail, Good Motion: 1Mbps, 720p Default is Highest Detail, Best Motion: 4Mbps 1080p.
Interop	Determines whether this particular Cisco TelePresence multipoint meeting accepts legacy Cisco Unified Video Conferencing (CUVC) systems (interop). Options are **Yes** and **No**. Click the appropriate radio button to select.
CUVC Number	Defines the number that CTMS dials to establish contact with CUVC. Each CUVC number must be unique for each CTMS conference. The CUVC number consists of the service prefix and then the remaining dialed digits. The service prefix can be the same for different meetings. The remaining digits in the dialed number designate the CUVC meeting instance. Each CTMS conference requires it owns CUVC meeting instance. **Note** This number must start with the CUVC service prefix defined during CUVC configuration.

- To register new or modified settings, click ***Apply***.
- To restore the original settings, click ***Reset***.

Troubleshooting Cisco TelePresence Interoperability

Table 8-6 describes some specific problems and possible solutions.

Table 8-6 Specific Problems and Possible Solutions

Problem	Possible Solutions
Cisco Unified CM sends an error message of "Service not available" to CTMS when CTMS tries to establish call to CUVC.	• Check to see if there are sufficient ports. Cisco Unified CM delivers a "Service not available" message when CTMS places a call to CUVC and there are an insufficient number of ports available. • Disable SCCP on the CUVC.